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Editorial Preface

From the Desk of Managing Editor...

It may be difficult to imagine that almost half a century ago we used computers far less sophisticated than current home desktop computers to put a man on the moon. In that 50 year span, the field of computer science has exploded.

Computer science has opened new avenues for thought and experimentation. What began as a way to simplify the calculation process has given birth to technology once only imagined by the human mind. The ability to communicate and share ideas even though collaborators are half a world away and exploration of not just the stars above but the internal workings of the human genome are some of the ways that this field has moved at an exponential pace.

At the International Journal of Advanced Computer Science and Applications it is our mission to provide an outlet for quality research. We want to promote universal access and opportunities for the international scientific community to share and disseminate scientific and technical information.

We believe in spreading knowledge of computer science and its applications to all classes of audiences. That is why we deliver up-to-date, authoritative coverage and offer open access of all our articles. Our archives have served as a place to provoke philosophical, theoretical, and empirical ideas from some of the finest minds in the field.

We utilize the talents and experience of editor and reviewers working at Universities and Institutions from around the world. We would like to express our gratitude to all authors, whose research results have been published in our journal, as well as our referees for their in-depth evaluations. Our high standards are maintained through a double blind review process.

We hope that this edition of IJACSA inspires and entices you to submit your own contributions in upcoming issues. Thank you for sharing wisdom.

Thank you for Sharing Wisdom!

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Content Based Image Retrieval Using Gray Scale Weighted Average Method

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Abstract—High feature vector dimension quietly remained a curse element for Content Based Image Retrieval (CBIR) system which eventually degrades its efficiency while indexing similar images from database. This paper proposes CBIR system using Gray Scale Weighted Average technique for reducing the feature vector dimension. The proposed method is more suitable for color and texture image feature analysis as compared to color weighted average method as illustrated in literature review. To prove the effectiveness of retrieval system, two standard benchmark dataset namely, Wang and Amsterdam Library of Texture Images (A LOT) for color and texture have been selected to evaluate the system retrieval accuracies as well as efficiencies generated by each method. For the purpose of image similarity, Euclidean distance has been employed which matches query image feature vector with image database feature vectors. The experimental results generated by two methods showed that overall performance of the proposed method is relatively better in terms of average precision, average recall and its average retrieval time.

Keywords—Color Weighted Average Method; Gray Scale Weighted Average Method; Feature Extraction; Precision; Recall; CBIR

I. INTRODUCTION

Due to over increasing growth in digital images on the web their storage management as well as retrieval is becoming a challenging task. Traditional Text Based Image Retrieval (TBIR) system is an inadequate method to retrieve similar images from the databases, because it is difficult to extract visual features with the aid of textual information. To address this problem the Content Based Image Retrieval System (CBIR) was introduced in early 1990, since then this area has been most extensively researched for images. The CBIR aims to retrieve semantically relevant images from database on the basis of low level visual image features such as color, shape, texture and object location. However, content based image retrieval system also suffers from either feature extraction complexity or by high dimensional feature matching [1]. Because of these factors image retrieval system outcome decreases in terms of accuracy and at the same time system behaves poorly in response time. Until now many image

retrieval methods have been proposed by many authors where they have presented multilevel feature extraction using color, texture and shape to improve the accuracy of image retrieval process. And also various indexing and machine learning techniques have been implemented to reduce the feature vector dimension for improving the efficiency of system retrieval [2]. For instance, Tree Based Index (R Tree), Hash Based Index (LSH), Visual Based Index (BOW) [3], Principle Component Analysis (PCA), Kernel Independent Component Analysis (KPCA) and Multidimensional Scaling (MDA) [4]. However, reducing the feature vector dimension after features extraction certainly losses discriminative visual features from the image, which eventually affects the accuracy and efficiency of the retrieval system.

The goal of this paper is to introduce one dimensional color feature vector for image retrieval using gray scale weighted average method. The similar method has been used in [7] that take weighted average of RGB color image. The proposed method uses gray scale intensity image instead of color image because it occupies less space in memory and retains important features related to the image. The rest of present paper is arranged as follow: The literature review has been presented in Section II. The methodology of proposed method has been explained in Section III. Section IV, demonstrates the experimental results and discussion. The efficiency of each system on two different color and texture dataset has been evaluated in Section V.

The conclusion has been given in Section VI and future work is elaborated in Section VII.

II. LITERATURE REVIEW

Since last two decades, various color based image feature extraction schemes have been suggested by many researchers for CBIR. A simple histogram method for color image retrieval was proposed by Swain *et al.* [5], which counts total number of occurrence of each distinct color in the image through multi-dimensional feature vectors. Similarly Kekre *et al.* [6] suggested color average technique. In this paper, total average of row and column from each red, green and blue have been calculated to form a feature vector. However it also

supported multidimensional structure for image retrieval. Moreover Ali *et al.* [7] described a novel color image retrieval using RGB color weighted average method. In that work, firstly feature vector of corresponding red, green and blue color have been added together that forms one dimensional structure. Secondly histogram is generated to find unique pixel values and then using probability distribution function weighted average feature vector is formed. The generated feature vector has been used for image to image similarity measure from the database. Color image retrieval using visual weighted block was demonstrated by Wu *et al.* [8]. In this paper, authors have divided an image into block and generated histogram for it. After that each pixel was given a saliency score for finding the local pixel variation as to form separate visual weighted blocks. The obtained sub-blocks are concatenated for color image similarity. In other paper An and Le [9], presented color feature extraction from salient region with spatial layout. They have used color contrast method for silent region and spatial layout for each dominant color, which was obtained through binary map. Color based image retrieval was presented by Varish and Kumar [10]. Here probability histograms for each RGB plane have been generated and from them significant bins are obtained. The statistical values are calculated from each bin to describe the feature vectors. Color histogram and color correlogram approach for image retrieval was given by Soni and Mathai [11]. In this paper, HSV and RGB color space have been used for local and global color histogram analysis. And color correlogram is used for image spatial layout. Finally both color feature were combined together to form a feature vector. Image retrieval using fusion of cell color histogram (CCH) and color coherence vector (CCV) was presented by Salami and Boucheham [12], in their method classification of coherent and non-coherent pixels have been performed using CCV and then CCH was implemented for color distinction.

III. GRAY SCALE WEIGHTED AVERAGE METHOD

In this paper, gray scale weighted average method for content based image retrieval has been proposed and also it is compared with colorful image weighted average method. The proposed algorithm works as similar to color weighted average method [7]; except it takes grey scale intensity image for image retrieval. The gray scale image produces different shades of pixels with eight bit value, which range from 0 – 255 as opposite to color image, which assigns eight bit value to each red, green and blue planes separately. Similarly each plane has distinct value from 0 – 256. The advantage of using gray scale intensity image is that it simplifies amount of information than three dimensional color images and also takes less space in the memory [13]. The key information related to image in grey scale is not lost such as edges, regions, blobs etc. In Fig. 1 (a) and (b), RGB color fruit's image is converted into gray scale image as shown below. However one can still distinguish red and green apples from gray scale image.

The formula for converting RGB color image to gray scale image is given in equation 1.

$$(n, m) = \alpha I_{colour(n,m,r)} + \beta I_{colour(n,m,g)} + \gamma I_{colour(n,m,b)} \quad (1)$$



Fig. 1. (a) RGB color image (b) Grey scale image

In this paper, grey scale intensity image is used for image feature analysis as its algorithm is described in Table I below.

TABLE I. ALGORITHM FOR GREY SCALE WEIGHTED AVERAGE METHOD

The proposed weighted average algorithm has been divided into six steps, which are explained below.

1. Convert RGB color image into Gray scale Image.
2. Generate image histogram and calculate the sum of the occurrence of all unique gray shades from this array.

$$\text{total} = \sum_{g=0}^N \text{his}(g) \quad (2)$$

3. Now apply probability distribution function for finding the occurrence of individual unique value in the gray scale image array.

$$\text{prob}(g) = \frac{\text{his}(g)}{\text{total}} \text{ where } 0 \leq g \leq N \quad (3)$$

4. Then calculate weighted average from unique values and their corresponding probabilities.

$$\text{weighted average} = \sum_{g=1}^N g \cdot \text{prob}(g) \quad (4)$$

5. Finally generate the feature vector value for gray scale image.

$$\text{Grayscale}_{\text{image}} \quad f_v = (g) \quad (5)$$

6. Compute Euclidean distance between the query image feature vector and image feature vector in database for retrieving most similar images. It can be calculated using following formula.

$$\text{Euclidean Distance} = \sqrt{\sum_{i=1}^n (q_i - d_i)^2} \quad (6)$$

Here, his(g) denotes the histogram of gray scale image and N represents the total number of unique values in the gray scale image. The prob(g) shows the probability of each unique value in the gray image array and fv represents obtained feature vector. The q_i and d_i measures distance between query image and images in the databases.

IV. EXPERIMENTAL RESULTS AND DISCUSSION

The experimental works for proposed method and color weighted average method have been generated using MATLAB. The image retrieval testing has been carried out on two standard benchmark datasets, one is Wang [14] database and another is ALOT [15]. These databases contain 500 and 700 images respectively; we have selected five different classes from each of them. Although pre-processing is implemented for resizing all database images into 162x162 resolutions. The performance of retrieval system for proposed and color average weighted method have been evaluated through average precision, average recall and its retrieval time.

precision: it is defined as how many like images are retrieved by the system as similar to the query image.

recall: it represents how many images are exactly recognized as similar to the query image.

retrieval time: it calculates total time taken by the system during image retrieval.

For image retrieval, threshold is used on each image dataset to retrieve top twenty most relevant images.

A. Image Retrieval Testing on Wang_Dataset

In this section, Wang data_set has been used for simulation. Here image retrieval testing has been demonstrated for proposed method and color average weighted method. Fig. 2 shows that a query of bus image is input to system using proposed method. The system has retrieved 12 relevant images out of 20 images. The system generated precision for proposed method is 60% and recall and retrieval time are 12%, 0.85 seconds respectively.



Fig. 2. Image retrieval using proposed method

The similar image in Fig. 3 is given as input to retrieval system using color average weighted method. The system has retrieved only 8 images as similar to the query while 12 images are irrelevant. The precision and recall obtained using color average weighted method are 40% and 8% respectively.

These are considerably lower as compared to proposed method.



Fig. 3. Image retrieval using color average weighted method

The color average weighted method retrieved images in 0.98 seconds. The performance of proposed method is quite high than color average weighted method on similar dataset.

B. Image Retrieval Testing on ALOT Data_Set

In this section, another experiment has been performed for further investigating the performance of proposed method. Here ALOT image dataset has been used to test retrieval query results. The obtained accuracy and efficiency of system has been compared with color average weighted method. In Fig. 4 below, coins texture image is input to the system using proposed method.

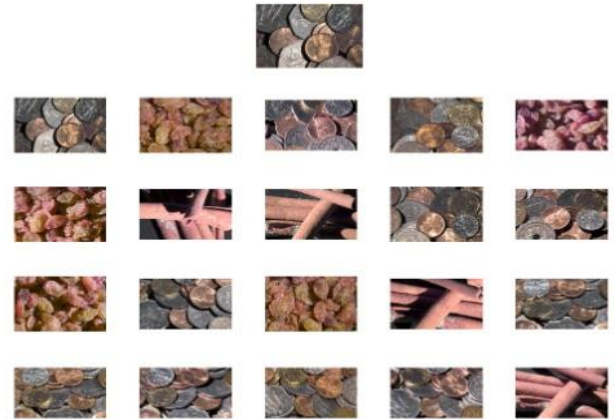


Fig. 4. Image retrieval using proposed method

The generated precision and recall rates using proposed method are 55%, 11% respectively i.e. retrieval system has retrieved 20 images out of them 11 are like matches as query image. The retrieval time it takes is 1.10 seconds.

Another experiment has been performed using color average weighted method. Fig. 5 depicts that similar image is input as reference to retrieval system. It has retrieved 10 images as similar to query image and rest of 10 are irrelevant. The system precision and recall rate are 50% and 10% respectively. It takes 1.26 seconds during image retrieval.



Fig. 5. Image retrieval using color average weighted method

V. RETRIEVAL SYSTEM PERFORMANCE ANALYSIS ON COLOR AND TEXTURE IMAGE DATASET

This section evaluates the performance of retrieval system using proposed method and color average weighted method. For this purpose, five different query image classes from each Wang and ALOT database are randomly selected for testing image retrieval system as shown in Fig. 6 and Fig. 7. It can be observed from Table II that average precision, average recall obtained by proposed method is quite higher than that of color average weighted method for Wang database images these are enriched with color features. To further validate the effectiveness of proposed method, it has been tested on ALOT database, which contains all images that have high concentration of textural features. The output generated by proposed method for texture images in terms of average precision, average recall are also shown increased in Table II as compared to color average weighted method. However retrieval time taken by proposed method for color and texture images database separately are 0.85 seconds and 1.17 seconds respectively. Similarly color average weighted method takes 0.96 seconds and 1.25 seconds for color and texture database images retrieval respectively. The advantage of proposed method is that it improved the accuracy without affecting retrieval system efficiency.

Fig. 8 and Fig. 9 are showing generated average precision and average recall for query results buses, dinosaurs, roses, horses and natures images. The proposed method has retrieved more like images as compared to color average weighted method, except for dinosaurs query image, where both have similar retrieval accuracies.

In Fig. 10 average retrieval time of system for Wang color images dataset is approximately equal for proposed and color average weighted method that is 0.85 seconds and 0.96 seconds respectively. The average precision and recall in Fig. 11 and Fig. 12 shows that accuracy of proposed method for ALOT textural images. This is quite improved for raisins and twined reed query images. Although for query fur, cinnamon sticks and coins images both retrieval systems have generated almost similar results. Fig.13 shows that computed retrieval time of system for ALOT textural image dataset by proposed method and color average weighted method, which are 1.17 seconds and 1.25 seconds respectively, which is almost similar.



Fig. 6. Wang color image data_set (image from [14])

TABLE II. PERFORMANCE ANALYSIS

Images	Performance	Method		
		Color Weighted Average Method	Proposed Method	Remarks
Wang Dataset	Average Retrieval Time (Seconds)	0.96	0.85	Almost Equal
	Average Precision (%)	57	70	Big Improvement
	Average Recall (%)	11	14	Big Improvement
ALOT Dataset	Average Retrieval Time (Seconds)	1.25	1.17	Almost Equal
	Average Precision (%)	53	61	Big Improvement
	Average Recall (%)	10	12	Big Improvement



Fig. 7. ALOT image data_set (image from [15])

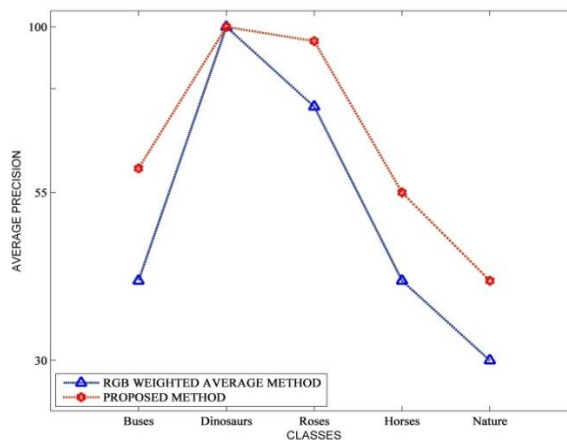


Fig. 8. Average precision of proposed and rgb weighted average method for wang data_set

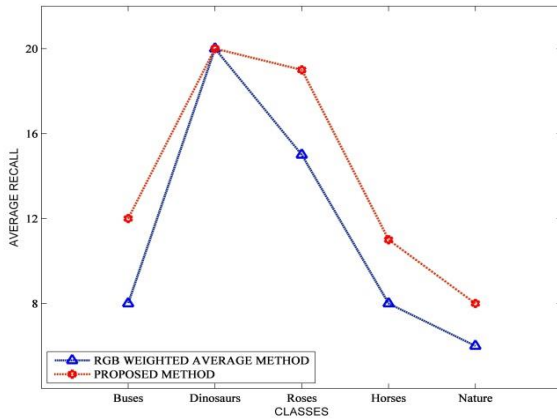


Fig. 9. Average recall of proposed and rgb weighted average method for wang data_set

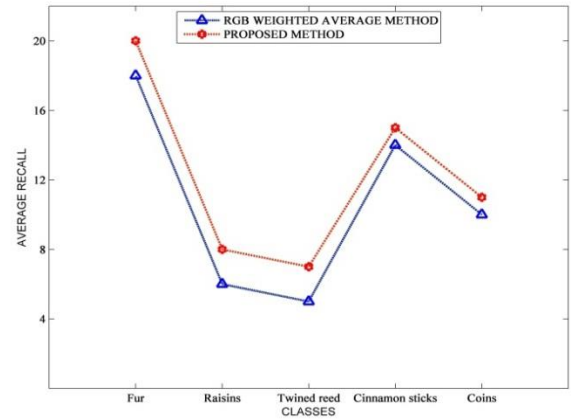


Fig. 12. Average recall of proposed and rgb weighted average method for alot data_set

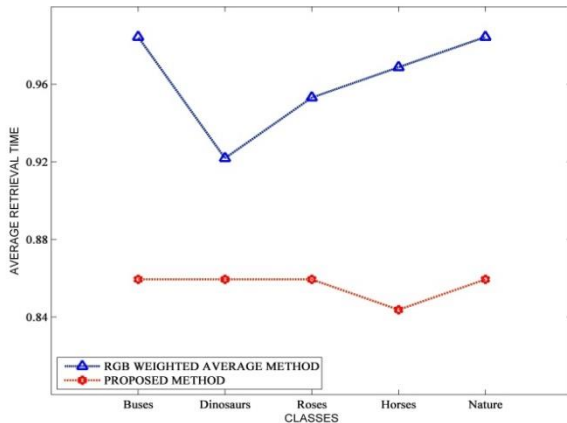


Fig. 10. Average retrieval time of proposed and rgb weighted average method for wang data_set

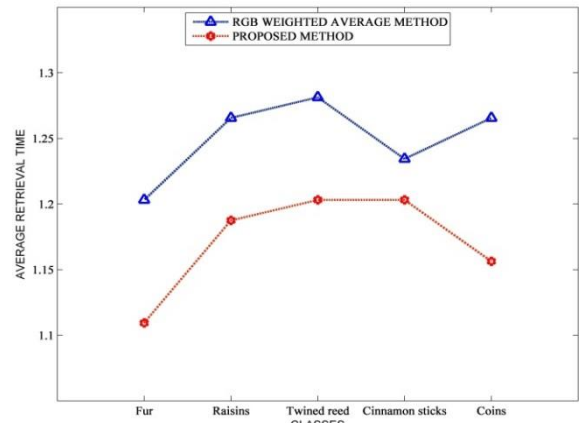


Fig. 13. Average retrieval time of proposed and rgb weighted average method for alot data_set

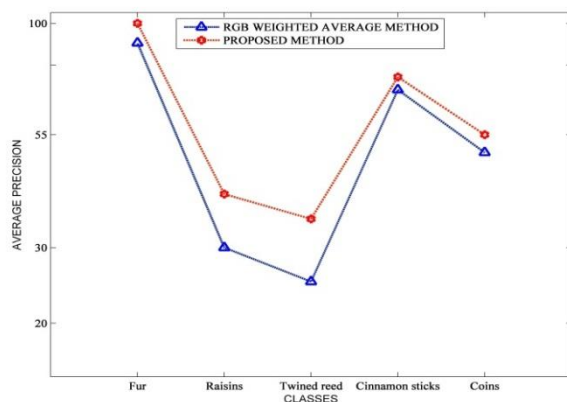


Fig. 11. Average precision of proposed and rgb weighted average method for alot data_set

From these above figures it can be concluded that proposed method is more efficient for color and texture image analysis than color average weighted method. It has improved the accuracy without affecting the efficiency of the CBIR system.

VI. CONCLUSION

Image feature extraction remains a big dilemma for researches in the context of image retrieval since decades, which directly affects the performance of retrieval system.

In this paper, gray scale weighted average method was proposed to reduce the feature vector dimension for increasing the overall throughput of image retrieval system. The performance of CBIR system using proposed approach is quite better than color weighted average method.

VII. FUTURE WORK

We proposed content based image retrieval using gray scale weighted average method for reducing the feature vector dimension as well as for improving the efficiency of retrieval system. The effectiveness of the proposed method was evaluated on two color and texture image datasets where obtained results showed the usefulness of our image retrieval system. Although proposed method works well on texture and color images but still this algorithm can be improved such that it could able to identify color and texture images from hybrid database and then produce refine results on the basis of user preferences. For this purpose, in future we plan to incorporate image clustering scheme.

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3D Virtual Worlds: Business and Learning Opportunities

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Abstract—Virtual worlds (VWs) are rampant and easily accessible to common internet users nowadays. Millions of users are already living their virtual lives in these worlds. Moreover, the number of users is increasing continuously. The purpose of this paper is to review all the business opportunities on these virtual worlds along with the learning opportunities for the real world companies and business students. This paper clearly and precisely defines the virtual worlds in the context of social networking sites and also aims at discussing the past, present and future of VWs. All the possible business opportunities for the real world companies including advertisement & communication, retailing opportunities, application for human resource management, marketing research and organizations' internal process management through virtual worlds are critically reviewed here. In addition to the discussion current learning and training opportunities for the real world companies and business students are also reviewed. The paper aims at proving that the VWs are full of business and marketing applications and they could be widely used by the real world companies for effective and efficient business operations.

Keywords—Virtual Worlds; Social Networking Sites; Virtual Reality; Virtual Education Environments; Virtual Commerce

I. INTRODUCTION

Man is a social animal and he cannot live without society, therefore a social environment around him is imperative to be connected with. In past, man had to go out (Physically) and meet others in order to socialize. However, later, with the technological development man was capable to fulfill his social needs through telephone, internet, online social networking sites (SNS) and online messengers (e.g. MSN and Yahoo) without going out (physically). In the present era of technology world has been transformed from global village to a super global village. VWs are another gift of modern technology for the mankind. These three dimensional VWs are also known as 3D social networks. These 3D VWs provide a simulated real world's environment in which users can do almost all the real world's activities including buying, selling, constructing buildings, dancing, clubbing and even learning and training (Peter, Mark, Aukje and Marcia, 2008).

A large number of users are engaged in these 3D social networks and this population is increasing enormously day by day (Hemp, 2006; Fletcher, 2008; and Melancon, 2011). It is an established fact that the marketer goes where the consumer goes. Therefore, the purpose of this paper is to provide an extensive review of the past studies to sum up all the possible business opportunities within these VWs. Learning and

training opportunities for the real world companies and business students will also be focused. In this paper social networking sites (SNS) are defined and then discussed within the context of 3D virtual worlds. History and future prospects of VWs are also reviewed with a view to finding that if the VWs are going to replace Web 2.0 technology before 2020 (Rawlinson, 2007).

Advertisement opportunities in virtual worlds for real world businesses through flagship stores, buying advertising spaces, sponsoring an event and press release of virtual activities are discussed in detail (Kaplan & Haenlein, 2009). Moreover, virtual worlds are also providing an opportunity for the real world businesses to open retail stores for selling virtual/real products and services or only for promotional objectives (Vrechopoulos, Apostolou & Koutsouris, 2009; Kaplan & Haenlein, 2009). Some other business activities could be done through virtual worlds for instance; application of human resource management, marketing research and internal process management. Educational opportunities are also reviewed in this paper with a view to proving that the real world businesses and business academics can use VWs for learning and training purposes (Boulos, Hetherington & Wheeler, 2007; Eschenbrenner, Nah & Siau, 2008; Graves, 2008; Ondrejka, 2008; Guru & Siau, 2008; and Belei, Noteborn & Ruyter, 2011). Besides the benefits for real world businesses and educational institutes, some risks for them are reviewed.

The next section of the paper provides a detail review of the virtual words, their past, present and future. Furthermore, business and learning opportunities have also been discussed which provide basis to make recommendations and suggestions at the end of the paper.

II. REVIEW OF 3D SOCIAL NETWORKS

A. 3D Social Networks

Boyd and Ellison (2008) articulated that the web-based Social Network Sites (SNS) allow users to create public (open) or semi-public (close) profiles within those networks in which they can get connected with other users and can see, connect or share other users' connections as well. This definition shows three basic characteristics of SNS i.e. (1) making own identity on these sites, (2) connecting with other users and (3) visiting, connecting or sharing other users' connections. In the present era most of the young and middle aged people are busy in interacting and communicating with

one another on internet. For this, they use social networks e.g. Facebook, Twitter, LinkedIn etc however, all of these mentioned sites are 2 dimensional (2D) web based social networks. Fetscherin & Lattemann (2008) claimed that 3 dimensional (3D) virtual worlds have provided users with another medium of communication in which besides simple text chatting, voice over IP is enabled and users can do voice chat as well. Hence, interactivity in virtual worlds is more affluent than that of the other 2D web based social networks.

Virtual world is a new development and it is a type of cyber technology. Since man is a social animal and needs a social environment around him to interact with, Virtual worlds have fulfilled the human socializing needs by enabling users to go out and meet others in a virtual environment through their avatars, as a matter of fact users do not go out physically but just in a virtual environment where they are able to visit different places, cities, clubs, hotels, islands, stores and many other simulated to the real world places (Wyld, 2010). These virtual worlds (e.g. Second Life) have all such characteristics of Social Network Sites as are mentioned by Boyd and Ellison (2008), since 3D virtual worlds allow users to create their own identity separately and uniquely through an avatar, users can make list of connections with other users and can see, connect and share other users' connections as well.

Since virtual worlds and other web 2.0 social network sites share basic characteristics therefore, virtual worlds also lie under the definition of Social Network Sites however, there are many differences between web 2.0 social media and 3D virtual worlds (Kaplan & Haenlein, 2009). A few differences are explained by Kaplan & Haenlein (2009), between basic web 2.0 social network sites and 3D virtual worlds. Authors explained the communication and interaction among the virtual worlds users (also called Residents) as a real time interaction which is not possible on web 2.0 social network sites (e.g. Facebook, Twitter, Bebo and Hi5) where one user posts the content at one time and it is consumed by other users at some different time when they come online. Moreover, web 2.0 social network sites allow users to create their online identity however, it is limited while, on the other side residents of virtual worlds are able to create their fully customized self-identity through avatar creation and in some cases they can even make avatars' faces resembling to their real life faces. Residents of virtual worlds can even create the body of their avatars exactly like their real life body by adjusting the height, skin colour, hair style and body fat etc. Moreover, 3 dimensional environments of the virtual worlds also distinguish them from web 2.0 social network sites. The restriction free environment of the virtual worlds is providing the real world businesses with huge opportunities to market their business into these emerging channels of communication.

B. Virtual Worlds

Virtual worlds also known as metaverses, are 3D online environment where every user has his own Avatar which shows his identity and they are online new media where users, through avatar, can move and communicate (speak and chat). Parmentier and Rolland (2009) stated that the number of users on VWs is increasing continuously. According to a research the number is going double every year (Fetscherin & Lattemann, 2008). (Gartner, 2007) predicted that by the end of

year 2011, 80% internet users would show their presence on virtual worlds. Though the forecasted number of users has not shown its presence but another careful research has asserted that in year 2018 there would be about one billion users of VWs (Renaud & Kane, 2008).

C. Some Important Concepts About Virtual Worlds

Chesebro (1985) determined that any 'definition' should focus on the unique, exceptional and core elements of the defined thing. However, if all the available (till 2008) definitions of the virtual worlds were analyzed, it would be found that they do not fulfill the basic requirements of 'definition' (Bell, 2008). Bell (2008) opines that the term 'virtual world' is defined by the academics and media differently as per their own understanding and requirement. These definitions are somehow contradictory to each other.

Koster (2004) declared *Persistence* and *Numerous Participation* as the essential characteristics of virtual worlds and posited that virtual worlds are virtual spaces where participants represent themselves through avatars. Castronova (2004) determined that these are the computer based environments which could be used by a large number of users simultaneously. If both definitions are analyzed scrupulously, number of dissimilarities will be observed (Bell, 2008). In Koster's (2004) definition of virtual worlds the element of technology was thoroughly missing, though these virtual worlds base entirely on technology and their existence is impossible without it. Castronova (2004) tried to fill this gap but at the same time two characteristics of virtual worlds i.e. *persistence* and *numerous participation* were missing in his definition (Bell, 2008). Bell (2008) further critically reviewed the definition by Castronova (2008) and stated that this definition is pointing towards simple chat rooms and hence difference between virtual worlds and chat rooms becomes vague.

Reviewing the prior definitions and indicating the gap between them Bell (2008) defined virtual worlds more precisely. Bell (2008) defined virtual world as a technology based electronic environment where groups of people can interact through avatars at the same time. This definition of Bell (2008) mentions all the attributes of VWs that were missing in the definitions by Koster (2004) and Castronova (2004). Bell (2008) mentioned the element of people in his definition very clearly which was also not obvious in the previous definitions and without presence of people VWs are just like vacant data warehouses. Schroeder (2008) maintained a similar view to that of Bell (2008). Schroeder (2008) defined virtual worlds as an online social place where people, in great number, go overtime and experience social interaction simultaneously. Schroeder's (2008) definition seems to fulfill the gap from the previous definitions like that of Bell (2008). Schroeder's (2008) definition about VWs contain elements of technology (online), human interaction and simultaneousness.

In an essay Toward a Definition of "Virtual Worlds" written by Mark W. Bell in year 2008 Synchronous, Persistent, Network of People, Represented by Avatars and facilitated by networked computers are the described characteristics of virtual worlds. Bell (2008) argued that these unique characteristics are present in real virtual worlds which

make them different from traditional social networks and video games.

Sivan (2008) also defined virtual worlds from a slightly different angle. Sivan (2008) argued that 'Real Virtual Worlds' must need to be the combination of 3D worlds, Community, Creation and Commerce and that is what he called 3D3C (3D, Community, Creation and Commerce). Researcher further explained these elements and described 3D world as a place where through roving camera users can see a world inside these environments (e.g. avatars, houses, cars, land, sky, sun, wind, gravity, water and fire). Community is defined as a place where man, being a social animal, fulfils his social needs by communicating and interacting with others around him. This sense of communication was removed from our society since man started using technology (emails, SMS etc). However, real virtual words are providing this sense of community where users through avatars communicate and interact with each other (Sivan, 2008). Researcher explained about the 2nd C of 3D3C and argued that real virtual worlds provide users with an opportunity of creation and experimentation through which users can learn and enhance their knowledge and experience. Commerce element of this definition is explained as real virtual worlds provide opportunities to earn money. Users can earn by doing business activities or through employment opportunities (Sivan, 2008). Some of the virtual worlds which can be viewed by 3D3C prism are WOW, IMVU, Penguin, Second Life, Active Worlds, Sony Home and Google Earth. Messinger, Stoulia & Lyons (2008) agreed with Sivan (2008) and argued that social networking (community), creating virtual items (creation) and then selling or buying these items (commerce) is the three main features which make any online 3D world a real virtual world.

D. Present and Future of the Virtual Worlds

Virtual worlds are emerging very rapidly and have been doing so since 1990 (Messinger et al. (2008). Today we have a long list of virtual worlds with us. There were a few users in the beginning; however participation is increasing day by day with the rapid growth in technology, greater broadband access with high speed and lower prices (Messinger et al., 2008). Organizations are considering virtual worlds as fields of opportunities; therefore it is necessary to describe the current situation of the virtual worlds and future predictions about them. In this section, present and future of the virtual worlds is discussed briefly.

The number of virtual world users that are registered has reached around 671 million globally, and their disbursements are approximately around 1.8 billion dollars on virtual assets in the third quarter of 2009 (Kzero, 2009). The interest of people in virtual worlds is increasing globally and most recent figures indicate that over 800 million users have registered their account on virtual worlds (Kzero, 2010). Residents of *Second Life* traded in virtual items worth of 150 million dollars during the third quarter of 2009 (Linden, 2009). This increasing interest has instigated the virtual worlds to focus on various segments such as children, education, adults and special interests of individuals. Daley (2010) stated that VWs will become a necessity for enterprises with various branches, scattered vendors and large numbers of teleworkers.

OECD (2011) revealed that 70% of all the users of virtual worlds are mainly engaged in VW entertainment and games. This percentage should be carefully inferred as virtual world offer many other applications besides games and entertainment. Some of the users of VWs play games and engage in entertainment, whereas others join lectures and simulations, or participate in research. Kids World is the second largest targeted group with approximately 26% and 3% with socializing, and 1% with workspace worlds. Moreover, users who belong to the age group of 11 to 35 year olds spend an average of 20-25 hours per week in Virtual worlds.

Renaud and Kane (2008) stated that there will be one billion individuals who will be using Virtual worlds by 2018. In 2009, there were already more than 90 million individuals in Habbo Hotel, 20 million in Cyworlds and 13 million in SL. Multinationals, for instance, Adidas, BMW, and Vodafone, are already developing commercial activities in virtual worlds (Parmentier and Rolland, 2009).

E. Background and History of Virtual Worlds

According to Nood & Attema (2006) the virtual world is not as novel a concept as it was in the past. It is as old as "Dreaming". There are two kinds of worlds from the beginning: primary world and secondary world (Auden, 1968). The primary world is the world in which a man can feel or see by using his sensory organs, whilst secondary world is the world of a man's imagination. Man always, consciously or unconsciously, imagines many things in his mind. Computerized virtual worlds were initially played as 3D video games only. VWs are also known as Massively Multiplayer Online Games and different terms are being used synonymously, e.g. Massively Multi-Player Online Role Playing Games (MMORPGs), Multi-User Online Virtual Environments (MUEs), and Networked Virtual Environments (NVEs). Instead of using virtual worlds just for entertainment purposes, users are now using them for social interaction amongst one another in their daily routines (Wyld, 2010). The number of users of these worlds is increasing day by day and it seems that Gartner's (2007) prediction will become true. According to DFC intelligence the world's gaming market was \$67 billion in 2012 and it is expected to grow up to \$82 billion in 5 years. In the past, video games were less interactive, single player oriented and users were of a young age. Nonetheless, now virtual worlds are highly interactive, multi-players can interact instantaneously and the users belong to groups of all ages (Adolph, 2011: ITU Telecommunication Standardization Bureau, 2011).

Today's virtual worlds are more developed, interactive and collaborative; therefore, there are hordes of opportunities for different disciplines of life (Barnes and Mattsson, 2008). 3D virtual worlds are commonly categorized into two forms, game oriented and free form virtual worlds (Bainbridge, 2007). Game oriented virtual worlds, e.g. World of War craft, are only used for gaming purposes. Avatars are bound to wear some specific items to play within that environment. In game oriented virtual worlds users usually play with computer-controlled characters and try to win the levels of the game just to get pleasure or entertainment. For this purpose users might have to purchase some virtual items to make him powerful in the game to win the level. Free form of virtual worlds (e.g.

Second Life) are totally different in nature, as users are not there to play games but are free to perform most of their real life activities. Free form VWs are more close to the real world; they are also known as open virtual worlds (Messinger et al., 2008). People buy and sell different kinds of applications, such as virtual apparels for their avatars, lands, islands, virtual vehicles and many more items. They can do many activities within these virtual worlds that are more simulative than the real world.

F. Parents of Virtual Worlds

Sivan (2008) argued that real VWs came into being by the happy marriage of two technology based concepts. Sivan (2008) posited that father concept of the virtual world is 'Virtual Reality' and mother concept is 'Gaming Worlds'. However, these worlds are not based merely on these two concepts but also on economy, sociology, business, law, biology, computer science and mathematics (Sivan, 2008). Messinger et al. (2009) agree with Sivan (2008) and believe that online gaming and social network led to today's virtual worlds.

Sivan (2008) defined virtual reality as a computer based environment which provides *immersion, interaction and imagination* to its users. He further argued that the concept of virtual reality is not new but it has been there since 1962 and was initially used only by the armed forces for training purpose (Sivan, 2008). Sivan (2008) said that if the father of the VW is virtual reality then no doubt the mother is Gaming World. Messinger et al. (2008) also posited that antecedents of the virtual worlds are gaming worlds. Gaming worlds are known to the world since 1978 and the first multi user game was MUD (Multi User Dungeon). MUD is a well-known first multi user game but it was without graphics and was totally based on text (Bartle, 2004). Other well-known gaming worlds are determined below (Bartle, 2004 and Sivan, 2008).

Ultima on-line: It was first 3D graphical gaming world that had registered highest number of users in year 1997. However, it was not free and users had to pay \$9.95 per month.

EverQuest: Founded in year 1999, it was the 1st game which allowed the user to move along the camera inside the game to see around its avatar. However, in past the user was not provided with this option by other games and avatar's eyes used to be fixed. EverQuest provided the sense of social networks where users played and shared with friends making it a unique game.

Sims on-line: Developed in 2002, it was the first game which allowed users to create the contents. It allowed creativity in 3D construction.

Word of Warcraft (WOW): Founded in year 2004, the game has millions of users today and every user is charged \$10 to validate his or her membership. It is a gaming world which not only provides users with gaming opportunities but also with business opportunities.

Second Life: Second Life also known as SL, introduced first time in year 2003, attained immense exposure and media coverage in 2006. This is the 1st gaming world in which users

are provided with all in one i.e. community, creation, and commerce. Such kind of gaming worlds come under free-form of virtual worlds (Bainbridge, 2007).

G. Avatar

In virtual worlds a physical simulated environment is presented through electronic environment where users show their presence by unique characters known as avatars. By means of which, more people interact with one another as well as with the environment. Though an avatar is an electronic presence of the user inside the virtual world, it has much resemblance to a real human being (Melancon, 2011). Taylor (1999) found that users perceive these digital bodies not only as electronic representatives, but also as true bodies controlled through keys under their fingers. Barnes & Mattsson (2008) defined the word avatar and suggested that it is derived from the Sanskrit word 'Avatara' which means 'Descent' and in general it is refer to the representation of an entity (Jin & Sung, 2010).

However, today the word 'Avatar' is being used to describe the virtual presence in the virtual worlds. Second Life, a leading virtual world, was the first to use this word for virtual presence of the users (Lee & Warren, 2007). Each user is free to customize its avatar (social presence) and can shape it to his/her real world's appearance. In addition, it is possible to shape the avatar into any mythical creation. Avatar's each and every part of the body could be customized, e.g. he can increase the height of his avatar to show his real world's appearance. Moreover, skin complexion, eyebrow, hair length, hair style, colour, and chest size etc. could also be customized. In addition, users can show their race and ethnical belonging by customizing their avatar (Jin & Bolebruch, 2009).

Jin and Sung (2010) focused on avatars by marketers' perspective and argued that these 3D avatars in 3D electronic environments could be used by marketers as a marketing communication tool and as a media for building brands in virtual worlds. And it is no more future as real world companies are already using these *Avatars* in virtual worlds for the sake of marketing communication and brand-building (Komiak, Wang & Benbasat, 2005; Holzwarth, Janiszewski & Neumann, 2006; Komiak & Benbasat, 2006; and Wang & Benbasat, 2008). Jin and Sung (2010) further argued that marketers can use avatars in a retail environment to increase its effectiveness.

One of the causes of lack of shoppers trust on online stores was the absence of face to face interaction; however this dispute has been resolved by the presence of avatars in 3D virtual stores (Aldiri, Hobbs, & Qahwaji, 2010; and Alves & Soares, 2013).

Online business organizations already had knowledge that human or some kind of apparent relationship is important to build some meaningful relationship with their customers (Qiu & Benbasat, 2005). Therefore, presence of avatar (face to face interaction) is giving a huge comparative advantage to virtual worlds over traditional web stores. Presence of the Avatar in a 3D virtual reality retail environment also provides greater satisfaction and better attitude towards the product (Holzwarth, 2006).

III. BUSINESS OPPORTUNITIES IN VIRTUAL WORLDS

Computer based virtual worlds are known since 2003 and they are rampant nowadays (Shen & Eder, 2009). As mentioned earlier VWs were initially used for merely the gaming purpose however, lately marketers found huge business opportunities in them. Hemp (2006) agreed and argued that a large number of users is engaged with loads of intangible things in VWs e.g. virtual businesses, learning, social groups where residents keep busy in various social activities that range from dancing parties to Christmas celebrations. Therefore, VWs are providing the real world businesses with a gigantic opportunity to market their business inside virtual worlds (Hemp, 2006). Further, Fletcher (2008) reported that in year 2007 there were 14 million accounts alone in Second Life (one of the leading virtual world) while the number of accounts later hit 21.3 million worldwide (Melancon, 2011), as it is an established fact that the marketer goes where the consumer goes. Furthermore, VWs have become extensively important technological tool for the marketers and advertisers to market their product or service in it (Hemp, 2006). Therefore, in order to chase and target the right consumers, marketers began exploring business opportunities within these unique 3D worlds.

One of the research posited that in year 2020 VWs would be as common as world wide web is today (Rawlinson, 2007), a large number of residents attracted businesses to show their presence in these worlds and to target their potential customers. That is why; IBM purchased a private island in Second Life (SL) and showed its virtual presence in April 2006 to promote its brand among its customers and partners (Fletcher, 2008). The IBM's virtual island consisting of 12-islands complex including healthcare and Code Island, aimed at showing the demo of the products and to present lectures on software engineering. The real world (RL) companies already present in these virtual worlds are IBM, Dell, Sun Microsystems, Nokia, Sony Ericson, Coca Cola, Microsoft, Toyota, Mercedes, Adidas, Reebok, BMW, L'Oreal, and American Apparel etc. These global companies aim at increasing their brand awareness in VWs to increase the purchase intention of customers in the real world.

A. How Can Real World Businesses Use Virtual Worlds as a Communication Channel?

The significance of this emerging channel of virtual worlds has been discussed by one of the researches on interactive communication channels (Dad, 2012). Research has pointed out that though the academic researchers did not attach substantial value to this channel, the big brands like Toyota and Reebok have already set off their virtual presence in these worlds. It is thus highly recommended to investigate and explore these VWs in context of their importance for real world businesses and to know that how this emerging communication channel could be used for marketing, branding and advertising the real world products and services (Dad, 2010).

Real world businesses can widely benefit from these VWs especially for advertisement purposes. Kaplan & Haenlein (2009) stated that virtual worlds could be used in four different ways by real world businesses to advertise their

products or services. These four ways of advertising and communication business are briefly explained below:

- **Flagship Stores:** Virtual presence of Real world companies in VWs can be shown by setting up their virtual flagship stores. In these virtual flagship stores companies can create, sell or test their real world products by the provision of real world's resembled products. One of the big brands from the automotive industry; Toyota already has a virtual store in one of the well-known virtual worlds i.e. Second Life. In past Toyota had launched the virtual edition of its real world Scion xB model for advertising and testing purpose.
- **Buying Advertising Spaces:** VWs can be used by the Real world companies for advertising purpose through buying advertising space in virtual malls in the same way as they do in the real world or online web sites e.g. Banners and Billboards etc. There are many companies (e.g. MetaAdverse) working inside the VWs as advertising agencies who buy and sell advertising spaces for the real world businesses and also manage ads for them hence earning huge money. MetaAdverse makes money by letting virtual spaces for advertisement to the real world companies. The real world companies are charged by this company for advertising their brands in VWs. Not only does MetaAdverse advertise the real world's businesses within VWs but also tracks those advertisements to view and provide information regarding their effectiveness. Canada IMAX used this way of communication to advertise the Harry Potter Saga. This way of advertising is proven by research to be cost effective and cheaper as compared to the traditional online advertising on traditional websites.
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- **Sponsoring an Event:** As mentioned earlier in this paper, the VWs are used for variety of social events i.e. dance parties to religious practices like Christmas and

Haji Pilgrimage. Therefore, these events can be sponsored by the real world companies for promoting their real world brands. This method of promotion has already been practiced by one of the British newspapers; Guardian, when it sponsored *Secondfest* i.e. a music festival inside the Second Life.

B. Press Release of Virtual Activities

Companies can also give the press coverage to their activities in virtual worlds. Its positive impact has already been proven in research.

There are four different modes of advertising in virtual worlds specifically in SL (Barnes, 2007). These modes include placing a 3D object as Reebok did by placing virtual shoes in its virtual retail store in Second Life (SL), advertising through virtual static billboards, playing ads through multimedia in virtual worlds and through cross-promotion by dancing and camping in virtual malls and casinos etc.

Advertising is always considered crucial for e-marketing promotion (Hemp, 2006; and Sharma, Bakou & Lijuan, 2012). Other researches also explained the pivotal role of advertisement and promotion for any business and posited that virtual worlds, especially Second Life, could be considered a very favourable interactive communication channel (Hedley, 2006; Sharma et al., 2012; Dad, 2012; Messinger et al., 2009; and Hassouneh & Brengman, 2011)

For companies, one of the major attractions to use virtual worlds for E-marketing is promotion and branding (Hemp, 2006; Messinger et al., 2009; Dad, 2012; and Sharma et al., 2012). These aforementioned researches have explored the VWs and the business opportunities within these worlds. It is found that advertisement is one of the major benefits that companies can enjoy by using VWs. Since Communication plays imperative role in the success of any business therefore, VWs provide the residents with a unique and highly flexible platform for the interactive communication in addition to a new channel for the effective two sided communication between companies and potential/current customers. Word of mouth has always been an effective form of advertisement therefore; companies can use VWs for brand promotion through word of mouth and blogging.

VWs can also be used by the Real world companies for advertisement and promotion of their brands and vast range of products or services in variety of ways. Some useful means to fulfill the promotional needs of real world companies through virtual worlds are discussed below:

- Real world companies can create their virtual stores or islands within these VWs to promote their brands and vast range of products. Many real worlds' big brands have already shown their presence in these 3D VWs e.g. Dell, Toyota, Adidas etc.

- Since VWs do not have any geographic boundaries so; customers from all around the world could be gathered at one specific place in them where companies can communicate with their potential and current customers simultaneously. This was something impossible without the existence of VWs. Companies can promote their newly launched product or get

feedback for their existing products or give virtual product to their customers for test marketing.

- Virtual worlds are much synchronous therefore; an effective communication could be expected between companies and customers.

- A large number of populations could be targeted through virtual worlds therefore; these worlds could be considered as a new interactive communication channel and an alternative to TV and Web advertisements.

- As VWs are cost effective and most interactive communication channel therefore; Small and Medium Enterprises (SME) can benefit immensely by targeting and persuading large number of audience at one time with comparatively meager cost.

- SMEs can also use virtual magazines and newspapers as an alternative to the real world print media. In VWs an ad which normally reaches to 25,000 SL residents on virtual magazines or newspapers costs about \$11 (Second Style, 2008).

C. Retailing in Virtual Worlds

Virtual reality retail stores gave a new concept to the retail industry. Virtual reality retail (VRR) stores are the stores present within these 3D VWs, some of which are freebies and some that are really costly. The virtual world has provided businesses with a new opportunity to market and sell their products. Virtual worlds (e.g. Second Life) are offering an alternative, improved, and quite potential medium to the consumers where they can shop for their virtual lives by paying with virtual money (e.g. Linden Dollars in Second Life). In the same way consumers can also buy for their real lives, but this is just at an introductory stage at the moment (Vrechopoulos et al., 2009).

Unlike web retail stores where many components of traditional retail environments are absent, e.g. Social Factor, the VRR stores that are made up of computer graphics provide a real world simulated environment. VRR stores, like real world stores, are constructed with walls, colours, lighting, in-store music, floors, shelves, layout and design. One of the components of a retail store's environment is social presence, i.e. presence of other customers or employees.

VWs are providing real world retailers with a unique opportunity to set up their retail business inside these virtual stores (Kaplan & Haenlein, 2009), but the concept of retailing in VWs is quite different to that of traditional web 2.0 technology. If traditional online stores are analysed, many discrepancies will be found, for instance the products' images placed on traditional online stores are not true representation of the real product (Keeney, 1999). Moreover, whilst visiting traditional online stores there is a feeling of loneliness and inadequate interaction with other customers (Wang, Baker, Wagner & Wakefield, 2007). This is not the case with VWs as they provide the customers with a real environment. The retail stores are built with 3D electronic environment where all the products are 3D electronic objects that closely resemble the real world product. Moreover, the visitors of VRR stores can also interact with one another (Kaplan and Haenlein, 2009).

Virtual reality retail stores are the most appropriate representation of the real world stores, which could enhance a company's branding and advertising campaigns. It has also been proven through previous studies that this 3D object placement in VVs has a very positive impact on users' intentions to purchase the same product in real life (Schlosser, 2003 and 2006).

Kukreja and Humphreys (2014) argued that in traditional online stores, goods or services were shown on 2 dimensional flat interfaces where shoppers were not able to see a 3D view of the product. Moving inside the traditional web stores was known as 'scrolling down or up'. Kukreja and Humphreys (2014) determined that 3D virtual reality retail stores are the substitute of 2D web stores. They are far better than web store and shoppers can move around the stores by walking, flying or running with the help of their avatars. There is no more navigational difficulty in 3D VRR stores which shoppers were facing in traditional web retailing (Kukreja and Humphreys, 2014).

D. Applications of Human Resource Management (HRM) in Virtual Worlds

Kaplan and Haenlein (2009) have discussed in detail that the application of Human Resource Management (HRM) in VVs is possible along with marketing implications. *TMP Worldwide Advertising & Communications* is a service provider company that is already involved in recruiting employees from virtual worlds. Recruitment from VVs could be more helpful for technology oriented firms for instance T-Mobile and eBay because in this way they can have more technology oriented employees.

However, in many cases such recruitment could not be very effective rather many risks are associated with it. A few risks are mentioned below:

- Companies could not generate a large number of populations in VVs as it is a new technology and only a few people are familiar with it.
- Applicants like to be contacted in a more traditional way and they might take virtual recruitment messages as a spam.
- There is a risk of fraud for the companies as they cannot see the real person behind the avatar.

Besides many risks companies can still use VVs for recruitment process. It is recommended that companies should not conduct interviews in VVs though they can post advertisements for vacant positions at any relevant event in virtual worlds.

E. Marketing Research in Virtual Worlds

Another interesting application of the virtual worlds for real world businesses is to use them for marketing research. There are two major reasons to use VVs for marketing research. It costs very low as compared to the marketing research in real world (Kaplan and Haenlein, 2009). Secondly, a large number of consumers ready to be queried, analyzed and understood could be approached at a single platform

(Messinger et al., 2009). 33% reduction of cost while conducting research in virtual worlds is already proven (Kaplan & Haenlein, 2009). Companies are currently researching in VVs to observe the attitude, motivation and behaviour of the residents.

Companies are also researching in VVs for product testing and then modification on the basis of feedbacks they receive (Messinger et al., 2009). For instance, Toyota got its virtual island in Second Life where it sells virtual Toyota car for 300 Linden Dollars which costs about 1 real U.S dollar to the customer after which the customer drives and shows it off to his/her virtual friends and then customizes that car as per his/her choice and need. The customer then gives feedback to real life Toyota engineers. Later in compliance with the customers' feedback the company modifies the real world's model. Another example of test marketing is discussed by Kaplan and Haenlein (2009) when *Starwood Hotels & Resorts*, a real world hospitality company, decided to build a new building for their Hotel in real world, it first built the virtual building of the proposed design in Second Life and then invited Second Life residents to visit it and give feedback. Later, the company made many modifications in its real world's design on the basis of resident's feedback in Second Life.

Hence real world companies are highly recommended to use VVs for market research, testing and innovation.

F. Internal Process Management

Kaplan & Haenlein (2009) posited that virtual worlds can be used to manage the internal processes of companies e.g. virtual meeting with employees & business partners, and also for knowledge sharing. Cases of Cisco, IBM and Crown Plaza Hotel chain should be analyzed by the real world businesses to review how these companies managed their internal processes through virtual worlds (Kaplan & Haenlein, 2009). Cisco has built two virtual islands in Second Life where its employees are provided with an opportunity for formal meetings. Moreover, Cisco has also provided its employees with virtual code of conduct. Similarly, IBM is managing its internal processes through Second Life. It has twenty four islands inside this virtual world. One of the greatest real world hotel chains known as Crown Plaza Hotel has also established its virtual presence in Second Life where it offers customers to book the meeting rooms in virtual Crown Plaza hotel in exactly the same way as they do in real world in United States, England and Switzerland.

These examples could be the benchmark for other real world companies for they can also manage their internal organizational processes through VVs in almost no time at very cheap cost. If a company's CEO has to meet his partners, with the help of VVs he needs not to travel all the way around the world wasting money and time. He can meet his partners in VVs with very effective time and cost. Real world companies are highly recommended to review the above mentioned cases and adopt virtual world technology as a core business application to reduce the cost and time for internal processes management.

G. Risk and Challenges for Real World's Businesses in virtual Worlds

Along with many benefits there are certain risks which companies might face while setting up their business in VWs. Some of the risks and challenges for real world companies in virtual worlds discussed by the academic researchers (Hemp, 2006; Kohler, Matzler, & Fuller, 2009;) are mentioned below.

- There could be some technological constraints for all the organizations considering to show their presence in VWs but do not have up to date computer systems. It is evident that VWs need updated hardware and software to run. So, it could be a big challenge for organizations to jump in virtual worlds before updating their computer systems.

- Virtual worlds have some technological issues like sudden breakdown while using them which could cause data loss for the companies. However, VWs are getting more sophisticated day by day and this problem does not seem to be faced by users in near future.

- Another risk for the companies in virtual worlds could be the recruitment of an employee whom they do not know at all in real world, who could be any one else behind his or her avatar and thus could damage company's brand name in virtual worlds.

- Companies cannot sign a contract within VWs as law codes are not yet completely formulated and implemented.

IV. LEARNING AND TRAINING POSSIBILITIES IN VIRTUAL WORLDS

Porter, Weisenford & Smith (2012) mentioned that in a few recent years many companies have joined virtual worlds to train their employees as VWs provide them with a very cost effective environment for training. Virtual worlds provide a simulated real world's environment where users can do almost all the real world activities including buying, selling, constructing buildings, dancing, clubbing and even learning and training. Many universities and institutions including Harvard University have already built virtual campus inside these metaverses.

Besides many other challenging objectives engagement, interactivity, experimentation and idea generation remain the most challenging for the education sector and even these objectives became more challenging and imperative after the introduction of online course formats (Brenda, Nah & Siau, 2008). Though to meet these challenges, instructors from education sector have already used, analyzed and examined Wikis and Blogs (Guru and Siau, 2008) but these technologies were not as interactive as 3D virtual worlds (Brenda et al., 2008). In 3D virtual worlds residents are free to interact and engage with one another; moreover they are enabled to create any kind of space they imagine, hence they can perform many experimental activities (Brenda et al., 2008).

Virtual worlds are not fruitful for businesses merely but also for many other fields like hospitals, armed forces and educational sectors. Researchers have explored the usefulness of virtual worlds in educating students through modern and innovative ways (Shen and Eder, 2009; Wagner, 2008; Sarah,

2009; and Belei et al., 2011), distance learning and also for medical and health education (Boulos et al., 2007). In VWs students can go in virtual class rooms where they can interact with one another via text and voice chat options; moreover, they can share and submit assignments using file transfer option. It has been proven through research that VWs could possibly be used by medical, health trainers and libraries in novel ways (Boulos et al., 2007). If used for education and training purposes, VWs would be highly beneficial. Some of the advantages are as follows (Brenda et al., 2008):

- VWs provide the students with a risk free environment for learning where they can share knowledge at very low cost.

- VWs provide the residents with enhanced collaboration and communication where not only can they communicate in real time but also experience a sense of real presence (through avatars) which was absent in traditional internet based learning.

- VWs have not only been credited for providing high engagement where users can learn more with less mental efforts but also for enhancing interest and motivation towards learning.

- Moreover, in VWs institutes can arrange visits of students to all those environments and situations not possible to be visited in real environment, such as a place already vanished in a disaster or a situation perilous to be visited or experienced in real life.

At first IT converted Stone Age world into global village and now this dramatic development and change in IT and internet has transformed this global village into a super global village. Now companies do not base merely in a single town, city or country but they spread their business all around the globe. At present though companies have employees from all around the world belonging to different cultures and countries but still they want to train them all at equal standards. Porter et al. (2012) discussed Veteran Health Administration's (VHA) *Disaster Emergency Medical Personnel System* (DEMPS) whose employees are geographically scattered however, DEMPS needs to treat and train all the employees equally so that they could perform unanimously as a team moreover; organization also needs its employees to be trained for some real situations that could not be created unless a real disaster happens. Therefore, Porter et al. (2012) examined virtual worlds for two years to test whether these metaverses provide an ideal required environment where DEMPS can train all its employees even if they are geographically scattered. Researchers undertook a study and found that virtual worlds provide an environment which ideally addresses all the challenges, since the employees can get together at one virtual place where they can communicate, collaborate and learn at same standards and patterns, besides it is possible to create variety of scenarios and settings where employees can experience and learn how to handle such situations if confronted with in real world.

A. Virtual Worlds in Education

Virtual worlds are not only providing opportunities for businesses but also to the academics and students to fulfill their learning and teaching needs (Eschenbrenner et al., 2008;

Boulos et al., 2007; Belei et al., 2011; and Shen & Eder 2009). Academics have always been trying to achieve engagement, interaction, collaboration, experimentation and idea generation as common objectives but it remained challenging to achieve all of them simultaneously (Eschenbrenner et al. 2008). It became rather more difficult after the courses moved to online formats. However, many academics tried to overcome these challenges by means of different online technologies such as Blackboards, Wikis and blogs (Guru and Siau, 2008) but still there are many barriers in achieving these objectives by using the above mentioned technologies.

However Eschenbrenner et al. (2008) noticed that these fore mentioned common objectives could be achieved online by using 3D virtual world technology. Guru and Siau (2008) supported virtual world as a potential place for learning as many users can simultaneously log in from different locations of the world and interact with one another at same time. Researchers further argued that these virtual worlds also provide their users with an opportunity to create a customized environment which fulfils their demands of a particular course work (Eschenbrenner et al., 2008). In these VWs either simulating real world's environment could be created to interact and explore, or an entirely new environment could be built to achieve engagement, interaction, collaboration, experimentation and idea generation objectives (Eschenbrenner et al., 2008). Researchers additionally motivated all the users to visit the *Second Life Education Wiki* and the *Second Life education (SLED) Listserv* to get to know about the educational opportunities in the Second Life (SL). Guru and Siau (2008) agreeing with Eschenbrenner et al. (2008) posited that schools and colleges can go beyond the conversational mode of teaching by using virtual worlds where students could learn more effectively by enhancing their practical experiences within these VWs.

Wagner (2008) also wrote about the learning and teaching experience in virtual worlds especially in Second Life (SL). He arranged a course called '*Virtual organization and global teamwork*' in Second Life and assigned a task to each group of 5 students. Each group was given U.S \$12 i.e. equal to 3,000 Linden dollars and then was asked to build virtual companies and generate the revenue. Wagner (2008) argued that teaching and learning experience of both students and teacher was more than expected. Students produced interesting work through this task which demonstrates their high level of explicit and tacit learning. Moreover, they also generated revenue through their virtual activities on second life though the generated revenue was not enough, as the exchange rate between Linden Dollars and U.S dollars is not that high and on top of it that activity was just for short span of time. Wagner (2008) had a great teaching-learning experience in virtual world thus he concluded that these worlds are offering a unique platform where teaching-learning objectives could be achieved effectively. Moreover, virtual worlds are offering a unique environment which is potential for multi-disciplinary courses. Wagner's (2008) study confirms Guru and Siau's (2008) study in which they proclaimed that VWs could be a source to improve the quality of learning experiences. Guru and Siau (2008) posited that the users' perception of control leads them to get engaged within the environment more easily therefore,

virtual worlds are more suitable places to get students and instructor involved in it to practice their theoretical knowledge into practical experience.

Along with Wagner (2008), Eschenbrenner et al. (2008) also mentioned some of the opportunities on hand for education sector in Second Life. Eschenbrenner et al. (2008) opined that *simulations* and *visualizations* that are not feasible in reality; are the top ranked capabilities of virtual worlds which could be integrated to fulfill educational purposes. Moreover, virtual world environment fosters innovation and real time communication which encourages educational opportunities within these worlds (Eschenbrenner et al., 2008). Researchers further cited Second Life (2008) as the most suitable 3D world for educational activities. Eschenbrenner et al. (2008) referred to some major features of Second Life which could foster educational activities in it. These features include distance learning facilities, simulation along with interaction and visualization opportunities. In addition, SL could be used for virtual training, seminars and conferences. Virtual worlds could also be used as land of experiments where new ideas could be tested and security could also be assured by purchasing private islands for experimentation purposes (Second Life, 2008 cited in Eschenbrenner et al. 2008). Boulos, et al. (2007) also approved the Eschenbrenner et al. (2008) investigation about the potential of Second Life for educational purposes and argued that many virtual conferences have already been held or going to be held in Second Life. The above quoted examples evidently prove that virtual worlds and especially free from virtual worlds like Second Life could provide another platform for learning.

Shen & Eder (2009), like the fore mentioned researchers, investigated these immersive virtual worlds and tried to explore their potentiality for educating students. Their research demonstrated that along with many other activities virtual worlds are providing a useful platform for learning. In this study researchers aimed at investigating the students' intention of using VWs for fulfillment of their educational needs. Shen & Eder (2009) tested Technology Acceptance Model (TAM) and tried to find out the intention of those students who are already using Second Life in MIS courses. The research determined that if students are well expert in computing and have computer playfulness, their '*perceived ease of use*' fosters their intention to use Second Life.

B. Use of Virtual Worlds in Marketing Courses

The above discussion about uses of VWs for education purposes demonstrates that the virtual world is useful only for the students studying computing, information system or gaming but not for the business students. It poses as if merely computing and information system students need their theoretical principles to be applied in practical field. Guru & Siau (2008) claimed that possibility of gaming in virtual world is not an end whereas, this is just a beginning and further development in the VWs would reconceptualise and restructure our entire education system.

It is true that for many decades' business studies especially marketing studies have been focusing on theoretical principles only but ignoring the practical experiences and the development of all those real skills which students would need

imperatively when they start their career (Belei et al., 2011). Elam and Spotts (2004) argued that it is truly difficult to simulate real marketing situation through textbooks and class room lectures only and this need is really important to be fulfilled for all marketing courses especially for advertisement and brand management.

Belei et al. (2011) argued that though the lecturers have been trying to fulfill this need of students by assigning those projects but still a huge gap between theory and practice persists. Researchers, after careful analysis of virtual worlds, argued that virtual worlds could be used to enhance students' experience, set up a real scenario and provide countless activities for experimental learning. Further arguing researchers determined that using virtual world is an innovative solution to integrate theory and practical experience (Belei et al., 2011).

As virtual world is suitable place for experiments and wide range of experiences therefore, researchers took benefit from it by assigning branding project to undergraduate students and asking them to develop and practice the branding strategies of their particular brand (Belei et al., 2011). Belei et al. (2011) claimed that through this activity students learned how to practice theory into real market. Though use of virtual worlds for academic curricula still has some limitations but overall it could be much useful and beneficial to both students and teachers. At the end Belei et al. (2011) concluded that virtual worlds are useful for business students to enhance their practical knowledge especially for branding and advertising students.

C. Some Other Examples of Learning Experiences in Virtual Worlds

As mentioned earlier technology and internet have achieved that level of sophistication and capability which has changed the whole human cognition process (Haeckel, 1988). Educational institutes are also trying hard to synchronize with this level of technological advancement, therefore constantly exploring and adopting new technologies to get the classrooms online (Erickson and Siau, 2003).

Boulos et al. (2007) discussed some examples of training and educating in Second Life. Researchers argued that there are number of medical and health education projects going on in Second Life. Some of the examples discussed by researchers are presented here.

Ohio University Nutrition Game: Ohio University started a training project in Second Life to educate visitors about the fast foods and their impacts on health. This project was named as 'Nutrition Game' in which users were to choose different eating styles for their avatars and they got more points if food caused good impact on the health of their avatars and less or minus points if the eating style did the contrary job. The main objective of this project was to train users to choose the healthy diet plan.

Healthinfo Island: Another example of medical training and education on Second Life discussed by Boulos et al. (2007) is about *Healthinfo Island*. This island was solely funded by US National Library of Medicine, also known as NLM, with US \$40,000. Its sole aim was to provide the users

with all the required information about health. This project was started in collaboration with Alliance Library System (ALS), the University of Illinois Library of the Health Sciences-Peoria, the Centre Groningen in the Netherlands and TAP information Services. During the project one to one information, health advice and support was provided to the Second Life residents.

Virtual Neurological Education Centre (VNEC): The institutes of UK have also started considering virtual worlds for training and education purposes. Virtual Neurological Education Centre is a virtual education centre developed by University of Plymouth, UK in virtual world whose purpose was to provide the VW residents with information about the neurological diseases, disabilities and their symptoms. Moreover, it aimed at providing the patients suffering from neurological disabilities with a unique platform where they could get support, information and psychoanalysis trainings.

D. Benefits of Using Virtual Worlds in Education

If educators adopt VWs into their education systems, many benefits are expected by the researchers. Some of the potential benefits of adopting VWs for education and training purpose are discussed henceforth.

- VWs can play an important role in distance learning system (Boulos et al. (2007).
- Boulos et al. (2007) argued that students could be educated through VWs' game-based learning system which does not cause boredom and keeps them active as well as interactive.
- Many experiments and activities can be done in virtual worlds in almost risk free-environment (Eschenbrenner et al., 2008 and Graves, 2008).
- Students can develop sense of shared learning which gives students more knowledge and confidence about asking questions (Ondrejka, 2008).
- Students can achieve new dimensions of innovations and creativity by having learning experience in Virtual Worlds (Goral, 2008).
- Conway (2007) argued that VWs can bring attractiveness in online courses for all the interested students since they would be interacting with avatar which gives the sense of presence of a teacher.
- Boulos et al. (2007) determined that in virtual world students can communicate in real time by means of text and voice chat, they can also see one another in avatar shape which increases their sense of presence in real environment.
- VWs could be very beneficial for old and disabled people who can not move easily from their place but are curious to quench their thirst for knowledge (Boulos et al., 2007).
- Mikropoulos (2001) administered an experiment in which it was found that participants put more efforts and show less engagement and learning while performing a task in real life however, it is totally

opposite in virtual reality worlds where they showed more engagement, gained more knowledge with less efforts and less mental stress. Eschenbrenner et al. (2008) also argued that more attention and engagement could be achieved while performing task in virtual worlds as compared to doing the same in real world.

- Eschenbrenner et al. (2008) argued that by using VWs teachers can teach the courses into a different environment than that of the classroom moreover; they can offer the students a visit to those simulated places which no more exist in the real world e.g. a place that has already been destroyed by a natural calamity.

E. Issues and Challenges to Use Virtual Worlds in Education

Along with the benefits of using VWs for education and training purposes there are some challenges and issues too. A few of these issues and challenges are discussed below.

- It is difficult to determine in which scenario VWs can be of more value to the learners than the traditional education system (Mantovani, Castelnovo, Gaggioli and Riva, 2003).
- It is difficult as yet for the educators to determine how these VWs could be utilized in making their education system more effective (Mantovani et al., 2003).
- One of the issues faced by the students in Wagner's (2008) study was the repetitive crash down of system during lab experiments.
- Mantovani et al. (2003) and Dickey (2005) mentioned *cost* as an issue in adopting virtual worlds for educational purposes. Whenever new technology is considered to be adopted in education sector the main issue has always been the cost issue (Schultze, Hiltz, Nardi, Rennecker and Stucky, 2008) therefore, Eschenbrenner et al. (2008) argued that it could be another issue for the educational institutes to adopt VWs for training and education purpose. Though VWs could be used freely but for experiments and virtual campuses institutes need islands. So the issue could be the cost of purchasing a virtual island or the cost of maintaining an island (Eschenbrenner et al., 2008).
- Non-verbal communication has always been important for human beings, Dickey (2005) argued that it is imperative in traditional classroom environment for students and teachers however, Graves (2008) determined that though avatars present some body language but that does not match with chat or voice so it is just meaningless therefore, virtual classrooms also check the learning process owing to an absence of non-verbal communication which surely exists in traditional classrooms.
- Eschenbrenner et al. (2008) determined that high system performance and high bandwidth power is required to run these virtual worlds however, it is not necessary that every student could have these facilities in his/her home.

- Graves (2008) put forward that keeping an eye on virtual world residents is difficult and it is quite possible that students who use it for educational purposes could deviate from their track and get involved in disruptive and unethical activities. Researcher further explained that it happened with the students of Ohio University and Woodbury University as they were found engaged in virtual shooting and disruptive and hostile activities.
- Eschenbrenner et al. (2008) argued that all those students who are not expert in using virtual worlds can become a nuisance for the teachers by prolonging the scheduled period for the project.
- Wagner (2008) mentioned that the cost issue is not merely a problem for the universities in running projects in VWs but also for students. Though virtual worlds are free to use but if the students have to build something there, which is part of their assigned class project, they have to pay for it.
- Guru and Siau (2008) highlighted another challenge in adopting virtual worlds for education and training. They argued that there are many virtual worlds present today therefore it is necessary for educators to review all these worlds and decide which one should be adopted and at which stage to full fill their objectives.

V. RECOMMENDATIONS

As a result of this review paper, there are many recommendations for the real world companies, academics and students from business schools around the world. All the recommendations and suggestions derived from this paper are given below:

- All those real world companies who are using interactive communication channels must consider virtual worlds as new interactive media for advertising (Dad, 2012) since these VWs are described as much cheaper advertising media than the other web based interactive communication channels (Kaplan and Haenlein, 2009).
- Companies should set up virtual flagship stores in VWs for promotion by sponsoring an event in virtual world or through press release of their activities in VWs.
- Companies can use these virtual worlds as a new point of sale where they can sell their products through retail stores.
- Virtual worlds are also highly recommended for service industry where many consultancy companies can set their virtual offices for consultancy. Through these worlds companies can target customers from different countries at a very low cost.
- The tourist companies are also recommended to promote different destinations of the world by making 3D cities on virtual islands simulated to the real world tourists' destinations.

- All the real world companies investing a huge amount on their Research & Development departments should focus VWs as a place of innovation and product testing.
- As mentioned earlier in literature section, Small and Medium Enterprises (SME) can immensely benefit from VWs as a cost effective and most interactive communication channel.
- SMEs should utilize virtual magazines and virtual newspaper as an alternative to the real world's print media.
- Though in literature section, application of human resource management has also been reviewed but still companies are recommended to be very careful while recruiting through VWs because employers cannot see behind the avatar and they could be deceived.
- Real world companies can use virtual worlds as a place for market research and product testing in the same way as Toyota and Starwood Hotels & Resorts had done (Messinger et al., 2009; and Kaplan & Haenlein, 2009).
- Real world Companies should consider virtual world companies for internal process management where they can meet their partners and employees living in the other corner of the world. It would be very cost effective for them.
- All those companies who want to jump into the virtual worlds are highly recommended to consider their organizational technological structure as virtual worlds need highly upgraded computer systems to run smoothly.
- Companies should also be very careful while making any contract within these worlds as they are not protected by real world's law as yet.
- Virtual worlds are highly recommended for marketing academics and they should assign projects to their marketing students within these worlds.

VI. LIMITATIONS AND FUTURE RESEARCH DIRECTIONS

Though this review paper is very helpful for companies and academics but it is not free from limitations. One of the major limitations of this research is the fact that it is based on literature review of the past study and no parallel original research has been done. Moreover, different business opportunities are discussed in this short paper instead of focusing on one single opportunity. There are many virtual worlds today but in this paper main focus was Second Life only which is another limitation of this research.

Therefore, it is recommended first to work on these research limitations in future studies. Original research should be done instead of reviewing literature only. And one business opportunity in virtual worlds should be discussed and explored in detail instead of discussing many business opportunities in one short paper. Researchers are also recommended to explore virtual worlds in the context of any specific industry as Dad

(2012) did by exploring interactive communication channels in the context of FMCG industry in United Kingdom; therefore, future researchers should also explore advertising activities in virtual worlds in the context of FMCG or any other industry. In this way a more focused outcome could be achieved which would be more useful for the specific practitioners or academics. Last but not the least, other virtual worlds should also be explored along with Second Life.

VII. CONCLUSION

Surely, marketer goes where consumer goes and it is proven in the literature reviewed above that the population of the virtual world is growing day by day and it is posited that in year 2020 virtual worlds would be used in the same way as web 2.0 is being used today. In the above review it is proven that virtual worlds are kind of social networks and they got opportunities for real world businesses to use them as a new interactive communication channel which is more cost effective than the traditional interactive communication channels i.e. websites etc. Moreover, these virtual worlds are also very fertile for the application of HRM, marketing research and internal process management as they provide very productive platform for experiments, research & development, product testing and innovation therefore, these worlds could facilitate real world businesses and academics simultaneously. Academics and students can also use virtual worlds for multi purposes especially for distance learning and conducting experiments. Marketing academics can assign projects to their students on virtual worlds which would be definitely very cost effective and enhancing learning experience as virtual worlds provide a simulated real environment.

Along with many benefits virtual worlds pose some risks too for businesses and academics which they should consider before jumping into these worlds. However, overall this review paper shows that virtual worlds are very useful and beneficial for both academics and businesses and risks are ignorable as compared to the advantages therefore, the academics and businesses are highly recommended to use virtual worlds performing their academic and commerce activities.

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A Unified Forensic Framework for Data Identification and Collection in Mobile Cloud Social Network Applications

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Abstract—Mobile Cloud Computing (MCC) is the emerging and well accepted concept that significantly removes the constraints of mobile devices in terms of storage and computing capabilities and improves productivity, enhances performance, saves energy, and elevates user experience. The consolidation of cloud computing, wireless communication infrastructure, portable computing devices, location- based services, and mobile web has led to the inauguration of novel computing model. The Mobile social networks and cloud computing technology have gained rapid and intensive attention in recent years because of its numerous available benefits. Despite being an advanced technology to communicate and socialize with friends, the diverse and anonymous nature of mobile cloud social networking applications makes them very vulnerable to crimes and illegal activities. On considering the point of mobile cloud computing benefits, the forensic assistance based mobile cloud computing could offer a solution to the problem of social networking applications. Therefore, this work proposes a Mobile Cloud Forensic Framework (MCFF) to facilitate forensic investigation in social networking applications. The MCFF comprises of two components such as the forensic logging module and the forensic investigation process. The forensic logging module is a readiness component that is installed both the device and on the cloud. The ClouDroid Inspector (CDI) tool uses of the record traced by forensic logging module and conduct the investigation in both the mobile and the cloud. The MCFF identifies and collects the automated synchronized copies of data on both the mobile and cloud environment to prove and establish the use of cloud service via Smartphones.

Keywords—Mobile cloud computing; forensics; mobile cloud forensics; social networking applications

I. INTRODUCTION

The skyrocketed explosion of mobile applications and the support of cloud computing for a variety of services for the mobile users has motivated to integrate the cloud computing into the mobile environment. The Mobile Cloud computing introduces new types of high scalability and low-cost on-demand services and benefits for the mobile users to experience the full advantage of cloud computing [1] [2].

The benefits of MCC are the extended battery lifetime, improved data storage capacity and processing power, and more reliability. Several business organizations have invested

in the applications of MCC, specifically in social networking applications [3].

The Social networks allow its users to access the networks soon after publishing their personal data such as name, age, gender, interests, whereabouts and habits. The cyber criminals take advantage of these abundant personal information that are uploaded to the social networking websites to manipulate and exploit at their hire and commit illegal activities. The common illegal activities that are being committed on these websites include uploading illegal material, defaming and stalking [4]. The vulnerability of the social networking applications to commit the numerous criminal activities has increased the importance of digital forensics in this domain. The digital evidences collected from social networking applications on a suspect's smartphone have great significance in investigating a crime scenario to prove the suspect's innocence.

The mobile users also communicate through the cloud storage services to store the data and access storage through a variety of Inter- connected devices [5]. The toughest aspect of investigating a cloud storage service is that it is hard to find out the user's activity till the end of their use of service [6]. Several free and paid cloud storage services are available such as Dropbox [7], Microsoft SkyDrive [8], and Google Drive [9]. These remote cloud storage services store the data of mobile social networking applications such as Twitter, Facebook, Skype, Google plus, and WhatsApp. Therefore, the information collected from the mobile cloud storage has a high forensic value. A Mobile cloud forensics is an interdisciplinary of cloud computing and digital forensics. There are no forensic capabilities between the cloud service providers and customers that could assist investigations of illegal activities in the cloud [10]. There is an urgent need to frame a well-established cloud with forensic capabilities, including a set of toolkits and advanced procedures for cloud investigations [11]. The forensic cloud in [12] suggests the investigators to concentrate more on enhancing the investigation process rather than concentrating on technology used in the investigation process. The mounting demand of mobile device and cloud platform, throws a poses several threats perpetually while there are only limited security features. A review conducted by Ruder Finn (PR agency) estimated that 91% of smartphone users stay in online to access social networks [13]. The number of active

social network users in 2014 was around 1.79 billion. In 2018, it is estimated that the number of active social network users will be around 2.44 billion [14]. Cyber criminals and illegal activities exploit the advantage of the mobile cloud environment to commit crimes in social networking applications. The criminals can broadcast terrorist ideology, share information, and manipulate other's personal information manipulating the social networking applications. The innocent-seeming social network user profiles are susceptible to the danger of secrecy of Android devices [15]. Therefore, incorporating forensic investigation in social networking applications has a high scope and significance to prove the suspect's innocence.

This work attempts to conduct forensic investigation in mobile cloud social networking application by developing a mobile cloud forensic framework. The framework comprises of two components such as a forensic logging module and an investigation process. The investigation process exploits the forensic logging module to conduct investigation process. The framework deploys a CDI tool to perform investigation in mobile cloud based social networking application. The main idea of the MCFFF is to identify and collect the synchronized data copies of mobile and cloud environment.

The next section defines Aim and Objective followed by problem statement, contributions and related work in the area of mobile cloud application forensics. The section 3 is an overview of Mobile cloud forensics frame work and two case studies. In section 4 we explain experimentation and results of the case studies followed by conclusion and future work

A. Aim and Objectives

It is a challenging task to identify and collect the evidential artifacts from the mobile cloud environment. The primary goal of this work is to develop a forensic framework to identify and collect the evidence from the social networking applications to ensure the device's use of cloud service by correlating the evidences collected from mobile and cloud. The core aim and objectives of this work include:

- a) *To develop a mobile cloud forensic framework to identify the use of the cloud-based application and to collect the evidential artifacts on the cloud with the assistance of data identified and collected from the mobile device.*
- b) *To develop a unified forensic framework for data identification and collection in mobile cloud social network applications. This framework is designed to support various applications that run on different mobile devices and flexible enough to work with various cloud service providers. It validates and ensures a forensically sound approach.*
- c) *To solve the multi-jurisdiction and multi-tenancy issues in collecting the evidential data from the cloud environment and establish and maintain a chain of custody for evidence.*

B. Problem Statement

The Mobile devices are not pre-installed with security software to protect from malicious applications. In the absence of security software, the cyber criminals could use malicious application with ease to get an access to the user's private

information. There is no unified framework for assisting the forensic investigation. It is still a challenging task to conduct forensics in a mobile cloud environment. The activities performed through social networking applications can be stored both in the mobile and cloud environment. However, several earlier works are limited to the retrieval of very basic information related to the use of social networking applications. There are possibilities to determine whether the activities performed through social networking applications and collect them. The current forensic investigation procedure lacks the recovery of data from both the mobile device and the cloud. The identification of the real suspects is a major issue in the mobile cloud environment. There is no unified mobile cloud forensic architecture to support several applications running on diverse Smartphones with different cloud service providers. For any mobile cloud social network application, there is some possible provenance to prove the ownership or the connection from the seized device, but it is absent while examining the cloud. Therefore, there is a requirement for identifying and collecting the residual artifacts of mobile applications left behind the device and a cloud.

C. Contributions

The key contribution is the development of a forensic framework for the mobile cloud environment, especially for social network applications. The main contributions of this work include:

- 1) *This work extends the social networking applications with a forensic readiness component called forensic logging module. The Mobile Cloud Forensic Framework exploits forensic logging module for identifying and collecting the forensic rich data in mobile and cloud side. The proposed unified framework supports a variety of social network applications running on diverse mobile devices, and it is a standard framework flexible enough to work with different cloud service providers.*
- 2) *The proposed framework facilitates the forensic capabilities that collect the automated synchronized copies of data on both the mobile and cloud server to prove the evidence of cloud usage using CDI tool. The add-on component (forensic logging module) enhances the traceability of events performed by the user with respect to local mobile device and a remote server.*
- 3) *This work identifies the evidence of using a cloud service by correlating the artifacts recovered from a mobile device with the cloud artifacts. It validates and ensures collected evidence in a forensically sound condition. It establishes and maintains a chain of custody for evidence.*
- 4) *This work solves multi- jurisdiction and multi- tenancy using information traced in the Forensic logging module.*

II. RELATED WORK

A forensic examination of an Android phone's logical image reveals that basic Facebook friend information is stored in the contacts database as the device synchronizes all contact's Facebook status updates with the device's contact [16]. It also reveals that the device stores Twitter passwords and updates performed through the Twitter application in the plain text.

However, forensic research works on Blackberry and Windows Smartphones do not state on the recovery of any artifacts associated with the use of social networking applications.

Most of the third-party applications on the Apple Mobile platform contain a significant amount of forensic-rich data [17]. The user's information during the interaction with applications is stored in plain text format and can be recovered from the user data partition of the device. Artifacts such as authentication credentials, timestamps, and Geo-locational references locate a device at a particular time. The data related to third party applications can be retrieved from a forensic image of the device. These data may also be available as backups stored on the machine with which the device has been synced.

The forensic analysis of social networking applications of Facebook, Twitter, and MySpace on three popular smartphones such as BlackBerry Torch 9800, iPhone 4, and the Android-based Samsung Galaxy S was conducted to determine whether activities performed through these applications were stored in the internal memory of the device [18]. Moreover, it extends to estimate the amount and locations of data if stored in the device's memory. The results revealed that the activities could not be recovered from BlackBerry devices, whereas activities performed through these three applications left a significant amount of valuable data that could be recovered and used by forensic investigators in the iPhones and Android phones.

The data of WeChat provides evidential artifacts that may be of considerable value in an investigation [19]. The data that could be retrieved from WeChat identifies the suspect's contacts and affiliations, habits and interests, ideas and beliefs in the iPhone. Though the forensic investigation of WeChat application extracts the evidence in the iPhone, it fails to extract those evidences in the cloud environment.

The forensic investigation of Skype calls and chats on the Android devices revealed that evidence can be extracted from the device. It analyzed both the RAM and NAND flash memories in different scenarios and time. The results of the analysis revealed that the Skype call patterns and chat messages get stucked in the RAM, and NAND flash memories even after deleting calls, chat histories and signing out of the Skype [20].

There are several gaps in the forensic investigation of the mobile cloud environment. There are only a few works in the literature investigating the social networking applications in the mobile cloud environment. Most of the works recovers forensic rich data only from the mobile device. There are no add-on components specifically deployed for the forensic purpose. Moreover, there is no valid solution to multi-tenancy, and multi-jurisdictional issue on the cloud side. There are no efforts made to probe the activities performed through social networking applications by the user in the mobile and cloud environment. This paper mainly focuses on the forensic investigation of cloud-based mobile social networking applications.

III. MOBILE CLOUD FORENSIC FRAMEWORK

The methodology proposes a forensic framework for the mobile cloud environment consisting of two major modules such as a forensic logging module and an investigation module using a tool called ClouDroid Inspector (CDI).

A. Basic Overview of Mobile cloud forensic framework

This work introduces a novel Mobile Cloud Forensic Framework (MCFF) that supports forensic investigation in social network applications. Any of the social networking applications can extend the forensic logging module to help the forensic investigation. An investigator starts the investigation initially with the mobile device after the investigation process has been triggered internally or externally. With the evidence collected from the device, the investigation process is continued on the cloud side. The forensic investigator extracts the potential evidence traced in readiness component and other sources in the mobile and cloud environment using the CDI forensic tool. The main objective of the proposed MCFF is to ascertain the right direction related to a particular investigation scenario.

The MCFF attempts to prove the cloud usage via Smartphone. Therefore, in MCFF, the investigator conducts an investigation in the mobile device and uses the data collected from the mobile device to access the cloud account. The proposed forensic model uses the potential evidence collected from the mobile device to search for similar correlating evidence in the cloud to prove the cloud usage accessed by the Smartphone. It facilitates to handle a large volume of data efficiently so that an investigator has a clear view of the investigation process.

The proposed framework utilizes the advantage of synchronized copies of data on both the device and the cloud server to prove the usage of cloud services. This work attempts to solve the multi-jurisdiction issues to some extent. This work also deals with the Multi-tenancy property of the cloud that introduces difficulties in identifying the real suspect among several cloud users.

Figure 3 demonstrates the architecture of MCFF to identify and collect the synced data of cloud usage through mobile devices. The mobile device acts as a concept of interface for both accessing the social networking application and for forensic analysis.

a) The forensic logging module represents the pre-investigative readiness components with its components of identity management and event management. The Identity management is the ability of a cloud entity/ mobile entity to deal with individual user identities such as authentication and authorization. The functions of identity management enabler are mainly concerned with authorization and authentication. It includes a policy discourse to define with attributes (roles, and identity), and the request for credentials to grant permission to access the resources.

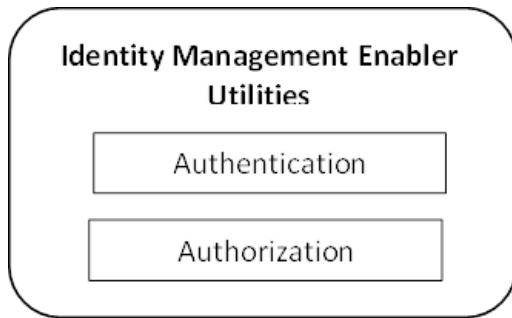


Fig. 1. Identity Management Enabler

The Event Management conceptually constructs the unit of an “event” and technically implement the corresponding concept so that it can be constructed, traced, reconstructed whenever required. Event management is a range of high-level of interoperability among different cloud users. The Event construction defines the event in the cloud environment and mobile environment that answers the questions of “who”, “what”, “when”, “where” and “how”. The Event freezing freezes the event at the immediate state in case of a criminal misdeed or an investigation. The Event traceability traces the event’s current state of the cloud system and mobile system or back to its original state. The Time sequence capability records a definite and synchronized time series in the mobile and cloud system. The Event reconstruction capability reconstructs the past state of the event with a level of acceptable accuracy so that the reconstructed information can be declared as the digital evidence.

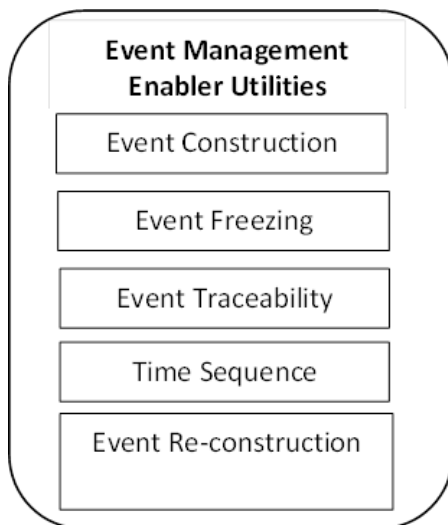


Fig. 2. Event Management Enabler

b) The investigation module in the mobile and cloud indicates the core forensic procedures. This work is designed an investigation tool called CDI that exploits the data traced by the forensic logging module to perform a forensic

investigation initially on the mobile device followed by the cloud shield. Finally, the forensic investigation can prove the mobile device’s use of cloud service through correlating the evidence extracted from mobile and cloud environment. The existing identification and data collection methodologies in forensics do not design a separate logging module. Moreover, the existing forensic data collection methods do not use data reduction techniques. The proposed data collection technique uses reduced data collection technique to collect only the potential evidence that is relevant to the crime scenario. These potential evidences are enough to prove the cloud usage via Smartphone.

B. Forensic readiness - Forensic Logging Module

The proposed approach designs a separate logging module to act as a forensic readiness component and traces the log related to only social network applications despite system log. This is because, the device stores the log files in circular buffer and the storage capacity of those buffers is relatively small. The new information of log file overwrites the oldest information if the list is full. The time factor becomes an obstacle for the forensic investigator. It is not possible in all the cases for the investigator to seize the device right after a crime has been committed. Critical information regarding the crime can be overwritten by any other new information in the log file. Therefore, this work intentionally designs a logging module to assist forensic investigation.

The forensic logging module comprises of a log generator, a log transport, and a log harmonizer. The log generator enables the logging of component from which the logs are to be collected. The status of an application has to be logged with respect to the authorization, whether it is a success or a failure. Some of the essential activities to be traced regarding social networking applications are the login and logout in both local and remote access, password and authorization changes, denied authorization and all the events performed by a privileged account. The log transport module of the logging module transfers the collected records to the device’s local storage (location of the forensic SD card). This store is not accessible to mobile users. The log harmonizer is a centralized component that tunes different log type to appropriate or common log type.

The log fields of forensic logging module are selected based on two functionalities such as identity and event of a user. The identity functionality provides the answers to who committed the crime. The event functionality provides the answers of what, why, and when. Therefore, the forensic logging module covers the logging fields in every log such that it provides answers for who, what, why and when in social networking applications. The log fields relating to the identity of a user include User_Unique_ID, and Geo-location while the log fields relating to events performed by the user include Time_stamp, Application, Event_Performed, Session_ID, Status, Severity, and reason.

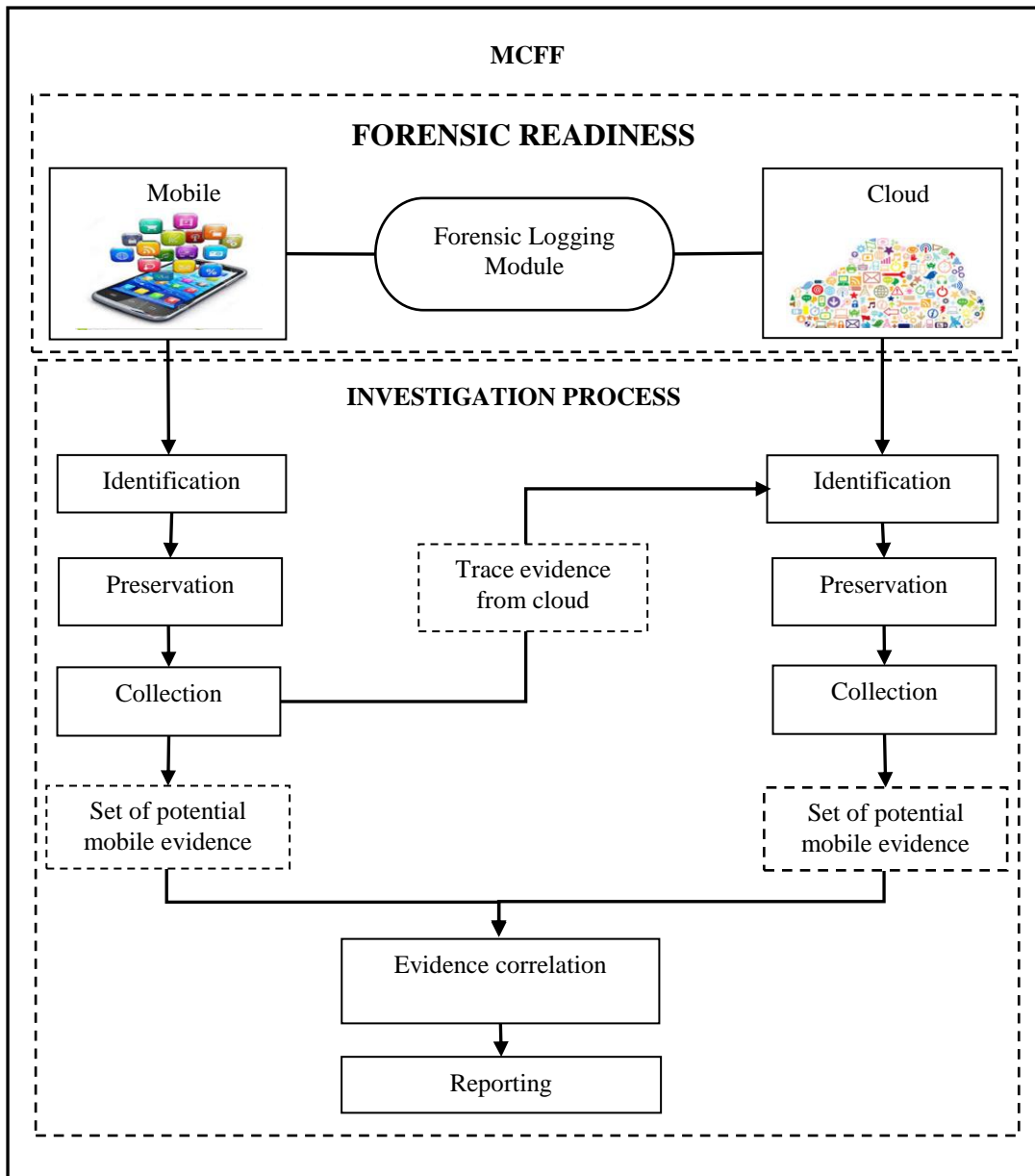


Fig. 3. Architecture of Mobile Cloud Forensic Framework

```

{
  User_Unique_ID, Time_stamp, Application,
  Event_Performed, Session_ID, Geo_location, Status, Severity, reason
}
    
```

The User_Unique_ID log field reveals the unique ID of the particular mobile user accessing the cloud service (social network application). The unique ID considered in this work is International Mobile Equipment Identity (IMEI) number of a device. Time_stamp indicates the time of the traced event happened. The Application field indicates the social networking application that generated the log entry. A Session_ID represents a single request from the mobile to cloud service. The Geo_location represents the latitude and longitude of the device. The status represents whether the login request for accessing the social network is success or failure. A Severity categorizes the type of log such as debug, info, warn,

or error identifies the reason why something has happened. For instance, the reason explains why the access was denied. The investigator can even apply data reduction techniques in the logging module to filter the log records in terms of any of the logging fields to obtain most relevant evidence to the crime scenario.

C. Forensic Evidence

The android OS offers an open development platform. This facilitates developers the control to develop applications that gain full advantage of some functionalities such as location information, the ability to set alarms, add notification services to the status bar, and several other features. The files and meta-data extraction process involves 3 steps: initially, obtains the image of the disk, extracts the files from the image and analyzes the evidences. This process is implemented initially

on the mobile side and followed by the cloud. Finally, correlates the evidence collected from mobile and cloud side. Two scenarios are created to conduct a forensic investigation using the proposed ClouDroid Inspector tool to prove the use of social networking applications installed in Smartphones by running Android operating system. The CDI tool is responsible to conduct investigation in mobile and cloud environment. The CDI tool is capable of locating the log records traced by the forensic logging module in cloud and mobile. The CDI tool initiates the forensic investigation in mobile side followed by the cloud side. This work develops a sample social network application called Model Social Network Application (MSNApp) and SocialApp. The functionality of the MSNApp and SocialApp are similar to other social network application that allows registered users to a create account, upload images, audios, and photos, send texts, and keep in touch with friends, family and colleagues.

Scenario 1: An employee (suspect) is believed to be transferring the stolen confidential documents to another company using his cloud account through Smartphone. This scenario is considered to illustrate the capability of the proposed work to prove the data possession of the suspect.

- Step 1: Identification in mobile device

The investigator identifies the sources of evidence in the suspect's mobile device. The Identification and collection of evidence in mobile devices are not formally conducted as the real identification would proceed with law enforcement identifying mobile devices. The mobile device is kept in a controlled virtual machine environment. Enable the USB debugging tool called Android Debugging Bridge (ADB) present in the SDK to get connected to the device and access the command shell with rooting privileges and make a copy of the system partitions stored in the internal memory. The most common technique to identify the source of evidence in mobile devices is "Imaging". This virtual image collection would be equivalent to the physical disk image file. There are several system partitions in the Android OS such as boot, system, recovery, data, cache, and misc. The MTD (Memory Technology Device) based devices have "/proc/mtd" populated with the partition layout, by the linux kernel. Thus, no specific partition layout file is required by Online Nandroid, on MTD based devices. This work is interested in collecting data from the system and data partitions. The identified files are preserved using a software Write-blocker.

- Step 2: Collection in mobile device:

The CDI tool initially collects the log files traced by the readiness component. The logging module stores the log files (social network application logs) in the system reserved area so that it can be accessed only by the CDI. The CDI not only collects the log files but also other files such as sync and file meta-data. The sync and file meta-data artifacts in mobile device have high forensic importance that reflect the use of cloud computing. These artifacts not only provides log data, but also determines the cloud instances that assist to link the user actions with the data stored in the cloud environment through the file meta-data such as authorization and timestamps.

With the rooted Android mobile device, the CDI tool access the protected directories on the system related to social network application's data (/data/data directory) and a backup of all the files in this directory in the suspect's Smartphone. Install a new forensic SD card in the device and the location of this SD card is selected as the location to write the collected data to the forensic memory card. The files copied in the forensic memory card are then analyzed individually. The "dd" / "cp" command is used to collect the files from the mobile and copy to the forensic SD card. The CDI collects the details of the logging module as shown below.

```
358472042445412, Fri Apr 24 09:54:37 GMT+05:30 2015,  
MSNApp, upload, 07GdfrGGRjYHFS3KIYHGR, 121.47856  
46.52379 4362, success, Info, does not exist
```

The CDI exploits SQLite database browser to collect data from the SQLite database. The SQLite database contains numerous significant data required for the current investigation process. The CDI retrieves the username and password from the tables of SQLite database and uses this information to access the social networking application.

The sync metadata is stored on the client in SQLite database (DB.sqlite). The DB sqlite file contains a wide range of information of forensic importance. The database comprises of a table called "metadata". It stores the cached files in synchronized directories on the local disk. The file metadata configurations/ databases can locate the files/directories synced to the local client. The data related to the cloud service and authentication is collected from "cloud.cfg" which is a configuration file located in the path directory of the corresponding application. The configuration file comprises of a list of "url", "user", and "passwords". This information is enough for the investigator to prove the usage of cloud services through mobile device. The mobile device offers cloud details and credentials for the user and cache files accessed and a list of files stored on the Cloud instance during the last sync.

The android device does not facilitate a unified backup solution. Several organizations have developed backup tools to facilitate the user for backing up the device using SD card or organization's server. To recover the deleted data on the mobile device, the configuration files have to be synced with the "Backup" directory using the extension of ".cfg". This allows the CDI to recover the cloud instance, username, and password used on the mobile device even after the suspect has deleted the content on the mobile device. Some application stores the username and password in the SQLite database in the encrypted form. It is difficult to retrieve the original password from encrypted form. The possible solution is to use the source code of the installed application. If the password is stored in encrypted format, it can be decrypted using the decompiled Java code for decryption from apk. The encrypted password is passed as an argument to the decryption method that returns the original password.

- Step 3: Identification and collection in cloud:

The CDI precedes the investigation in the cloud side with the data collected from the device. The simplest method of

identification of cloud artifacts is the installation of the forensic logging module. This module is deployed in the cloud purposely for forensic investigation in a special storage. The username and password retrieved from mobile (SQLite database) is used to access the cloud account. In a cloud environment, the data uploaded by the suspect are more important evidence. Data generated by a cloud instance corresponding to a particular user is valuable evidence that links the user to the data located in the cloud environment such as log data or encryption keys. The evidences are extracted using “dd”/ “cp” command. CDI facilitates the investigator to use data reduction techniques to collect a data subset from remote cloud result in time and storage size savings and also makes the consequent investigative steps easier. For instance, the investigator can use the Time_stamp detail (Fri Apr 24 09:54:37 GMT+05:30) to search the data relevant to the given time_stamp.

- Step 4: Correlation of evidences:

The investigator currently has the data collected from the cloud and the device. The CDI collects log files from the forensic logging module installed both in the mobile and cloud and other critical information. The sync and file metadata, cloud authentication and service data collected from the device supports the investigation.

The file MSNApp.db (collected from SQLite database) holds a set of forensic-rich activities performed by MSNApp user includes chat messages, list of friends, mailbox, and uploaded files, photos and videos. The records include critical information such as the user’s ID, contents exchanged, URL of uploaded materials, and timestamps of performed activities. The investigator checks for the same details in the log files collected from the mobile device and from the cloud to prove the suspect’s data possession. The same set of details of a user is recorded in logs of both cloud and mobile, and hence, the cloud usage by the user can be proved by timeline analysis. The proposed framework solves the multi-jurisdiction issues and multi-tenancy issues by using user_unique_ID and Geolocation information in the logging module.

The forensic investigator correlates this evidence to reconstruct the digital crime scene after collecting the evidences. Moreover, the cloud system tracks every change occurring in the client data and keeps different versions of them using a versioning technique. This technique helps to retain the original data at different times. This work has chosen the versioning technique to reconstruct the evidence in a short time between the cloud and the mobile. The information collected from logging module, SQLite database, synced and file metadata proves the data ownership of the suspect. The investigation process is then documented for reporting.

Scenario 2: A Person ‘X’ uses the victim’s cloud account to spread (upload) inappropriate material via MSNApp on Fri Apr 24 09:54:37 GMT+05:30 2015. This scenario is considered to prove the innocence of the victim using MCFF (CDI tool). In this scenario, the person ‘X’ is a real suspect. Moreover, this scenario highlights the importance of considering the cloud evidential artifacts to deal with the scenario in addition to mobile evidential artifacts.

The investigator keeps the time at which the inappropriate material has been spread as the reference time. In this scenario, the investigator has to check whether the activity logged at the reference time in both the mobile and cloud is the same or different. The victim may or may not be accessing the social network application at the suspected time of inappropriate material sharing. In case, if the victim is accessing the social networking application at the reference time, the activity performed will be different from that of the victim’s friend. If the victim is not accessing the social network application, the event will not be traced in the logging module as the module traces only the logs related to social networking application.

- Step 1: Identification and collection in mobile side:

The device and the cloud traces and stores the log files in the forensic logging module. The CDI tool gains the root access and identifies the system partition. The CDI tool initially images the possible sources of evidence on the mobile side. The “data” partition consists of data related to third party application. The CDI tool identifies the logging module and other critical location holding files of interest. The use of software write-blocker preserves the identified files. The SQLite database in the device contains more critical information. The Data regarding the use of cloud service and authentication is identified and collected from “cloud.cfg” which is a configuration file located in the path directory of the corresponding application.

The configuration file comprises of a list of “url”, “user”, and “passwords”. The “dd”/ “cp” command is used to collect the evidences. The information is collected from the device to the installed forensic SD card as explained in the scenario 1. The information collected from the logging module of the mobile device is listed below. According to this scenario, the information in the logging module is enough to prove the victim’s innocence.

```
388391275335464, Fri Apr 24 09:54:37 GMT+05:30 2015,  
SocialApp, chat, 02KdgrFFRjYYHFS3KJKGGR, 124.75567  
35.61842 3648, success, Info, does not exist
```

The log files reveal that the victim’s device has used a social network application called Social App (a sample social network application) on Fri Apr 24 09:54:37 GMT+05:30 2015 and performed a chat activity.

With this evidence, the investigator cannot arrive at a conclusion because there is a chance for the victim to alter the log details. Therefore, the investigator has to investigate the cloud side to decide on victim’s compliance.

- Step 2: Identification and collection on cloud side:

The CDI is capable of identifying the location of files stored in the forensic logging module in the cloud. The investigator then makes a decision for selecting the method of collection based on the quantum of the instance. The information collected by the logging module of the cloud are listed below.

388391275335464, Fri Apr 24 09:54:37 IST 2015, SocialApp, chat,
02KdgrFFRjYYHFS3KJKGGR, 124.75567 35.61842 3648, success,
Info, does not exist

- Step 3: Correlation of evidences:

The investigator can prove the cloud usage using the collected artifacts only after the correlation of evidences. The investigator can correlate the log files (in this scenario) collected from the mobile and cloud side to link the victim with an activity. Cross referencing these artifacts in the aspect of timeline analysis determines the distributed user activity. There are several challenges in correlating the cloud and mobile evidential artifacts. The time_stamp of an activity differs from server to server. Therefore, time_stamp from the local file system of every server in the cloud does not represent the same time. This work suggests setting up a reference time in the cloud and to use the time synchronization protocol. The log records have revealed the same details on cross referencing the evidence collected from the device and cloud. It confirms that some other person has used the victim's MSNApp cloud account to spread illegal material on Fri Apr 24 09:54:37 GMT+05:30 2015. This set of evidences proves the victim's innocence. The investigation procedure is finally documented for reporting. The detection of real suspect is not the focus of this scenario but to prove the innocence of the victim using MCFE.

IV. EXPERIMENTATION

This work uses the Java language to conduct mobile cloud forensic investigation on the Android platform.

A. Development tools

The Eclipse is an integrated development environment which ensures a complete functional and commercial quality industrial platform. It contains a base workspace and an extensible plug-in system for customizing the environment. The Eclipse is an open source software comprising of the Eclipse platform, Java Development Tools (JDT), C/ C++ Development Tools (CDT) and Plugin Development Environment (PDE). The Eclipse platform is an open extensible IDE providing a common development platform. The JDT supports Java development, CDT supports C/ C++ development, PDE supports plug-in development.

An Eclipse SDK is a combination of all tools and components produced by Eclipse platform, JDT, and PDE. These components together offer a feature-rich and complete development environment to allow developers to develop tools that can be efficiently integrated with ease into Eclipse. The cloud uses Netbeans integrated development environment.

B. Android SDK and virtual device controller

An Android SDK offers the development components which are used to develop Android applications on Windows/ Linux/ Mac. The Android OS supports all platforms with a tool set for Android mobile application development. This tool set comprises of Android emulator, plug-in tools for Android development used in Eclipse (ADT), and tools for debugging,

packaging and installing applications in the Android emulator. The Android SDK supports the Java language to develop applications on the Android platform. The developers could use the tools provided by SDK to introduce the program into .apk file as a package and use the Emulator to simulate and test the Android application.

The actual device has been designed in a virtual environment using Android Virtual Device (AVD). Each AVD defines the hardware and the software options and configures several projects. The virtual device controller offers a graphical user interface in which the developer can create and manage Android Virtual Devices (AVDs).

C. Dalvik Debug Monitor Server (DDMS)

A DDMS is a debugging tool that provides port-forwarding services, screen capture on the device, logcat, current progress, and ratio state information and location data spoofing. Delete the author and affiliation lines for the second affiliation.

D. Results

This section explains the correlation phase of the forensic process. It is this phase that analyses and compares the evidences obtained from the device and the cloud.

Scenario 1:

The CDI tool initially identifies the location of the log records traced by the forensic logging module in the device. The investigator obtains the entire log file and filters out specific details. These specific details correspond to the data relevant to the crime scenario. The files stored in the forensic logging module are recovered from the application package directory /data/data/com.example.ModelSocialNetworkApp. This directory contains four folders such as Log, cache, databases, and lib. The log files are obtained from the directory /data/data/com.example.ModelSocialNetworkApp/Log. The Log file consists of the information such as IMEI number, Time Stamp, Application, Event, Session ID, Geolocation (Latitude, Longitude), Status, Severity, Reason. The important source of evidence in device is SQLite database. The records in the database of the MSNApp can be obtained from /data/data/com.example.ModelSocialNetworkApp/databases. The database of MSNApp comprises of three tables such as UserDetails, android_metadata, sample_table. The actual username and password are recovered from the table called UserDetails. The username and password for accessing the MSNApp are "user1@gmail.com" and "user1@123" respectively.

The files in the database contain several other significant information such as the chat messages and the URL links of uploaded pictures. The uploaded files from the MSNApp have names preceded with the word "upload". The SD- card is installed in the directory /mkdir/sdcard/Forensics/ for forensic purpose. The evidential artifacts are written into the SD card. The password and the username extracted from the device are given as input to access the MSNApp. The CDI tool recovers the log files from the cloud directory /home/eucalyptus/Forensic/log/LogFile1.log. This log file is also written into the SD card.

```
358472042445412, Fri Apr 24 09:54:37 IST 2015, MSNApp,  
upload, 07GdftrGGRjYYHFS3KIYHGR, 121.47856 46.52379 4362,  
success, Info, does not exist
```

The log in the cloud revealed that the file has been uploaded using MSNApp on Fri Apr 24 09:54:37 IST 2015. The log in the mobile also has revealed the same. Therefore, the uploading of the file via MSNApp from the device has been proved using the CDI tool.

Scenario 2:

There are two social networking applications on the victim's device called MSNApp and SocialApp. The victim's friend has used the victim's MSN account to upload inappropriate material on Fri Apr 24 09:54:37 GMT+05:30 2015. To prove the victim's innocence, it is enough to collect and correlate the log files in mobile and cloud side. The CDI tool collects the log file from both the social networking application installed on the victim's device. There is no log information gathered from the MSNApp at the reference time from the victim's device. Instead, a log file is obtained from the SocialApp at the reference time from the directory /data/data/com.example.SocialApp/Log.

```
388391275335464, Fri Apr 24 09:54:37 GMT+05:30 2015,  
SocialApp, chat, 02KdgrFFRjYYHFS3KJKGGR, 124.75567  
35.61842 3648, success, Info, does not exist
```

The log collected from the victim's device reveals, that the victim was chatting on Social App at the reference time and no log was recovered from the MSNApp. The CDI tool progresses the investigation to the cloud side as there is a chance for the victim to delete the log details. The CDI tool collects the log records from the directory /home/eucalyptus/Forensic/log/LogFile1.log and filters the record for the entries related to the reference time. The log file collected from the cloud appears as the same as the log file obtained from the device. This confirms that victim has accessed the Social App at the reference time. Therefore, the correlation results conclude that someone has used the victim's MSNApp account to upload the inappropriate material.

V. CONCLUSION & FUTURE WORK

This work proposes a Mobile Cloud Forensic Framework to support forensic investigation in mobile cloud environment. The two major parts of the framework are the forensic logging module and the forensic investigation process. The forensic investigation process employs the forensic logging module to conduct investigation. This work has proved that the proposed MCFF has secured a number of significant evidences from both the mobile and the cloud. In future any social networking application can extend the forensic logging module. Two scenarios are created to validate the effectiveness of the proposed MCFF. The CDI tool, locates the files of the logging module in the device and the cloud. It initially collects the sensitive information from the device and then proceeds to the cloud. The CDI successfully conducts the forensic

investigation in two scenarios. The MCFF is able to trace the cloud instance, even if an evidential data is securely deleted on the mobile. The correlation of potential evidences of mobile and cloud alone considerably restricts the time spent on the investigation.

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Features Management and Middleware of Hybrid Cloud Infrastructures

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Abstract—The wide spread of cloud computing has identified the need to develop specialized approaches to the design, management and programming for cloud infrastructures. In the article were reviewed the peculiarities of the hybrid cloud and middleware software development, adaptive to implementing the principles of governance and change in the structure of storing data in clouds. The examples and results of experimental research are presented.

Keywords—Cloud Infrastructure; Distributed Databases; Hybrid Clouds

I. INTRODUCTION

Cloud computing is a paradigm for hosting clusters of data and delivering different services over the network or Internet. Hosting clusters of data allows customers to store and compute a massive amount of data on the cloud. Cloud computing is based on known technology of virtualization of systems in large data centers and the use of existing communication channels.

Currently, capturing and processing big data are related to improving the global economy, science, social affair, education and national security; processing of big data allows us to propose accurate decisions and acquire knowledge from raw data. Several traditional solutions have emerged for dealing with big data such as Supercomputing, Distributed Computing, Parallel Computing, and Grid Computing. However, elastic scalability is important in big data which could be supported by cloud computing services. Cloud computing has several capabilities for supporting big data which are related to handling of big data.

With the growing popularity of cloud services not only the scope and types of services expand (IaaS, PaaS, SaaS), but also the need for new solutions, "cloud" of tasks: providing security and data integrity, quality of service, construction of systems of data flow management and principles for data distribution in the cloud. All of this determined a large amount of scientific research with respect to the clouds that allows us to speak about current complex technologies - cloud technologies (cloud computing technology).

Cloud computing could address two major issues: big data storing and computing. Cloud computing provides a cluster of

resources for storage and computing that could be expanded anytime. These features allow cloud computing to become an emerging technology for dealing with big data.

Hybrid architecture at first glance seems to be optimal – it can provide distributed computing and guaranteed security. However, this data network creates a lot of features. These questions are consecrated in the article. In the second part common questions are presented, the third part is devoted to the study and management features. The fourth part describes the development of software applications. In conclusion, there are formulated problems and made general recommendations to overcome them.

The experimental stand with VMWare vCloud was created to simulate the work with hybrid storage. Within this stand it was possible to adapt a software application in dependence on changes in the structure of distributed data storage, as well as test methods for monitoring the operation of applications.

II. COMMON PROBLEMS

From security perspective hybrid cloud is the preferred architecture. As it is known, hybrid cloud deploys data in public and private parts. For safety reasons, we can store in the private parts data that requires secrecy. In the public part - free data access, processing of which requires large computational resources. For example, let's take semistructured data. In that case in public part the main volume is stored, and the codes and algorithms are stored in a private part of the cloud infrastructure.

The public part of the service may be provided by a specialized data center. A private part is an own servers, which can also be a virtual machine (VM).

To provide users with the same features found in commercial public clouds, private/hybrid cloud software must [1]

- provide uniform and homogeneous view of virtualized resources, regardless of the underlying virtualization platform (such as Xen, Kernel-based Virtual Machine (KVM), or VMware);
- manage VM's full life cycle, including setting up networks dynamically for groups of VMs and managing

their storage requirements, such as VM disk image deployment or on-the-fly software environment creation;

- support configurable resource allocation policies to meet the organization's specific goals (high availability, server consolidation to minimize power usage, and so on);
- adapt to organization's changing resource needs, including peaks in which local resources are insufficient, and changing resources, including addition or failure of physical resources.

Although this system has evolved around public clouds — commercial cloud providers that offer publicly accessible remote interface to create and manage virtual machine instances within their proprietary infrastructure — there is also a growing interest in open-source Cloud Computing tools that allow organizations to build their own IaaS clouds using their internal infrastructure [2]. The primary aim of these private cloud deployments is not to sell capacity over the Internet through publicly-accessible interfaces, but to provide local users with flexible and agile private infrastructure to run service workloads within their administrative domain. Private clouds can also support hybrid cloud model by supplementing local infrastructure with computing capacity from an external public cloud.

A hybrid cloud can allow remote access to its resources over the Internet using remote interfaces, such as the web services interfaces used in Amazon [3].

A lot depends on how data structure is decomposed. This is especially noticeable on large data sets. Minor errors in the partitions between the private and the public parts can significantly affect performance of the system. And increasing the number of processors, or other resources won't help. Unlike the paradigm of distributed databases, in a hybrid cloud there are many unknown factors that need to be responded. After all, the hybrid cloud is the infrastructure, there are two independent cloud storages with their own management systems. Compounding and switching systems is an unknown factor as well as information delivery routes between clients and parts of the system. In the absence of control, application may spend most of the time waiting for the response, rather than computing tasks of data processing. These situations should not only be addressed in time but are under constant monitoring. It is necessary to determine the point at which to start the mechanism of redistribution of resources in the network. All of the above applies primarily to large data. The experiments with small amounts of data and small queries are not sensitive to it. And this refers specifically to a hybrid cloud that constantly exchange data between private and public parts.

For designing information systems in the cloud, there are the following problems:

- Inability to assess the execution of individual requests and the flow request.
- There are no general principles of designing systems with large amounts of data (BigData).

- A considerable amount of data is semistructured (XML).
- No migration technologies to the cloud, hence the need to rewrite code when porting.
- There are no generally accepted principles for virtualization management and resource allocation in the cloud.
- Limitations associated with the use of data communication protocols.

The task was to study these features and develop the technology to create applications in a hybrid cloud. Within the framework of these limitations, principles were formulated to develop methods that provide guaranteed quality of functioning of the application.

- Design of the systems should be based on preliminary study, on simulation and experimental models.
- It is necessary to control basic parameters of the infrastructure.
- The use of object-oriented design technologies for database modifications.
- Technological systems should provide the flexibility to change structure, volume of data, the number of requests.

III. HYBRID CLOUD MANAGEMENT

Relative performance differentiation schemes have been identified as promising approaches to achieve the performance differentiation objectives of multiple classes workloads and applications. The main challenge of applying it in a shared resource environment is to maintain the above performance differentiation ratio by using dynamic resource allocation under varying resource demands and workloads [4]. The system controlled by the controller is referred to as the target system. The target software system provides a set of performance metrics for properties of interest referred to as outputs. The sensors monitor outputs of the target system, while controlling inputs. The controller is the decision making unit of the control system. Its main objective is to maintain the outputs of the system sufficiently close to the target values, by adjusting the control inputs. These target values are called the set point signals. This gives the option for the designer to specify the goal/desired values for the outputs.

The control system design generally consists of two main steps. First, a formal relationship between the control inputs and outputs has to be constructed. In control theory this relationship is referred to as the dynamic model of the system. System identification is typically used to construct the system model using the measurements of input and output data. This model of the system is then utilized in the subsequent steps which include controller design, simulation, analysis and testing using well established tools in control engineering.

Many control engineering approaches have been proposed for data center management in the past few years. Works in [5], [6], [7] have proposed different control techniques to

manage power and performance properties at absolute value. Several techniques control response time by manipulating CPU utilization in virtualized environments.

The relative performance automatic control with feedback have been utilized to manage web servers, storage systems and data centers . In [8] were investigated three different ways to formulate the input and output variables of the relative management scheme in order to reduce the nonlinearity, while maintaining the scalability and applicability of the feedback control. Final results indicated that taking the ratios of the response time and resource caps of consecutive client classes based on the priority is the most suitable and effective setting.

To develop applications with the distribution of big data in the hybrid cloud, the experimental stand can be used. Experience shows that for each system it is required to check the load and response time, generate popular searches [9]. The experiments allow to find out the nonlinearity in each particular system.

There was created an experimental stand (ES) that simulates the work with hybrid storage. Stand itself and some experimental results obtained with it are described in [10], see fig. 1. Deployed software VMware vCloud allows organization at all levels. VMware ESXi is used on two servers to create a cloud in the ES. Management system VCenter and application VMware vCloud Director are deployed. In ES there are more than 15 physical Cisco 29 switches and routers Series 26 and Series 28, as well as virtual switches Nexus. System based on ES allows simulating routes of access to data, converging and diverging channels (can be done dynamically).

As a criterion for the efficiency of the data structure it is proposed to use structure decomposition between distributed repositories, in which secure access to the data users will be provided. A criterion is given in the form of a range of data delivery time for test requests. Clearly, this criterion is strongly dependent on many parameters which are specific for each case in each system, namely:

- — actual distance to the end user (or data delivery route);
- — load of the network;
- — current number of users and the complexity of their application request;
- — downloaded applications, ensuring data collection in the private and public section;
- — actual loading of resources in the cloud.

Many are nonlinear and cannot be calculated in fully automatic mode for the general case [11]. To configure and operate instructions, it should be possible to implement monitoring system (and preferably BPCS system to monitor and control loading). Example of operational monitoring parameters is shown (Fig. 2).

A series of experiments with different demands and workload simulation channels (Fig. 3) shows a portion of the network traffic.

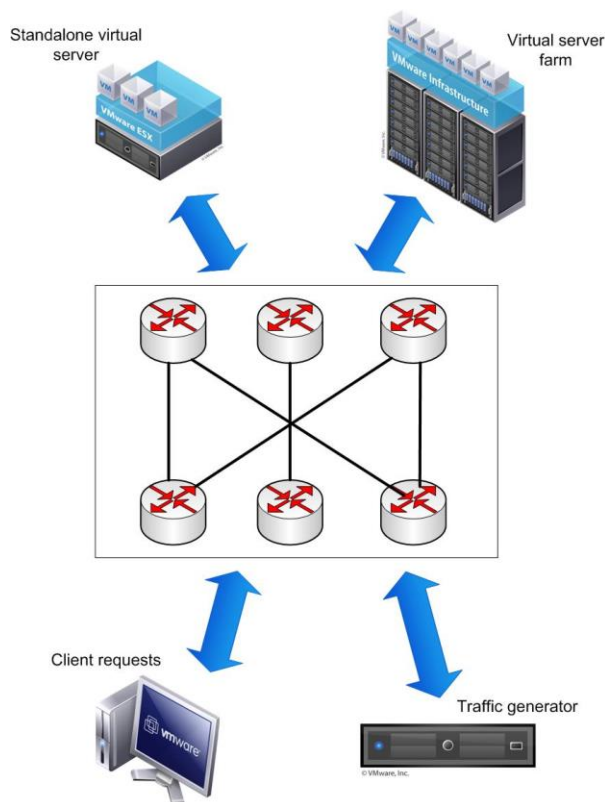


Fig. 1. Experimental installation

However, all these features need to be considered during development of applications. This is done by monitoring systems with feedbacks and policy management. In case of fully loaded system or one of the elements of the infrastructure (network, storage in a private cloud), it is possible not to give the implementation of requests of new users, allowing to perform the previous ones. There are a number of parameters that can be calculated during the design phase. Although it is impossible to evaluate the effectiveness of the algorithm for the cloud infrastructure for data distribution and virtual computing resources (due to the scale, it is impossible to assess in advance how much will be allocated to the algorithm), but you can choose the most popular complex queries, and define the data structure for them. In this case the execution of these requests will be guaranteed to get at a given level of service quality.



Fig. 2. Type of system for monitoring parameters of hybrid clouds in the experimental stand

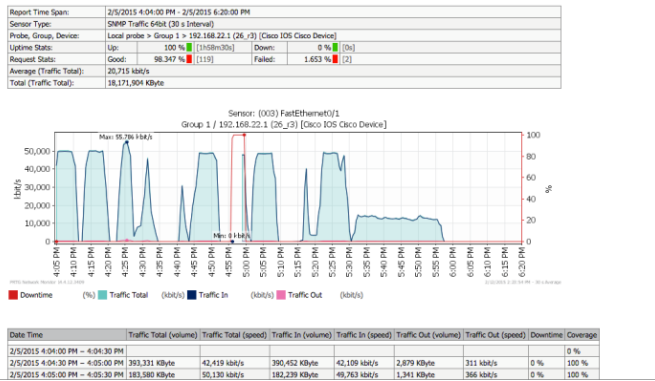


Fig. 3. Traffic Analysis

IV. MIDDLEWARE FEATURES

Development of the information systems and database applications led to the emergence of technologies such as Open Database Connectivity (ODBC), Data Access Object (DAO), Borland Database Engine (BDE), providing a common programming interface for working with various databases. In the future, the needs of the development of software and hardware that work with the data required to ensure access to non-SQL data stores, e-mail and directory services. To provide these functions new technologies have been developed — Object Linking and Embedding, Database (OLEDB) and ActiveX Data Objects (ADO). With the advent of powerful class systems Frameworks, such as .Net and Qt, data processing technologies became embedded into the database, providing full integration with them, as well as integration with semistructured data in XML. The latter became a common format for storing data in files. Efficient technology to communicate with relational data objects is object-relational mapping ORM [12].

Using middleware ORM is of great use in such implementations as QxORM, EntityFramework, Dapper, Hibernate, and others. But all of these technologies are effectively used for the data stored and managed in only one database. Using ORM technology automates location control for data. With classic design, the designer must be sure to specify data location in each request for a hybrid cloud software to connect and disconnect from the database. All this leads to an increase in complexity of software development and causes errors in the code. ORM allows to incorporate the essence of each attribute responsible for the physical location of data in a distributed system. The development of relevant ORM technology for a hybrid cloud infrastructure is the development of intelligent module that determines the optimum storage of data to improve the performance of the system. The optimality of data storage should be automatically determined based on many criteria, such as network bandwidth, server load, number of customers and others. Many of these parameters can be obtained experimentally. Therefore, intelligent control module must adapt based on the information collected from all storage systems during trial operation.

Big Data is not limited in the use of relational tables, semistructured data, data files, etc. ORM requires modification to be used with heterogeneous data: while

preserving the basic functionality it is needed to allow the programmer to operate with object classes.

It is necessary to come up with technology for development that would manipulate the data in the hybrid cloud providing following functions

1) *The basic operations for interaction with the objects:*

- - Select;
- - Update;
- - Insert ;
- - Delete.

2) *The presence of the transaction control:*

- - Commit the transaction (commit);
- - Transaction rollback (rollback);

3) *Storage of information about distant objects ("map"):*

- - Registration of the object (registration);
- - Cancellation of registration (unregistration);

Architecture prototype «ArPlatform», which is a middleware service for developing an application is shown in Fig. 4. The prototype includes service «ArNotifyService» Library and «ArLib». There are implemented classes in which you can specify the location and distribution of data, classes that provide functionality for management at the program level.

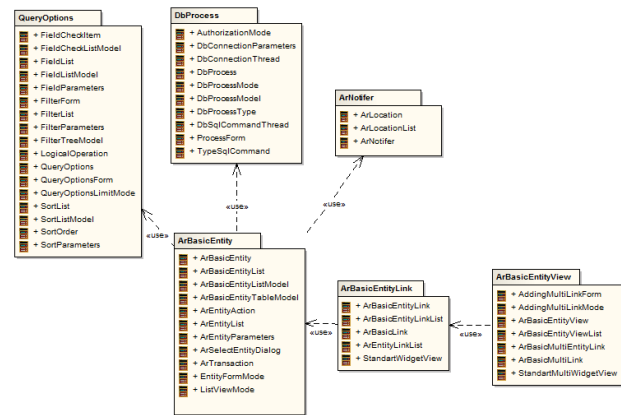


Fig. 4. Structure «ArPlatform»

Examples of operating with the service:

```

Student * student = new Student();
student->setId(5);
DbConnectionParameters db1;
db1.setDbConnectionFile(QDir::homePath()
+QDir::separator()+"connection.ini");
student->selectData(db1);
student->insertData(
    ArEntityNotifyReceiver::instance()->
    dbEntityMap().items().filteredByAppName("serviceSt
udents"));
Student * updateStudent = new Student();
updateStudent->setFio("testcommit");
    
```

```
updateStudent->setMark(4.3);
student->updateData(updateStudent,
    ArEntityNotifyReceiver::instance()->
    dbEntityMap().items());
filteredByHost(QHostAddress(192.168.0.3));
```

In this example data may be stored not only in relational databases but also in files, semistructured instances.

Data manipulation with the network structure:

```
DbConnectionParameters db1;
db1.setDbConnectionFile("connection.ini");
student->updateData(updateStudent,db1);
```

Data service manipulation during installation on a node in a private cloud

```
student->updateData(updateStudent,
    ArEntityNotifyReceiver::instance()->
    dbEntityMap().items());
filteredByAppName("serviceStudents");
```

The aim of the experiment is to determine the efficiency of separation of the DB into 2 parts: public and private. For the experiment 3 compute nodes were prepared, the overall structure of which is shown in fig. 5:

Public DBMS server, which is more powerful .

Private DBMS server, which is less powerful.

A client that makes requests to the published server using specialized software.

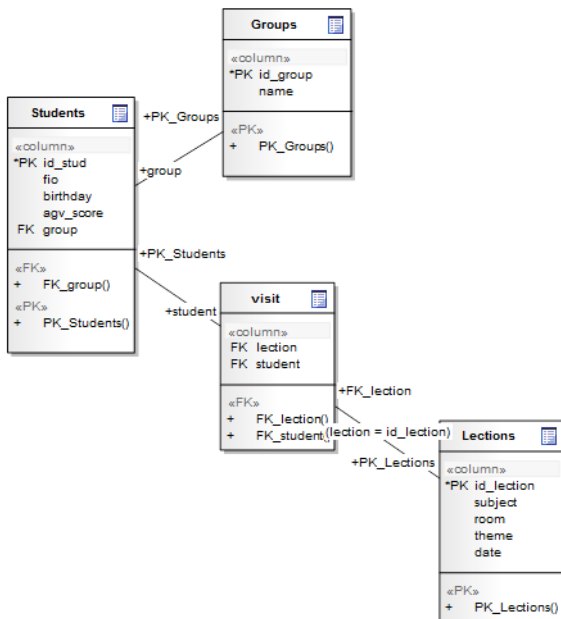


Fig. 5. The database scheme

Part of the database included in the educational process at the University was used for experiments (fig. 6).

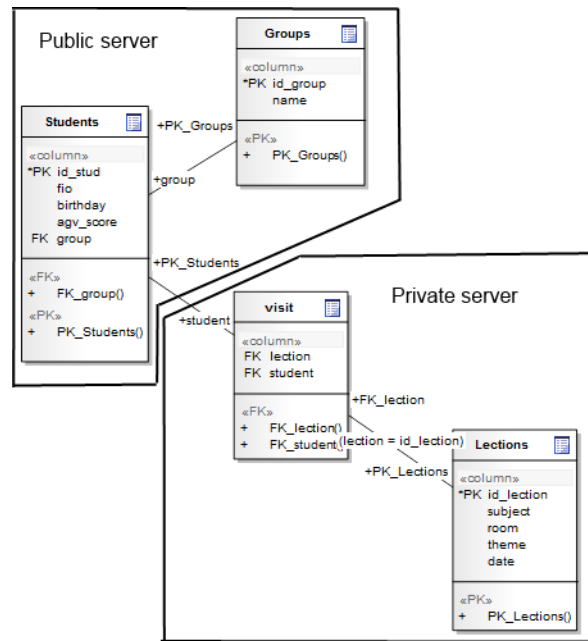


Fig. 6. Split the database between the public and private servers

Table "Students" contains the following information about students:

- "id_stud" - unique number of the student, it is the primary key;
- "fio" - surname, name and second name of the student;
- "birthday" - the date of student's birth;
- "agv_score" - the average score;
- "group" - the group number, which includes the student; is an external key of the table "Groups".
- Table "Groups" associated one-to-many with table "Students" contains the following information about groups:
 - "id_group" - unique group number;
 - "name" - the name of the group.

Table "Lections" associated many-to-many with table "Students" via table "visit", contains the following information about the lectures:

- "id_lection" - unique number of lectures;
- "subject" - the number of the held object;
- "room" - the cabinet number (the audience);
- "theme" - is the theme of the lecture;
- "date" - the date of the lecture.

- Table "visit" includes the attendance of students.
- "id_student" - the number of student. Is external to the key of the table "Students";
- "id_lection" – the number of the lecture. Is external key to the table "Lectons".

The results of the query fetching all data from table "students" (adding data from tables associated one-to-many and many-to-many) were compared to test the effectiveness of the separation of the database into public and private parts.

In the first case, all of the database was on a public server, to retrieve the data following query was used:

```
select * from students.students  
left join students.groups on students."group" =  
groups.id_group  
left join (select * from students.visit  
left join students.lections on visit.id_lection =  
lections.id_lection) t1  
on students.id_stud = t1.id_student.
```

In the second case the part that relates many-to-many with table "Students" was placed on a private server (figure 4). Data was obtained by the function PostgreSQL dblink, which allows to perform the query to another DBMS. Request in the second case:

```
select * from students.students  
left join students.groups on students."group" =  
groups.id_group  
left join (select * from dblink('hostaddr=xxx.xxx.xxx.xxx  
port=xxxx dbname=... user=... password=...', 'select  
id_student,lections.id_lection,id_subject,date,room,theme  
from students.visit  
left join students.lections on visit.id_lection =  
lections.id_lection') as t(id_student INTEGER,id_lection  
INTEGER,id_subject INTEGER,date DATE,room  
INTEGER,theme TEXT)) as t1  
on students.id_stud = t1.id_student
```

Using such functions as dblink (which allows to perform queries to another DBMS), makes it possible to bring part of the database to another DBMS with changes to the query without need to modify client application.

The experimental results shown in Fig. 7 and 8.

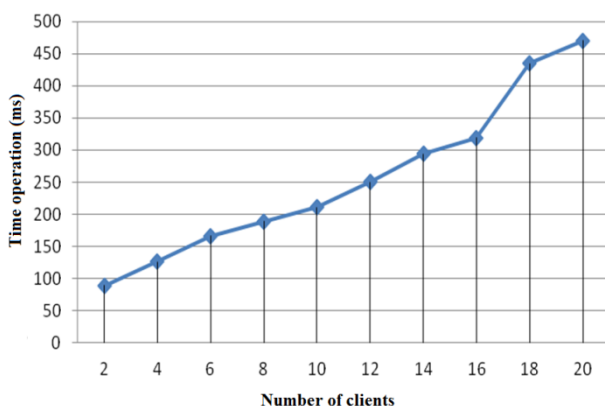


Fig. 7. Experimental query without using ORM technology

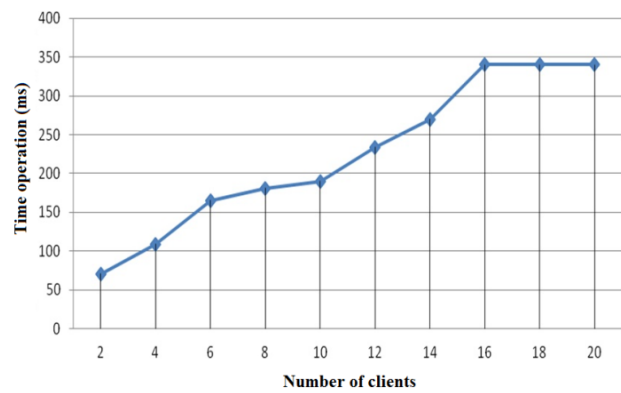


Fig. 8. Experimental query using development

It should be noted that when using a class developed with an increase in the number of clients during the operation grows

V. CONCLUSION

Main feature of the application is an intermediate layer that implements the connection of user requests to the location of distributed data. Presence of unknown destination switching when using public cloud and mobile client makes it impossible to estimate the time of the algorithms. That is why you want to use software technology to control all stages of the system. The hybrid infrastructure has many positive aspects of cloud computing: scalability, virtualization and also (due to the distribution of data) safety and security of data.

The task was to account these features and develop the technology that creates applications in a hybrid cloud.

The principles of development were formulated to provide guaranteed quality and functioning of the application.

- 1) The system design should be based on a preliminary study on the simulation and experimental models.
- 2) It is necessary to control the main parameters of the infrastructure.
- 3) The use of object-oriented technology modifications of database design.
- 4) Technology should provide the flexibility of system structure, data volume, number of requests.

The result shows that in the second case, when the database is divided, the average query execution time is much higher than with solid database. Therefore the separation of the database on the public and private parts adversely affects the performance of the system, and only has advantages from the viewpoint of safety. Distributed data complicates the development of the software, making it difficult and time-consuming to use common programming techniques. Despite the development of technologies such as .Net and Qt, developers eventually have to operate SQL queries and clearly prescribe the access to distributed data. In the context of widespread object-oriented development methodology and application systems, with relational DBMS having dominant position in the market, attractive solution is to use intermediate software that provides necessary object-oriented interface to data stored under the control of a relational

DBMS. To communicate with developed relational data objects there was used special technology. The essence of this technology is in accordance of programming entity to relational database object: each field of a table is assigned to a class attribute of the object.

The basic steps are the following:

- 1) *Determination of the basic structure of the physical distribution of data in the hybrid cloud.*
- 2) *Development of database structure.*
- 3) *Development of methods for data processing based on physical location of the data.*
- 4) *Creating classes of objects, including data and methods for their treatment.*
- 5) *Modification of the structure as a result of an experimental study on the simulation bench.*
- 6) *Changing methods of processing inheritance*

Depending on the task, system can be restructured to increase the speed of the most common queries, or to perform the most demanding requests in the public cloud. Formulated features and developed design technology for middleware will allow applications for hybrid clouds to be effectively designed. This provides the possibility to adapt a software application in dependence on changes in the structure of distributed data storage in clouds, gives methods for monitoring the operation of applications and management principles.

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Proposed Hyperchaotic System for Image Encryption

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Abstract—This paper presents a new hyper chaos system based on Hénon and Logistic maps which provides characteristics of high capacity, security and efficiency. The Proposed hyper chaos system is employed to generate the key for diffusion in an image encryption algorithm. The simulation experiments to the image encryption algorithm which based on the proposed hyper chaos system show that the algorithm security analysis it has large key space (10^{84} that ensures a strong resistance against attack of exhaustion as the key space will be greater), strong sensitivity of encryption key and good statistical characteristics. Encryption and decryption time is suitable for different applications.

Keywords—hyperchaos; logistic map; Hénon map; image; encryption; decryption

I. INTRODUCTION

Multimedia communications; such as, images audio, and video has become significantly more important, since communications of digital products over the network (wired/wireless) has expanded [1,2]. There is therefore, an increasing need to secure data and its transmission and also to identify the required levels of security depending on the purpose of the communication. A wide variety of cryptographic algorithms have been proposed to meet these requirements. Traditional ciphers methods are less efficient in securing real-time multimedia data encryption systems and exhibit some drawbacks and weakness in high stream data encryption[3,4]. The availability of a high computation machine may allow a brute force attack against these types of cipher. Furthermore, for cryptosystem applications that require high computation processes; large computational time and high computing power, as in the encryption of large-scale image encryption are seen to suffer from low efficiency levels [5]. Therefore, these encryption algorithms are not appropriate for many high-speed applications because of their slow real-time processing speed and some other issues related to the processing of different data formats. Current research into the development of new chaotic or hyperchaotic systems is highlighting the benefits of real-time encryption and communication applications. They show that chaotic systems are good schemes for designing cryptosystems, which have preferable characteristic [6]. Within this research a hyperchaotic system is proposed using a one-dimension logistic chaotic system and three-dimension Hénon chaotic system. The proposed hyperchaotic system is applied on image encryption.

II. CHAOS THEORY

Chaos Theory has been a branch of mathematics that has generated much interest; the notion of being able to describe

complex nonlinear phenomena, such as the weather or the stock market, using a series of deterministic dynamical equations is intriguing for many fields of study. The ability to generate ‘chaotic’, unpredictable data using a series of relatively simple deterministic equations is also attractive from a cryptographic point of view [7].

A. Logistic map

Logistic map is a very simple chaotic map and its mathematical expression formula is asin equation (1):

$$X_{n+1} = \mu X_n(1-X_n) \quad (1)$$

$\mu \in [0, 4]$ is called Logistic parameters.

When $\mu \in [3.569946, 4]$, Logistic map works in a chaotic state and produces non-periodic sequence [8]. The map is quadratic and thus nonlinear with equation(2):

$$X_{n+1} = bX_n (1 - X_n) \quad (2)$$

Where b is the control parameter governing the chaotic behavior and to ensure X_n in the range $[0, 1]$, parameter b has to be in the range $[0, 4]$.

B. Hénon map

The Hénon map is one of the discrete dynamical systems that exhibit chaotic behaviors. The Hénon map is defined by two equations and depends on two parameters a and b , and the system exhibits a strange attractor for $a = 1.4$ and $b = 0.3$ (system equation (3)). A Hénon map takes one point (x, y) and maps this point to a new point in the plane [9,10].

$$\left. \begin{aligned} X_{n+1} &= 1 - a(X_n)^2 + Y_n \\ Y_{n+1} &= b X_n \end{aligned} \right\} \quad (3)$$

The Hénon map is very sensitive to initial values, and different chaotic sequences with large translation can be generated by the adjustment of parameters and initial values indicating that is suitable for generation of cryptographic functions, due to the capability of generating massive chaotic sequences; and the is a periodic and non-convergent, so it has excellent pseudo randomness and unpredictability.

Three-dimensional Honen map as it refers to system equation (4).

$$\left. \begin{aligned} x_{n+1} &= a - y_n^2 - b z_n \\ y_{n+1} &= x_n \\ z_{n+1} &= y_n \end{aligned} \right\} \quad (4)$$

The Hénon map generated from this chaotic attractor is more complex than the maps from other chaotic attractors; when $1.54 < |a| < 2$, $0 < |b| < 1$.

III. PROPOSED THREE DIMENSIONS HYPERCHAOTIC SYSTEM

In this work the hyperchaotic system (hyper between 1D Logistic and 3D Hénon map) is employed to generate the key space which will be greater if generated using either one. The proposed hyperchaos system is described by system equation (5). Chaotic behavior of the proposed system when the parameters are $(\mathbf{a}=1.6, \mathbf{b}=0.2, \mathbf{\mu}=3.75)$. The proposed hyperchaotic (Hénon and Logistic) provide the high-dimensional hyperchaotic system which is more complex and unpredictable; through more 'chaotic sequences.' y_{n+1} in system equation 4 of henon chaos system can be computed by using equation 2 of the logistic map and the resulting system which represent the proposed hyperchaos shown in system equation 5:

$$\left. \begin{aligned} x_{n+1} &= a - y_n^2 - b z_n \\ y_{n+1} &= bX_n(1 - X_n) \\ z_{n+1} &= y_n \end{aligned} \right\} \quad (5)$$

IV. IMAGE ENCRYPTION USING PROPOSED HYPERCHAOS SYSTEM

The proposed hyperchaos system is now used in the design of an image encryption algorithm. The proposed image encryption algorithm input is a plain image whilst the output is an encrypted one. The algorithm main steps are:

Step1. Transformation Process.

In this stage the plain image is divided into 8×8 non overlapping blocks that are transformed using 2-D (DCT). The embedding will be performed in YCrCb color space. Then split luminance (y) of image into 8×8 block the 2-Dimension Discrete Cosine Transform (DCT) will be applied on to these blocks, then a quantization process by Quantizing DCT coefficients to the nearest integer value. Cosine Transform(DCT) will be applied to on these blocks by equation (6) [11],

$$\left[G_{ij} = \frac{1}{4} C_i C_j \sum_{x=0}^7 \sum_{y=0}^7 p_{xy} \cos\left(\frac{(2x+1)i\pi}{16}\right) \cos\left(\frac{(2y+1)j\pi}{16}\right) \right] \quad (6)$$

Where $C_f = \begin{cases} \frac{1}{\sqrt{2}}, & f = 0, \\ 1, & f > 0 \end{cases}$

where C_f is C_i, C_j and P_{xy} are the values of image component $i, j = 0, 1, \dots, 7, x, y = 0, 1, \dots, 7$.

Quantization process by using equation (7) [12]

$$q(i, j) = INT[f(x, y) / qm(i, j)] \quad (7)$$

Where

INT : rounding to the nearest integer.

$qm(i, j)$: Coefficient of inter and intra matrices.

$q(i, j)$: Final output of quantization process.

$f(x, y)$: DCT coefficients.

Step2. Diffusion key generation based on discrete hyperchaotic system generator (hyper between Logistic and Hénon map), key diffusion generated in two stages

step2.1 Pre-iterate equation system (5) for N times (number of iteration), where N is a constant. For computing the solutions of the equation system (5)

step2.2: The hyperchaotic system is iterated discretely. For each iteration, we can obtain three key stream elements from the current state of the hyperchaotic system according to the following formula:

$$\begin{aligned} X_n &= \text{mod}((\text{abs}(x_n) - \text{floor}(\text{abs}(x_n))) \times 10^{14}, 256) \\ Y_n &= \text{mod}((\text{abs}(y_n) - \text{floor}(\text{abs}(y_n))) \times 10^{14}, 256) \\ Z_n &= \text{mod}((\text{abs}(z_n) - \text{floor}(\text{abs}(z_n))) \times 10^{14}, 256) \end{aligned}$$

Where

N times (number of iteration) of the hyperchaotic system.

$\text{abs}(x_n)$ returns the absolute value of X .

$\text{Floor}(x)$ returns the value of x to the nearest integers less than or equal to X ,

$\text{mod}(x, y)$ returns the remainder after division.

The output from this stage three key (k_1, k_2, k_3) for each iteration time (N) used to encrypt first (Dc,2AC) for each block.

Step3. Diffusion process

Diffusion process (change value of pixels) is where a selective encryption approach is applied on DC and the first two AC coefficients. Those coefficients are selected from each block and then encrypted using one key of three generated keys by the hyperchaotic system, the selection is made using the following formula:

$$X_n = \text{mod}(X_n, 3), Y_n = \text{mod}(Y_n, 3), Z_n = \text{mod}(Z_n, 3)$$

the key sequence that is used to perform encryption operation is selected randomly. The encryption operation according to equation (8):

$$C(n) = P(n) \oplus S(n) \quad (8)$$

Where

C : ciphered image value

P : plain image value

S : one of the generated key (X_n, Y_n, Z_n)

The process does not end until the set $P = \{DC, AC, AC\}$ is all encrypted. Then the encrypted pixel set $C = \{C(1), C(2) \dots C(M \times N)\}$ is written to the cipher-image.

Step4. The Inverse of DCT (IDCT) and Transform Y'CrCb to RGB color

After finishing selective image encryption (diffusion) of each block, the IDCT is applied using equation (9) [11] to transform the image to spatial domain then, Transform Y'CrCb to RGB color.

$$p_{xy} = \frac{1}{4} \sum_{i=0}^7 \sum_{j=0}^7 C_i C_j G_{ij} \cos\left(\frac{(2x+1)i\pi}{2N}\right) \cos\left(\frac{(2y+1)j\pi}{2N}\right) \quad (9)$$

Where C_i, C_j as indicated by eq. (6).

Step5: Return the encrypted image

The decryption algorithm is similar to the encryption algorithm. It is for the encrypted image, firstly, decrypt the image using hyperchaotic system with the same parameters and initial values as that used in encryption, we will get the original image.

V. EXPERIMENTAL RESULT

The proposed algorithm was implemented using visual basic Net programming language. A set of examples were executed and analyzed as described in the following sub sections.

A. Key Space Analysis

The proposed algorithm took Logistic map parameter μ and iterative initial value x_0 , and Hénon map initial value (x, y, z) as the original key, iteration time N . Each digit has 14 digital numbers, therefore, the key space is $10^{14 \times 6} = 10^{84}$, the key space will be greater, and therefore, the algorithm has a strong resistance against attack of exhaustion as the key space will be greater. For example diffusion key space consists of 3 initial parameters (3-sub keys). The key space of each one is equal 256 bit and the attacker needs 3×2^{256} operations to find the exact key. If the attacker employs 1000 million instructions per second of computer processing to guess the key by *brute force attack*, the computational load in years is:

$$\frac{3 \times 2^{256}}{1000 \times 10^6 \times 60 \times 60 \times 24 \times 365} > 11.2634583 \times 10^{61} \text{ years}$$

B. Key Sensitivity Analysis

The key sensitivity property of the proposed cryptosystem, the test image (Figure 3(a)) is firstly encrypted using a randomly selected key ($a = 1.6, b = 0.2, \mu = 3.75, x_0 = 0.82, y_0 = 0, 1, z_0 = 0$, iteration time $N=20$), and the resultant cipher image is shown in Figure 1(a). Figure 1(b) decrypted image using the same as encryption key (correct key), then the ciphered image is attempted to be decrypted using the same key used for encryption except for $X_1=0.82000001$, or decryption, process with only a change to the iteration time $N=21$ in decryption the results will be as shown in fig.1 (c, d), from which we can see that even an almost perfect guess of the key does not reveal any information about the plain image. Therefore, it can be concluded that the proposed image cryptosystem fully satisfies the key sensitivity requirement. Fig.1 shows that the proposed cipher scheme has enough space and is sensitive to minor changes in the key that makes it impossible to get an original image of the decoded one in case of any minor changes. They generate completely different results and decryption cannot give the correct original image.

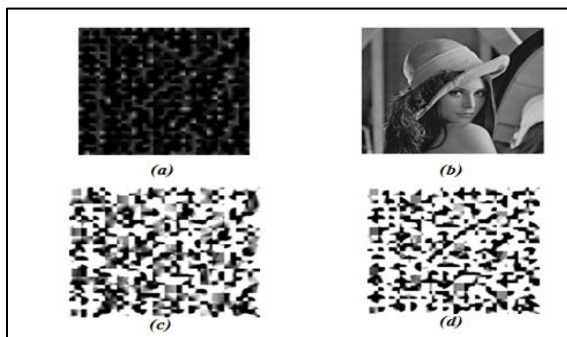


Fig.1. a) encrypted image, b) decrypted image with correct key, c) decrypted image with slight change in the one bit in the key $x=0.82000001$, d) decrypted image with change the iteration number from $n=20$ to $n=21$

C. MSE (Mean Square Error), SNR (Signal to Noise Ratio) and PSNR

In order to determine that the proposed algorithm can successfully conceal the pure image information and also enable the deciphered image to be reconstructed back to the original image without missing any information the MSE, SNR, and PSNR [14] are used for this purpose and a set of standard images are shown in fig.2. TABLE. I ,TABLE.II, and TABLE.III, shows a large results of MSE which mean that the proposed key is successful in concealing the pure image information and a small amount of SNR and PSNR means the proposed key caused large noise (i.e. small result implies better image concealment of original image).



Fig. 2. the standard image used for test

TABLE I. MSE RESULT

Image Name	MSE		
	Red	Green	Blue
Lena	15824.7	12588.4	12137.36
Pepper	15263.12	15676.95	6398.761
Fruit	25098.19	23590.67	18118.46
Camera man	22392.24	22392.24	22392.2

TABLE II. SNR RESULT

Image Name	SNR		
	Red	Green	Blue
Lena	2.299523	1.170863	1.199166
Pepper	1.589982	1.20896	1.013237
Fruit	1.598209	1.225291	1.099732
Camera man	1.230251	1.23025	1.23025

TABLE III. PSNR RESULT

Image	PSNR		
	Red	Green	Blue
Lena	7.137497	8.131199	8.289662
Pepper	7.2944	7.17822	11.07015
Fruit	6.134367	6.403363	7.549595
Camera man	4.62987	4.62987	4.62987

D. Processing Time

Processing time for encryption and decryption is also an important issue in real-time multimedia applications. To estimate the execution time of the proposed encryption scheme, different tests are performed on a PC with a 2.2 GB dual core processor and 3 GB RAM. Tests results of encryption time and decryption time are shown in table. II. We conclude from the results of input images that the proposed encryption system is of high-speed and flexible for various applications.

TABLE IV. PROCESSING TIME

IMAGE	ENCRYPTION (SEC)	DECRYPTION (SEC)
Lena	0.38411625	0.30611750
Pepper	0.39861708	0.30511745
Fruit	0.30811762	0.37991601
Camera man	0.33831708	0.3399160

VI. CONCLUSION

In this paper, an algorithm for image encryption was designed utilizing a proposed hyperchaos system. The simulation experiments showed that this algorithm was capable of achieving relatively good effects in terms of encryption and decryption methods. The proposed scheme

provides large key space, which is sensitive to slight change. The large values of MSE prove that the proposed key is successful in concealing pure image information; while the small results of SNR and PSNR indicate that the proposed key causes large noise (i.e. small result implies better original image concealment). The encryption execution time of the proposed encryption scheme is relatively fast and flexible to different applications. The future work is to employ the proposed hyperchaos system in design an authentication system.

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Automatic Approach for Word Sense Disambiguation Using Genetic Algorithms

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Abstract—Word sense disambiguation (WSD) is a significant field in computational linguistics as it is indispensable for many language understanding applications. Automatic processing of documents is made difficult because of the fact that many of the terms it contain ambiguous. Word Sense Disambiguation (WSD) systems try to solve these ambiguities and find the correct meaning. Genetic algorithms can be active to resolve this problem since they have been effectively applied for many optimization problems. In this paper, genetic algorithms proposed to solve the word sense disambiguation problem that can automatically select the intended meaning of a word in context without any additional resource. The proposed algorithm is evaluated on a collection of documents and produce's a lot of sense to the ambiguities word, the system creates dynamic, and up-to-date word sense in a highly automatic method.

Keywords—unsupervised method; genetic algorithms; word sense disambiguation; Natural Language Processing; Information Retrieval

I. INTRODUCTION

Most of the people use the web to find some contents. At searching process they never worry about ambiguities that occur between words. An ambiguous word is a word that has a lot of meaning in different contexts [2]. The context in which the ambiguous word appears is determined the sense of the word. When the person makes the search related to an ambiguous word, search engines shows all the results related to senses of the word. Several of them are relevant and others are irrelevant.

Word Sense Disambiguation (WSD) is the procedure of finding the senses of the words in textual context, when word has several meanings [5]. WSD is a long-standing issue in Computational Linguistics, and has main effect in many real-world applications involve machine translation, information extraction, and information retrieval [6]. Word Sense Disambiguation (WSD) systems try to solve these ambiguities and finding the accurate meaning.

There are two approaches of WSD [4]:

1) **Deep Approaches**: This is built on world knowledge. But such knowledge is not existing in computers readable format except in some restricted domain so this approach is not very common. However if such knowledge exist than this approach will be much more precise than shallow approaches.

Example: Man goes fishing for some bass.

Here knowledge is used to find the meaning of 'bass' in the sentence because an individual can go fishing for a kind of fish but not for low frequency sound. So here bass will refer to fish.

2) **Shallow Approaches**: This approach does not depended on the world knowledge. An individual can understand the text over the surrounding words.

Example: If 'crane' has words sky or fly near then it will refer to bird. If 'crane' has words parts or manufacture nearby it points to machine. The algorithm used in this paper considers as shallow approaches.

Dynamic languages grow by time such that even more work is required to develop new examples if new terms looked suddenly or gone. For instance, the word "rock" currently has the meaning of a stone as well as a music genre. To avoid preparing annotated corpora, effort needs to be oriented to new approaches in the knowledge-based unsupervised direction, one of the current trends to address WSD as a combinatorial optimization problem.[8].

The improvement of WSD systems is often expensive to acquiring the needed training data. The supervised methods are the best predictors of WSD difficulty, but the dependence on labeled training data limited them. The unsupervised approaches all implement well in many situations and can be applied widely. To avoid the problem of annotated corpora this paper presented an unsupervised approach based on genetic algorithms to solve the problem of ambiguity words that automatically find the sense of the word from the document's collection.

The rest of the paper is organized as follows: section II explains the related work. Section III introduces the proposed algorithm. In section IV the experimental part is presented. Section V explains the effect of some genetic algorithm factors in the performance of the proposed algorithm. Section VI explains the conclusions and section VII introduces the future work.

II. RELATED WORK

There exists important work on Word Sense Disambiguation for many languages using numerous approaches. The following paragraphs will explain some of the related work that used genetic algorithms to solve the problem of ambiguity words. Zhang, Zhou and Mmartin[3] proposed unsupervised genetic word sense disambiguation algorithm (GWSD). The algorithm using WordNet to describe possible senses for a set of words, to maximize the semantic similarity,

genetic algorithm is applied on this set of words. Domain information and conceptual similarity function is used to calculate similarity between senses in WordNet. They proposed a weighted genetic word sense disambiguation algorithm (WGWS) for word sense disambiguation in a general corpus. Experiments on SemCor are carried out to compare WGWS with previous work. Azzini, Pereira, Dragony and Rettamanzi [1] proposed a supervised method to WSD based on neural networks shared with evolutionary algorithms. Big tagged datasets are considered for each sense of a polysemous word, and used to evolve an optimized neural network. kumara and singh[9] using genetic algorithm with Elitism for Hindi language. a lexical knowledge base and Wordnet for Hindi is used. The central focus is on word sense disambiguation using the context by applying GAs. Menai[6] proposed to use genetic and memetic algorithms to solve the word sense disambiguation problem, and apply them to Modern Standard Arabic. Its performance evaluated and compared against a naïve Bayes classifier. results show that genetic algorithms can reach more precise prediction than naïve Bayes classifier and memetic algorithms. Alsaedan and Menai [11] proposed a self-adaptive GA for the WSD problem with an automatic modification of its mutation and crossover probabilities. The experimental results found on standard corpora (Senseval-2 (Task#1), SensEval-3 (Task#1), and SemEval-2007 (Task #7)) show that the presented algorithm considerably outperformed a genetic algorithm with standard genetic operators in recall and precision.

III. WORD SENSE DISAMBIGUATION USING GENETIC ALGORITHMS

Representation of the context in which an ambiguous word occurs has great effect to successfully applied machine learning methods for word sense disambiguation (WSD) problem [6]. So to apply genetic algorithms to word sense disambiguation (WSD) the search space of the problem must be represented in suitable format. The proposed algorithm applied to WSD without any resource like dictionary or thesauri, it worked directly on the collection of the documents. To start the method, words that require to be disambiguated are compiled and kept in a text file. Each word is sent to a search engine as a query to retrieve great number of Web pages. The Web pages are cleaned and control characters and HTML tags are removed. The Knowledge base contains a corpus of sentences consisting of the ambiguous words and a number of other words which co-occur with the ambiguous words. The proposed approach implemented in the following steps:

A. Preprocessing

The preprocessing procedure includes the following steps:

- 1) *Tokenize each documents and performed word segmentation to get the word set.*
- 2) *Remove the stop word like (the ,an, and ...) from the word set.*

3) *Remove the ambiguity word from the word set*

B. Applying genetic algorithm for WSD

After preprocessing step genetic algorithms applied as follows:

1) *Population Representation and Initialization*

GAs runs on a number of potential solutions, termed a population, containing many encoding of the parameter set simultaneously. To solve word sense disambiguation the population was initiated with 50 variables in length chromosome as following:

For each individual in the population do

- A. *Pick a random number representing the length of the chromosome in the population.*
- B. *Chose words randomly from the set of word resulted from the preprocessing step in number equal to the length of the individual.*
- C. *Repeat steps a and b until the population size reached.*

After these three steps are completed the population initiated with 50 variables in length individuals.

The variable length individual was used because the sense of the word may have multiple words like the possible sense edge of river of the word bank.

```
For n=1 to population size do
m=rand (1, maximum chromosome length)
For j=1 to m do
Chromosome [n][j]=word set[rand(1,word set size)]
End
End
```

Fig. 1. Pseudocode of the population initialization

1) *fitness function*

The fitness function for each individual measures how many documents covered by these words sense represented in that individual in another word how much the sense generated by the algorithm represent the actual meaning of the word in that context. The fitness function doesn't reward sense that are more frequently used.

2) *Termination of the GA*

Since the algorithm is unsupervised, there are no a specific results that should be found by the algorithm therefore, the GA terminated after a prespecified number of generations and then all individuals in the population that have fitness function more than specified threshold represented the discovered senses of the ambiguous word.

The algorithm allows overlapping so one document may have more than one sense of the ambiguous word.

```

The Input : an ambiguous word and a
collection of documents
The output : set of all possible senses of the
ambiguous word
procedure GA
begin
t = 0;
initialize Pop(t);
evaluate Pop(t);
while not finished do
begin
t = t + 1;
select Pop(t) from Pop(t-1);
reproduce pairs in Pop(t);
evaluate Pop(t);
end
end.
    
```

Fig. 2. The proposed algorithm

IV. EXPERIMENT

The method evaluated with some none ambiguous word like bank and as the Natural languages are dynamic in nature, with time new words emerge and usages of words tend to change. The method tested with the word python. Human beings can keep themselves updated with latest words or new vocabulary but to maintain same level of maintenance in updating latest knowledge for a machine needs continuous efforts, the genetic algorithm for WSD method is one of these effort. Table 1 below showed results of the system for the word **bank**, the results show that the dominant sense of the word is a financial business building and there are many senses with different fitness value and other senses have little rate of appearance.

TABLE I. SYSTEMS RESULTS FOR THE WORD BANK

Sense of word	Fitness function
Financial	70%
Building, business	60%
Services, group	30%
Financial ,Service	25%
Account ,money, security	18%
Fishmen	12%
Painting	10%
watershed	6%
sunset	4%
sand, wood	3%
right	3%
forest ,boat	1%
cutting	1%
autumn	1%
butterfly	1%

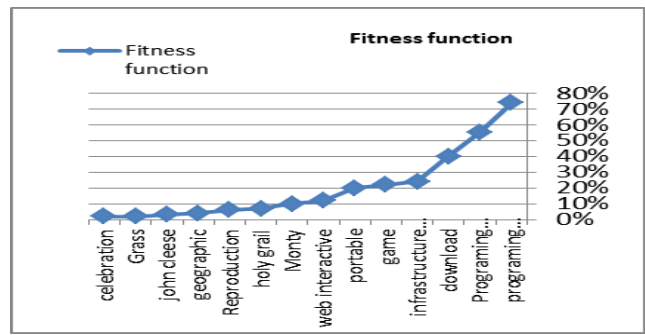


Fig. 3. The fitness of the senses discovered by the proposed algorithm for the word (bank)

Table 2 below showed results of the system for the word python, the results shows that the dominant sense of the word is programing language and the algorithm produces other senses with different values of fitness.

TABLE II. THE RESULT OF THE SYSTEM FOR THE WORD PYTHON

Sense of word	Fitness function
programing language	74%
Programing tutorial	55%
download	40%
infrastructure platform	24%
game	29%
portable	20%
web interactive	12%
Monty	10%
holy grail	7%
Reproduction	6%
geographic	4%
john cleese	3%
Grass	2%
celebration	2%

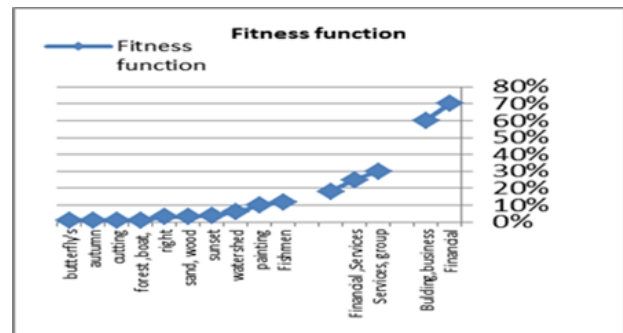


Fig. 4. The fitness of the senses discovered by the proposed algorithm for the word Python

V. EFFECT OF GENETIC ALGORITHMS PARAMTERS

In these experiments we are trying to study the effect of the genetic algorithm parameters in the performance of the proposed algorithm.

1) elitism

In these experiments we are trying to study the effect of the elitism operation in the performance of the proposed algorithm.

Genetic algorithm with the elitism operation saves the best population in the whole generation to be the solution and reject the new solutions that do not improve the existing ones.

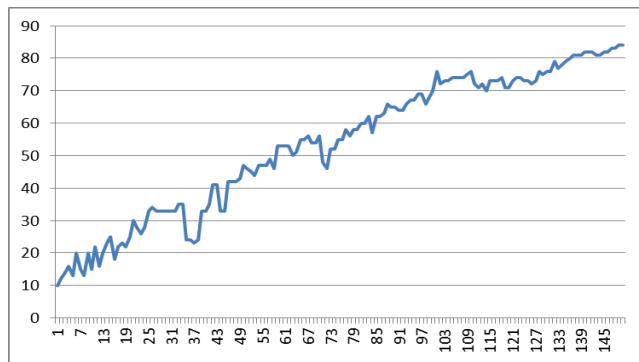


Fig. 5. Performance of proposed algorithm without elitism

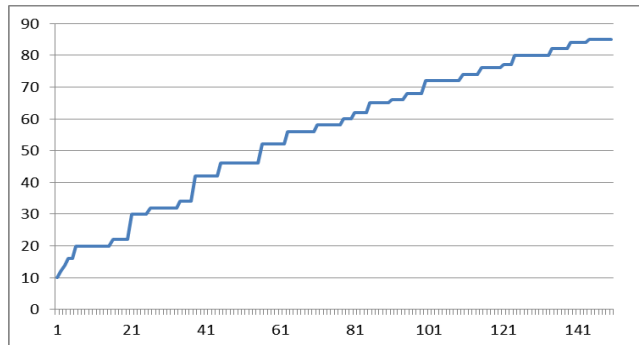


Fig. 6. Performance of proposed algorithm with elitism

VI. CONCLUSION

This paper proposed a word sense disambiguation method based on genetic algorithm. The results of the experiments reflect that the system works well in disambiguating the words without any need to an annotated example. The algorithm produced a wide range of word sense according to the context of the word in the document. Encoding the population with words taken directly from documents will be reduced the time required to reach the optimal solution, this is considered as an employment of some prior knowledge and the results can be directly introduced to the user and don't need any post processing. The experiments on the dataset collocated from the web have shown that the algorithm achieves promising results in WSD field.

VII. FUTURE WORK

In the future it will be possible to study some genetic algorithm parameters like population size and maximum chromosome length. It is also possible application of other algorithms such as swarm intelligence algorithms and compares the results with the algorithm proposed in this research.

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Hybrid Motion Graphs for Character Animation

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Abstract—Many works in the literature have improved the performance of motion graphs for synthesis the humanlike results in limited domains that necessity few constraints like dance, navigation in small game like environments or in games by the gesture of feedback on a snowboard tutorial. The humanlike cannot exist in an environment without interacting with the world surrounding them; the naturalness of the entire motion extremely depends on the animation of the walking character, the chosen path and the interaction motions. Addressing exact position of end-effectors is the main disadvantage of motion graphs which cause less importance expended to the search for motions with no collision in complex environments or manipulating motions. This fact motivates this approach which is the proposition of an hybrid motion graphs taking advantages of motion graphs to synthesis a natural locomotion and overcoming their limitations in synthesis manipulation motions by combined it with an inverse kinematic method for synthesis the upper-body motions.

Keywords—*motion graphs; inverse kinematic; virtual human; animation*

I. INTRODUCTION

Virtual humans have a wide range of applications with the developments of interactive entertainments such as games, learning and training applications or film animations. Responding to the needs of these applications, believable synthetic characters must populate the virtual environments. Character interactions and moves must be realistic to suit the visual richness of the virtual environments. To meet the interactivity demands, character must efficiently be animated and controllable by the simulation.

In literature model-based and data-driven are two semiautomatic techniques classes developed for creating human animations.

Simulation, search, and optimization are applied by model-based approaches to synthesis character motion by limiting the space of possible motion using biomechanical, kinematic, dynamic models, the representation of motion in these approaches is flexible, but may suffer from the difficulty of construction and control [11]. It was well used addressing human-like characters manipulating objects, but could fail to produce a motions with natural looking if the models do not appropriately constrain the motion particularly for motions with low energy, such as locomotion.

Data-driven approaches mainly refer to the reuse of existing 3D motion data to generate new motions. Which do not present a perfect solution either because when the motion is highly dynamic or involves constrained interactions with the

environment or other characters, the adaption of existing captured motions to new situations will mostly be difficult. To address these issues, several researchers have been proposed, the most popular approach is motion graphs where locomotion synthesis has been an active area of study. The search techniques based on motion graphs provide lowest distortion of the captured motions, and naturally allow the exploration of solutions in complex environments.

The main contribution of this paper is the proposition of a hybrid motion graphs based on the idea of using motion graphs to synthesis the locomotion motions to reach different locations in the environment and to achieve the reaching range of objects mainly needed for the forward and backward stages of the inverse kinematic method used to control the trajectories of the end-effectors in the generation of the upper body motions. Our hybrid approach proves to be very efficient for producing object manipulation in different locations in the environment with user control by allowing a coordination of hyper body actions with locomotion, and could be suitable for interactive application. Avoiding problems related to the explosion of the motion graph size in case of including both locomotion and upper-body actions which cause absolutely an increase of the search time and a complication of the search method which may fail to find the right motion in the right time, or problems related to only base on inverse kinematic methods that may fail in the generation of natural looking for low energy motions such as locomotion, the proposed hybrid motion graphs present a good solution for decrease search time also address a wide range of virtual character motions and could be useful to many applications in humanoid robotics, ergonomics, training, entertainment.

The paper is organized as follow:

In the first section, a review of works of the literature related to character locomotion, a discussion about data-driven methods, their advantages and drawbacks. Also works related to the inverse kinematic methods for the hyper body character animation. In the second section an overview on the proposed method that allows the synthesis of a natural locomotion and hyper body motions with a description of each step. In the third section a discussion of the obtained results of the proposed method.

II. RELATED WORK

A. Locomotion motions

Data-driven motion synthesis generates new motion from a library of motion clips. A lot of approaches have been proposed like blending approaches that create new motion by

blending multiple motions depending on blending weights, for synthesizing new motions that parameterize high level interesting features, it has been used to synthesize locomotion animation to lead a virtual character [9,24,26,27].

This approach have trouble to deal with the planning around obstacles, and problems in the generation of motions able to cover a sufficiently large solution space.

Motion capture data have been integrated with Planning methods in many way, Lau and Kuffner [14] use motion planning with pre-computed search trees to synthesis a long motion sequences even in dynamic environments containing many characters.

Motion clips are abstracted as high-level behaviours and then associated with a behaviour finite-state machine, motions are generated automatically by a planning algorithm that makes a global search of the state machine and computes a sequence of behaviours for the character that meet user requirements [15]. The motion database is annotated. Labels are used to describe the desired action, which is then built using dynamic programming [21].

Extracting motions from a motion graph structure is a popular approach for producing realistic full-body motions. Automatically processing motion capture data is the objective of motion graphs to facilitate the parameterization and interpolation of motions, and allow the motion planners to operate in the created parametric spaces. The motion graph structure is based on the idea of connecting similar frames in a database of motion capture clips. Once a motion graph is available, a graph search is performed to extract motions with desired properties [12,13,20,22].

The following work based motion graphs concentrated on achieving good connectivity in motion graphs by using interpolated motion clips with further similar poses than the original motion data to construct a well-connected motion graph [17], minimizing the size of the graphs by proposing an approach to select a good motion set by searching from a large motion graph that represent the motion capture data for a minimum size subgraph and using an Iterative Sub-graph Algorithm, to find a good approximation to the optimal solution [18], optimizing the search operations in motion graphs [28,29]. Mahmudi and Kallmann [20] propose feature-based motion graphs for the synthesis of locomotion motions among obstacles where transitions is chosen based on relevant features. They use a channel search method in motion graph search to avoid obstacles. Planning locomotion using motion graphs have received a lot of attention, in contrast to the manipulation of objects around obstacles because using motion graphs it is in general very hard to generate motion that needs tight coupling to the environment, like pointing to an object in the world, or walking up a stairs, unless exactly those motions are in the database. As more motion constraints are added, fewer paths will become available [7].

Yazhou [8] proposes a locomotion planner based motion graphs coupling with an upper-body actions planner that use a blending method to produce realistic motions of locomotion and interaction with the environment. The locomotion planner does a lot of iterations that required the execution of the upper

body motion each test to reach the target and in some cases it fails.

The interaction of the virtual human with the environment involves posing the character body to respect new constraints, motion graphs have the inconvenient to not precise the end effectors positions; methods of research sometimes can fail to generate the desired motion sequence because the motion corpus itself does not contain motion clips required.

B. Manipulation motions

The Animation of virtual character to realise many task and object interactions have been treated by many approaches, motion blending was used to generate reaching and grasping motions that satisfy spatial constraints [3,8].

Physics-based and procedural animation offers a great amount of control over motion by employing a mathematical model in the generation of the desired character motion depending on high-level parameters. Physically-based techniques use physic laws such as Newton's laws and conservation of energy. Determining the human motion using this techniques, require information more than joint angles and position, it need Information such as mass distribution for each body part and torques generated by each joint which is not easy specially for the synthesis of hyper body motions which involve object manipulation like reaching, grasp and displacing objects, often the Inverse Kinematics (IK) and collision avoidance are used to solve this type of problem, the sampling-based planners' methods are required when several obstacles are on the way.

Methods based on Inverse Kinematics (IK) mainly control the trajectories of the end-effectors, and can integrate some reactive collision avoidance behaviour in order to cope with obstacles. Inverse Kinematics is a method used to solve the problem of computing joint rotations in order to reach given targets with the end-effectors [19]. The analytical methods are the best choice to handle with simple kinematic chain that contains a few degrees of freedom [5]. These methods are fast but not support the complex articulated chain. Numerical methods based on Jacobian matrix are proposed to deal with complex articulated chain but suffer from the computational time, the heuristic method are proposed to solve this inconvenient [4,16,23]. Searching for a sequence of suitable hand locations in the workspace is one approach for reaching and arm planning. To reach each intermediate hand location, the IK method is used to lead the arm posture. Through interpolation of the postures, the final valid motion is obtained [11,25,30]. FABRIK (Forward And Backward Reaching Inverse Kinematics) [1,2] produces visually realistic poses with low computational cost, it bases on geometric algebra to resolve the inverse kinematic problem and support multiple chains with multiple end effectors because Constraints can easily be incorporated within FABRIK.

The virtual character is required to interact with different objects in different locations in the surrounding environment which involve posing the character body to respect new constraints, often in combination with environment navigation tasks. Motion graphs have the inconvenient to not precise the end effectors positions, this fact motivates this approach to

synthesis reactive motions in real-time, based on the use of the motion graphs in combination with an inverse kinematic method.

III. HYBRID MOTION GRAPHS

The proposed approach presents a solution to avoid many problems related to the explosion of motion graph size in case of including both locomotion and upper-body actions and preserve produced realistic motions. We propose a hybrid motion graphs taking advantage of motion graphs to synthesis a natural locomotion and overcoming their limitations to synthesis manipulation motions by combined it with an inverse kinematic method for synthesis the upper-body motion.

A. Overview of the proposed method

An overview of the proposed method is illustrated in the Fig. 1

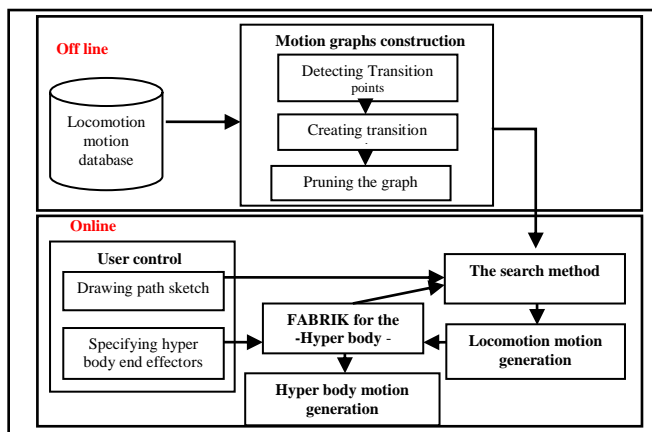


Fig. 1. System Overview

The proposed method has two phases:

1) Offline phase:

Allows the construction of the motion graphs according to Kovar and Gleicher [13] steps:

a) *Finding Candidate Transition Points:* Distance metrics are used to detect the similar frames for choosing the transition points, taking into account the relative importance of joints, changes in joint-velocity and acceleration and identifying compatible coordinate systems.

To calculate the distance $D(A_j, B_k)$ between two frames A_j and B_k , a point clouds is made over two windows of frames with length l defined by the user, every point cloud is composed of smaller point clouds that represent every frame position in the window.

The distance between A_j and B_k is calculated as follow:

$$\min_{\theta, x_0, z_0} \sum_i w_i \| p_i - T_{\theta, x_0, z_0} p_j \|^2 \quad (1)$$

Where p_i and p_j are the corresponding points in the two point clouds, a linear transformation T_{θ, x_0, z_0} formed of a rotation by θ degrees about the vertical axis and followed by a translation of (x_0, z_0) in the floor plane is applied to the second point cloud. Fig. 2 illustrates a point clouds made over two windows of frames.

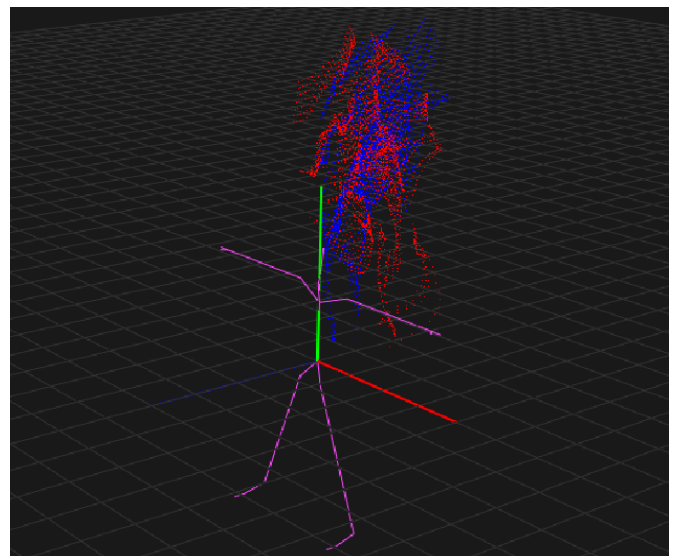


Fig. 2. Point clouds made over two windows of frames

The weights w_i may be taken to assign different importance to each joint in each frame and can be chosen by the user. Fig. 3 illustrates list of weights for each bones of an ASF/AMC skeleton [6].

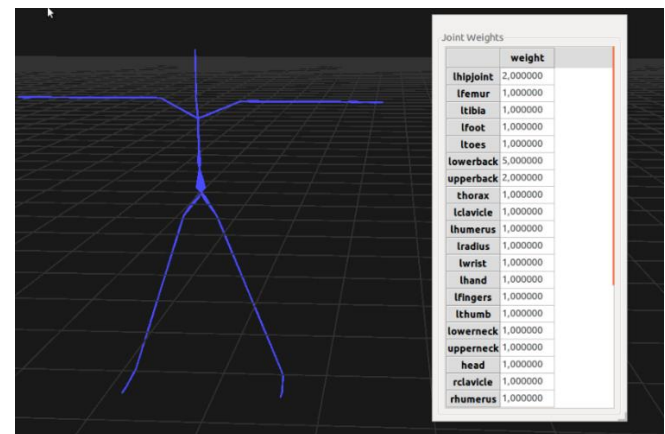


Fig. 3. ASF skeleton with list joints and weights

The distance metrics discussed above is computed for each pair of frames in the database to form a sampled 2D error function.

b) *Selecting transition points:* The distance that is found in this manner can be thought of to be a sampled 2D error or distance function. This function is normalized, so that, all the distances lie in the range $[0,1]$. We take this function and select only the local minima of this function, which gives us the candidate transition points. To guarantee that those local minima of this error function are actually good enough for transitions, a threshold must be defined by the user and the $D(A_j, B_k)$ must be smaller than it. The value of the threshold effect on the quality of the generated transitions, and the connectivity of the graph.

c) *Creating transition clips:* We have said that transitions are clips designed to connect two pieces of original

data, it is created by identifying the sufficiently similar portions of the initial clips that straight forward blending will produce a valid transitions. If the distance between frames A_j and B_k is less than the threshold and it is selected as a transition point, then we need to create the transition frames and add them as edges in the motion graph, by blending between the frames of the two windows of frames for A_j and B_k starting by applying the appropriate aligning 2D transformation to motion B. The frames A_j to A_{j+l-1} and B_{k-l+1} to B_k are interpolated as follows:

$$R_p = \alpha(p) R_{A_{j+p}} + [1 - \alpha(p)] R_{B_{k-l+1+p}} \quad (2)$$

Where R_p means the position of the root joint in frame p of the transition. The interpolation between the rotations is performed by:

$$q_p^i = \text{slerp}(q_{A_{j+p}}, q_{B_{k-l+1+p}}, \alpha(p)) \quad (3)$$

The following conditions must be fulfilled by the weight function of the interpolation

$$\alpha(p) = 1 \text{ for } p \leq -1, \alpha(p) = 0 \text{ for } p \geq k \quad (4)$$

To ensure the continuity:

$$\alpha(p) = 2 \left(\frac{p+1}{k}\right)^3 - 3\left(\frac{p+1}{k}\right)^2 + 1, -1 < p < k \quad (5)$$

To maintain the constraints of the original motion clips and ensure that the interpolation will not hurt it, the constraint will be activated in the first half of the new transition frames and passive in the second half in case it was active in the first clip but not in the second.

d) *Pruning the Graph*: In the resulted graph there may be portions with very high connectivity, and there may be portions with very low connectivity that may contain dead end nodes, which are nodes that are not part of any cycle in the graph, also contain other nodes that could be a part of one or more cycles which called sinks. These nodes may halt the motion search. To eliminate these problematic nodes we prune the graph by calculating the largest strongly connected component of the graph.

2) Online phase

The online phase allows the interactive generation of motions. This approach is similar to previous techniques that use sketching; the user draw the path sketched which will be followed by the character in the locomotion motions.

User chooses the end effectors position for the hyper body limbs before, after or during the locomotion motion which help the character to avoid obstacle (like moving head to avoid a bar) and reach objects.

The search method use a depth first search to obtain graph walks. Branch-and-bound strategy and incremental search are used to improve the efficiency [13].

The user supplies a scalar function $g(w;e)$ to evaluates the additional error accrued by appending an edge e to the existing path w The total error $f(w)$ of a graph walk is:

$$f(w) = f([e_1, \dots, e_n]) = \sum_{i=1}^n g([e_1, \dots, e_{i-1}], e_i) \quad (6)$$

Where w is comprised of the edges e_1, \dots, e_n . We assumed that $g(w,e)$ is never negative Therefore, the total error cannot decrease by adding an edge to a graph walk.

f is evaluated for all graph walks using a depth-search, then the best graph walks is selected.

We use branch and bound algorithm in which we prune any graph incapable of producing an optimal graph walk.

We apply path synthesis that is generating a motion stream that follows a user specified path P . The idea is to estimate the actual path P' travelled by the character through a graph walk and measure how different it is from P [13].

To compare the original paths P and the generated path P' , at each frame, the root joint of the character is projected on the floor giving a linear curve. Then calculate

$$g(w,e) = \sum_{i=1}^n \|P'(s(e_i)) - P(s(e_i))\|^2 \quad (7)$$

$P(s)$ is the point on P with arc-length distance from the start of P is s . The completeness condition of P' is if P' is sufficient long or surpasses the length of P .

Fig. 4. Illustrate how to extract the motion that travel with the path specified by the user

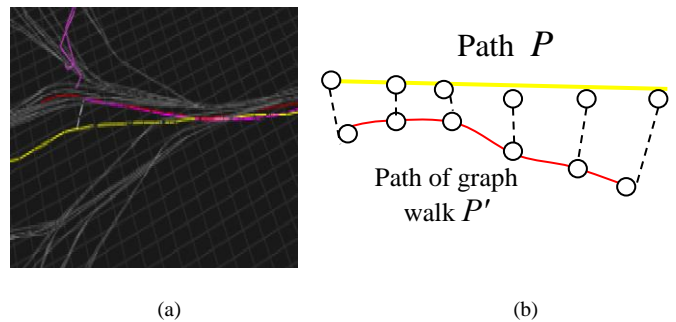


Fig. 4. (a), (b) Extract motion that travels along a path

For the hyper body IK method, the virtual character is modeled as a chain composed of rigid links attached at their ends by rotating joints, the chains can be formalized as follows: All joints with no children are labeled as end-effectors. FABRIK uses points and lines to solve the IK, it divide the problem into two relevant phases, a forward step and a backward steps, and support all the rotational joints limits and joints orientations by repositioning and re-orienting the target at each step.

The structure of the human skeleton is divided into smaller kinetic chains which are then sequentially adapted into the body posture in a hierarchical order [1,2]. Fig. 5 shows the hierarchical structure of the hyper body used in the approach.

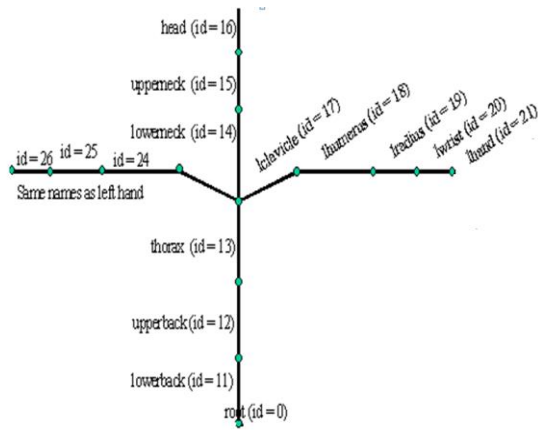


Fig. 5. Bones of the skeleton applied within the IK method[6]

The FABRIK method may be applied starting from the right hand (end effectors) until the root (pelvis), from the left hand (end effector) until the root, from the root until the head

When synthesizing reaching motion using the standard FABRIK, the focus tends to be on arms control, if the target is not reachable because the distance between the target and the first joint (root) is higher than the summation of the length of each links , the method do not provide a solution . However stepping motion can be a solution because humans often step in the direction of the target in order to help their reaching action. In doing so, the target falls within the convenient reaching range ,in this approach this solution is provided by allowing the search method to place the character near the target location and ensure that the target is reachable by identifying a reaching range H where

$$H = D - \sum_{i=0}^{i=n} d_i \quad (8)$$

D is the distance between the target and the first joint (root), d_i is the joints distances of the chain.

To follows the user specified sketch presented by a 2D path planner on the floor plan, the branching factor of the search must be limited to only allow expansions nearby this path, else when the users choose to manipulate an object which is far from reaching, an extensions is integrated to customize the locomotion toward the object location.

The extension is a new root position that insures that the target is in the reaching range. Whenever the locomotion search expansion generates a character position that is close enough to the target the forward and backward stages are applied. An example of the unrolling of the motion graph with the expansions obtained is illustrated in Fig. 6.

To show the progress of the IK method an example of eight link manipulator is used Fig. 7, which is similar to the virtual human hands; the same principal is applied for the head but for fewer links (joints). The initial pose of end effectors and target position and each point is expressed by the coordinate of Cartesian space.

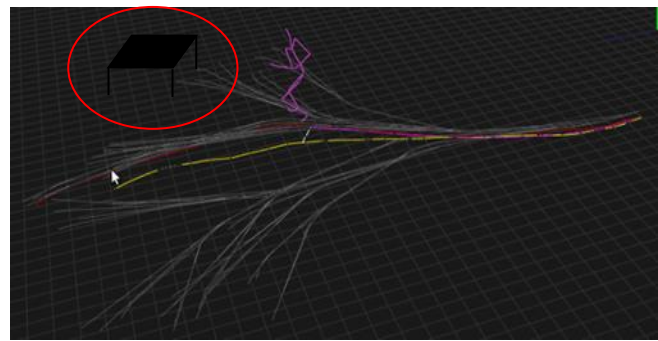


Fig. 6. Unrolling of the motion graph with the expansions obtained, white line expansions used to branch toward the reaching range (red circle)

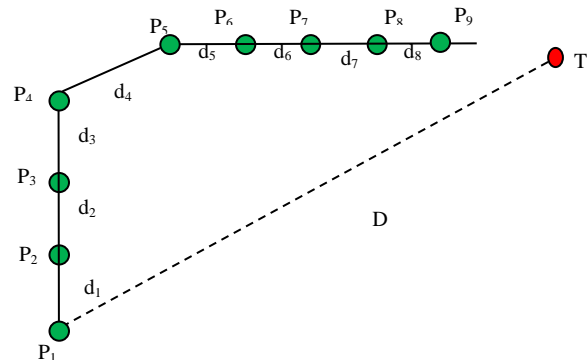


Fig. 7. example of eight link manipulator

The Fabrik [1] algorithm is extended as follow:

Input :

The target position T

The joints positions P_1, P_2, \dots, P_9

Distances between each joint d_1, d_2, \dots, d_8

Output : The generated joint positions for P_1, P_2, \dots, P_9

- 1- **Distance test** Check that the distance between the target T and the first joint P_1 is within the summation of length of each link.

$$D = P_1 - T$$

If $D \leq (d_1 + d_2, \dots, +d_8)$ target is reachable

2- run *Forward* stage

- (a) Move the hand P_9 end effectors to the target position, and the wrist point P_8 .

P_8 was moved on the line from P_9 to create P'_9 and P'_8 .

The distance between P'_9 and P'_8 is the same as d_8

- (b) Continue step (a) until new P'_1 is reached

3- Run *Backward* stage

- (c) Move P'_1 to its initial position P_1 .

- (d) Repeat procedure as (c), and P''_9 moves closer to the target than P_9 .

Run the *Forward* stage until the end-effectors is satisfactorily close to the target.

If $D \geq (d_1 + d_2, \dots, +d_8)$ target is not reachable

- 1- Calculate the new position P^*_1 of P_1

- (e) Calculate the threshold reaching range

$$H = D - (d_1 + d_2, \dots, +d_8)$$

- (f) Branch to H
- 2- run *Forward* stage
- 3- Run *Backward* stage

End

IV. RESULTS

In this section we investigate the results of the performed experiments relative to the automatic construction of motion graphs based on connecting between different motion clips in the database and then searching in the resulted graph for motions that satisfy user constraints using the procedures discusses above and the interactive generation of realistic controllable locomotion motions for character locomotion, other results demonstrate the hyper-body motions adjustment to reach different location using the inverse kinematic method.

For the locomotion motion the graph is automatically generated from an input set of motion clips like running clips, sharp turns used to animate a human-like character in a virtual environment. In experimentation the time to build the motion graph was 40 seconds for an example of a motion graph formed over 8 motion clips. Fig. 8 illustrates the motion clips list, the weight of each joints and the generated point cloud of the distance metric.

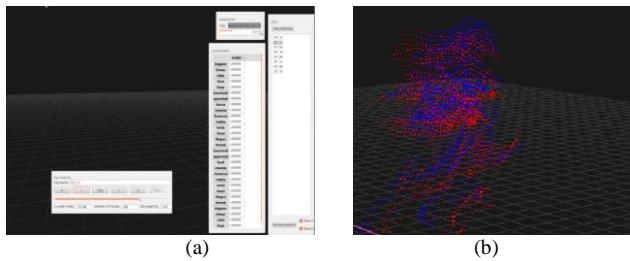


Fig. 8. (a) Motion clips list and joint weights (b)Point cloud distance metric

For the path synthesis the user first draw a sketch using the mouse, this sketch will be followed in the phase of generation of the new motions Fig. 9

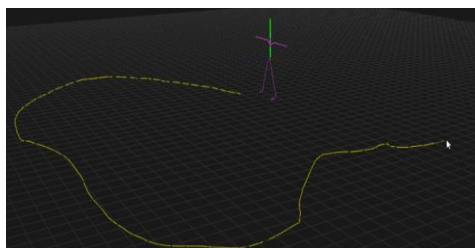


Fig. 9. Example of a user sketch

Fig. 10, illustrates the path synthesis and the generation of the new motion following the yellow line represents the user sketched path and the white lines represent the edges in the graph walk and red line represents the actual path followed by the character. The hyper-body pose are adjusted to new constraints using the inverse kinematic method. Fig. 11 and Fig. 12 illustrate right and left hand adjustment using the IK method.

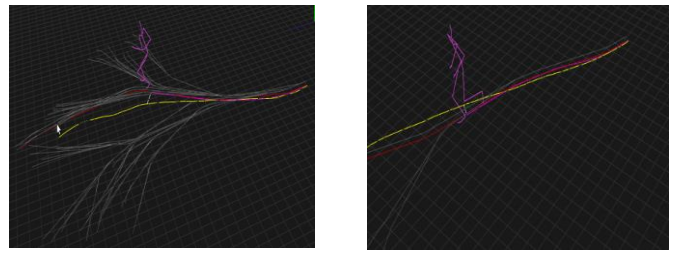


Fig. 10. Example of a motion extracted from a motion graphs , (a) from 8 motion clips (b)Another example with less motion

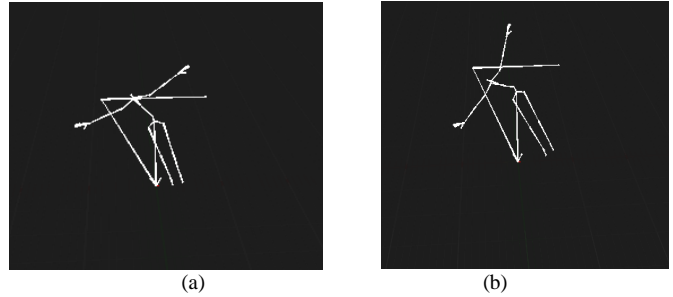


Fig. 11. (a),(b) Right hand adjustment using the IK method

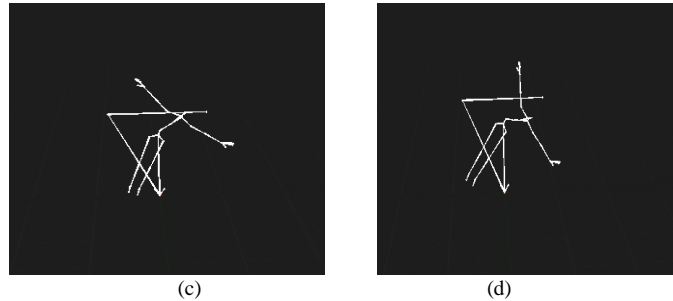


Fig. 12. (c),(d) left hand adjustment using the IK method

Hybrid motion graphs is a framework for generating realistic, controllable whole-body motion through the combination between two techniques, motion graphs and Inverse kinematic method to resolve the problems related to including both locomotion and upper-body actions in the same graph from the explosion of the size of the graph, the increase of the search time and the complication of the search method, also problems related to the fail in the generation of natural motions in case of using just inverse kinematic methods.

V. CONCLUSION

In this paper we have proposed a hybrid motion graphs for generating realistic whole body motion taking advantages of motion graphs to synthesis a natural locomotion and overcoming their limitations to synthesis manipulation motions by combined it with an inverse kinematic method for synthesis the upper-body motion.

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Toward a Hybrid Approach for Crowd Simulation

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Abstract—We address the problem of simulating pedestrian crowd behaviors in real time. To date, two approaches can be used in modeling and simulation of crowd behaviors, i.e. macroscopic and microscopic models. Microscopic simulation techniques can capture the accuracy simulation of individualistic pedestrian behavior while macroscopic simulations maximize the efficiency of simulation; neither of them assures the two goals at the same time. In order to achieve the strengths of the two classes of crowd modeling, we propose a hybrid architecture for defining the complex behaviors of crowd at two levels, the individual behaviors, and the aggregate motion of pedestrian flow. It consists of combining a microscopic and a macroscopic model in an unified framework, we simulate individual pedestrian behaviors in regions of low density by using a microscopic model, and we use a faster continuum model of pedestrian flow in the remainder regions of the simulation environment. We demonstrate the flexibility and scalability of our interactive hybrid simulation technique in a large environment. This technique demonstrates the applicability of hybrid techniques to the efficient simulation of large-scale flows with complex dynamics.

Keywords—crowd behavior; micro-scale representation; multi-layered framework; real time simulation

I. INTRODUCTION

Simulating the movement of pedestrian crowd is a field of research that receives significantly growing interest during the last years, due to their vast domains of application: movies industry and entertainment, security, emergency training, architecture, and many others [5, 37]. Several models for crowd simulation [23, 37] have been proposed and many efforts have been made to modulate intuitive navigation control and real time crowd behavior simulation, these models were classified into two categories: microscopic and macroscopic [25]. There are two conflicting requirements in crowd simulation.

The first major problem is that the most applications require a human crowd simulated in real time, with higher level of detail and an accurate realism of behaviors. Then, there is still a clear relationship between the accuracy realism of crowd behaviors and the computational costs of simulation. Satisfying these both constraints at the same time is particularly a challenge of great importance [21]. The majority of the previous models have a limited ability to response to the latter problem, they tend to focus on a single factor; there is no existing method that is able to reduce the computational cost while maintaining the high level detail of simulation. Almost all the existing models were agent-based (microscopic models). This approach describes the most natural way to

simulate crowds as independent autonomous pedestrians interacting with each other; it usually handles local collision avoidance and global navigation for each person. However, these kinds of models have the drawback that when animating a large crowd, they are computationally intensive. Microscopic models give more accurate results only for smaller crowds to achieve real time simulation. On the contrary, the macroscopic models are usually created to realize a real time simulation for very large crowds; they follow the features of the flow as long as the overall crowd behavior seems realistic. These models offer a coarse-grained simulation result with higher execution efficiency which is due to the lack of concerns on individual issues [25].

Finally, modeling the movement and behavior of the virtual crowd remains a major challenge as highly dynamic complex systems [10], the crowd is a large group of pedestrians with non-uniform spatial distribution and heterogeneous behavior characteristics, and it exhibits often distinct characteristics, such as independent behaviors, self-organization, and pattern formation, due to interactions among the individuals. Previous works have suggested that human crowd dynamic can be modeled on many different scales [25, 10], from coherent aggregate behaviors of the crowd on the largest scales to the individual behaviors, interactions among individuals on the small-scale detail. Such multi-scale systems are computationally expensive for traditional simulation techniques to capture over the full range of scales [31].

To overcome these two conflicting goals, we assume a scalable simulation is required to handle at least several hundreds or even thousands of pedestrians, running in real-time, particularly with respect to the complexity of the environment and the realism of behaviors required by the crowd, we investigate to find a good balance between visual credibility of complex crowd behaviors and computational requirements, where the behaviors of human crowd can be viewed on a two different level of detail: from the chaotic, fluctuating interactions between individual objects on the finest scales, to the coherent aggregate flow of the pedestrian crowd on the largest scales.

Our solution consists to introduce a hybrid simulation architecture that combines the strengths of two classes of crowd modeling to achieve flexible, interactive, high-fidelity simulation on large environment. This architecture couples a microscopic model of individual navigation [27] with a novel continuum approach for the collective motion of pedestrians; it can apply to simulate the behaviors and movement patterns of extremely large crowds at near real-time rates on

commodity hardware. Our approach is able to determine by itself the most suitable model of modeling for each region in the environment, regarding the simulation context, in real time and within a continuous environment. To do so, we first introduce the generic notions of dynamic change of representation, and we describe our method for handling the transfer of pedestrian between continuum and discrete simulation areas and discuss how the constituent simulation components are adapted to handle this transition. Then, we evaluate this approach experimentally along two criteria: the impact of our methodology on the computational resources, and an estimation of the dissimilarity between a full microscopic simulation and a simulation with our methodology. Finally we discuss the results obtained and propose enhancements for future works.

The remainder of this paper is organized as follows. First, the related works on the simulation behavior of virtual crowd are introduced to give the readers some background information. Then, section 3 describes our hybrid framework for simulating the movement of crowds. The macroscopic and behavioral models are introduced in section 4 and section 5 respectively. We describe a strategy that allows dealing with the interaction and the online-switching of simulation models for crowd behaviors studying. Section 7 demonstrates the simulation results in several common scenarios. Finally, the conclusion section outlines the directions for future work.

II. RELATED WORK

Crowd simulation has been the subject of studies of computer animation for a long time, there is a large amount of relevant work in the pedestrian dynamics, path planning and navigation [10]. In this section, we highlight some of recent models designed for simulating the real time navigation of large number of pedestrians in complex dynamic environments. Depending on the levels of simulation resolution, the existing models are largely classified into two extreme categories [25], i.e., continuum macroscopic models [28, 18], and individual-based microscopic models [9, 1].

The microscopic scale of modeling typically involves a detailed design and considers each pedestrian as an autonomous individual that has its characteristics and intentions, and interacts with other pedestrians. A microscopic model normally engages the local behaviors and emergent phenomena which are based on changes in pedestrian surrounding environment. These models include the social force model [30, 9], cellular automaton model [1] and rule based model [7, 8].

A large body of other studies from a variety of scientific fields [37] has developed to simulate various aspects of crowd movements and dynamics. Musse and Thalmann [32] proposed a model for crowd simulation with hierarchically structured crowds having different levels of autonomy. Their model is based on groups, rather than individuals: groups' are more intelligent structures, where individuals follow the group's specification. A cognitive model of crowd behavior has been proposed [13] by applying the Festinger's (1954) social comparison theory, which is the general process underlying the social phenomena, to simulate pedestrian movement in a simple virtual environment. Pelechano et al.

[24] described a multi agent model called HiDAC model to simulate the flow of a high-density population in the dynamic virtual environment. It is further expanded by adding OCEAN personality model to demonstrate personality influences on the crowd motion [12]. Shao and Terzopoulos [35] used perceptual, behavioral, and cognitive models to simulate individuals.

The local navigation model of crowd alone cannot realistically simulate the behavior of pedestrians that try to reach some specified goals. Consequently various behavioral approaches have been combined with global navigation [30, 23, 36]. Sung et al. [19] proposed a model for developing situation-based crowd behavior based on combining probabilistic roadmaps with motion graphs to find paths and to compute detailed locomotion to steer characters to a goal. Guy et al. [34] proposed a new algorithm called "PLEdgestrians" for simulating large crowds at interactive rates based on the Principle of Least Effort; this approach calculates a biomechanically energy-efficient, collision-free trajectory for each heterogeneous individual. In [16], a two level navigation method was introduced where the velocity obstacle concept from robotics is adapted to model human interactions. These geometric formulations are often based on (Reciprocal) Velocity Obstacles (RVO) and have been shown to exhibit many emergent crowd phenomena.

For simulating the realistic navigation in large and complex environments, a hierarchical representation is presented to handle the fast path planning [33]. Jiang et al. [14] proposed a semantic model for modeling multi-layered environment, where the semantic information is described by three different levels: a geometric level, a semantic level, and an application level. And each level contains different maps for different objectives. Lamarche and Donikian [11] presented a hierarchical path-planning and a reactive navigation algorithm based on topological pre-computation structure of the geometric environment. Pettré et al. [15] proposed a novel navigation approach which extracts automatically a navigation graphs from geometric model of an uneven and multilayer virtual scene and applied it to deal the path planning problem using a navigation graph. Sud et al. [4] present the Multi-agent Navigation Graph (MaNG) which is formed by the intersection of the regions of the first order Voronoi diagram with the second order Voronoi diagram. In [3, 29], Sud et al. suggested Adaptive Elastic Roadmaps (AERO), which are contain elastic edges that can modified along with the environment using particle-based dynamics simulators. Yersin et al. [6] present a hybrid approach for real time crowd motion planning. A navigation graph is used to divide the environment into zones of varying interest. The potential fields are used only for those parts of a navigation graph that lie in a high interest region. In the other regions, the pedestrian behaviors are ruled by the navigation graph and a short term avoidance algorithm.

A macroscopic level modeling considers an overall situation of a large crowd without taking into account local interactions; it is concerned with group behavior and deals with a crowd as a whole or fluid in the form of partial differential equations formulating the relationship between continuum density flow and average speed [28, 2]. Hughes

[26] implemented a continuum approach inspired from fluid dynamics to simulate the two dimensional pedestrian flow, this approach was later extended by Treuille et al. [2], who presented the continuum model in order to generate realistic motion planning for crowds. Their method calculates the potential function for a group once, and then it simultaneously derives the optimal paths for all group members. Recently, Huang et al. [18] revisited a dynamic macroscopic model of pedestrian flow proposed by Hughes and advanced an efficient solution algorithm. Jiang et al. [14] used the continuum model to simulate plausible crowd movement within multi-floor structures, such as a large pedestrian overpass, a subway station and an office building.

Various works have proposed to combine different models together to build a hybrid model. Anh et al. [26] propose a hybrid approach that combines a macro and micro models to simulate the pedestrian movements in the study of evacuation problem on a road network. They use an agent based Leader-Follower approach to simulate the pedestrian behaviors in the cross-section (where a decision must be taken), and the movements in the straight parts of the streets are calculated by the LWR-model. A multi-resolution method is proposed in [20]; it consists to incorporate both complete macroscopic and microscopic models and executes them inter-changeably. In this approach, the macroscopic model can only be executed when crowd movement is mostly stable (or becomes stable eventually). In [22, 21], a hybrid modeling method is proposed for crowd simulation, which combines macroscopic and microscopic models in a single simulation and executes them simultaneously by applying them to multiple partitions of a corridor. However, global and individual issues cannot be simultaneously reflected for any simulation area. Current hybrid models have two common weaknesses. The key component of the hybrid simulation model is to define how the two different types of simulations are combined together; this consists to limit an arbitrary number of interface points on elected regions of the boundary (e.g., doors, cross-sections) where the pedestrians under one type of simulation must be transitioned into the other. Therefore, a transition is not possible on the whole border of a model. Another weak point of state-of-the-art hybrid modeling is the inflexibility of dynamically-exchanging the combined models. There is no broad framework which supplies a method to adaptively change the pedestrian dynamic models in a region as needed to observe certain phenomena.

III. OUR MODEL

We propose a hybrid framework for real time simulation of pedestrian dynamics and movement patterns of huge crowd in a complex virtual environment. This solution preserves the granularity of simulation at the individual level to capture individualistic pedestrian behaviors, and at the same time is scalable and can richly exhibit emergent behaviors of dense crowds. The problem that we still try to address is both achieving realistic simulation and scaling up in terms of the number of individual that can plan their motion in real time. Our proposed architecture consists to integrate a more detailed approach with a coarser model, describing the individual pedestrian behaviors and crowd dynamics, within an unified crowd modeling framework, and execute them simultaneously

in different regions of the virtual environment. There are several reasons which have motivated our decision of a proper coupling of two philosophies of modeling focusing on different level of detail (discrete individuals and crowd as a whole) executing to produce most visually pleasant simulation.

First, pedestrian crowd is a multi-scale phenomenon, which can be described at both macroscopic level (continuous medium) and microscopic level (granular medium), in many scenarios, it is need to have pedestrians behave individually, continuously interacting with other pedestrian while trying to reach their own objectives, thus the behaviors of each pedestrian must be treated more precisely. In other situation, pedestrians' flows demonstrate some striking similarities between pedestrians' behaviors and particle flow dynamics. Hence the flexibility of combining two models of different level of detail is examined to capture and characterize the almost aspects of the crowd dynamic. Macroscopic models allow a better overall understanding by regarding the crowd system as a whole rather than on the details, and are usually designed to achieve a coarse grained simulation executing in real-time for very large crowds as long as the overall crowd movement looks realistic. Microscopic models focus on individual behaviors including pedestrian's psychological and social characteristics, interaction among pedestrians, and complex cognitive behaviors. Although microscopic models are very accurate only for modeling smaller crowds to achieve real-time simulation, they can simulate pedestrian in a crowd with more realistic individual behaviors.

Secondly, a multi-methods simulation can give a good equilibrium between computing resources and simulation properties, such as realism, coherence and complexity. The microscopic models can generate a fine-grain simulation in more detail than the macroscopic method. However, they have high computational and memory needs. The macroscopic models can save resources but tend to give less accurate results. Mixing both types of models can hopefully allow combining the strengths of both classes of crowd modeling to achieve flexible, interactive, high-fidelity simulation on large virtual environment. However, this fundamental choice leads to several challenges which can be classified along two axes. The first key important of our hybrid technique is related to the execution of the models themselves. We must precise how the two different types of simulation will be used (the two models executes interchangeably or simultaneously), and the way they will coupled together. Finally, the second challenge focuses on the needs to manage the transfer of pedestrians between macroscopic and microscopic areas and we must discuss how the constituents of our hybrid framework are adapted to handle this transition. With the aim to achieve these two fundamental challenges, in our hybrid, multi-method technique we divide the simulation environment into multiple disjoint (and not necessarily connected) areas each area is ruled by either microscopic simulation or continuum simulation. These mutually exclusive regions are dynamic, we can adaptively change the simulation method used in a specific region as needed cording to its density, meet performance requirements, or to observe certain phenomena (individual behaviors or crowd movements).

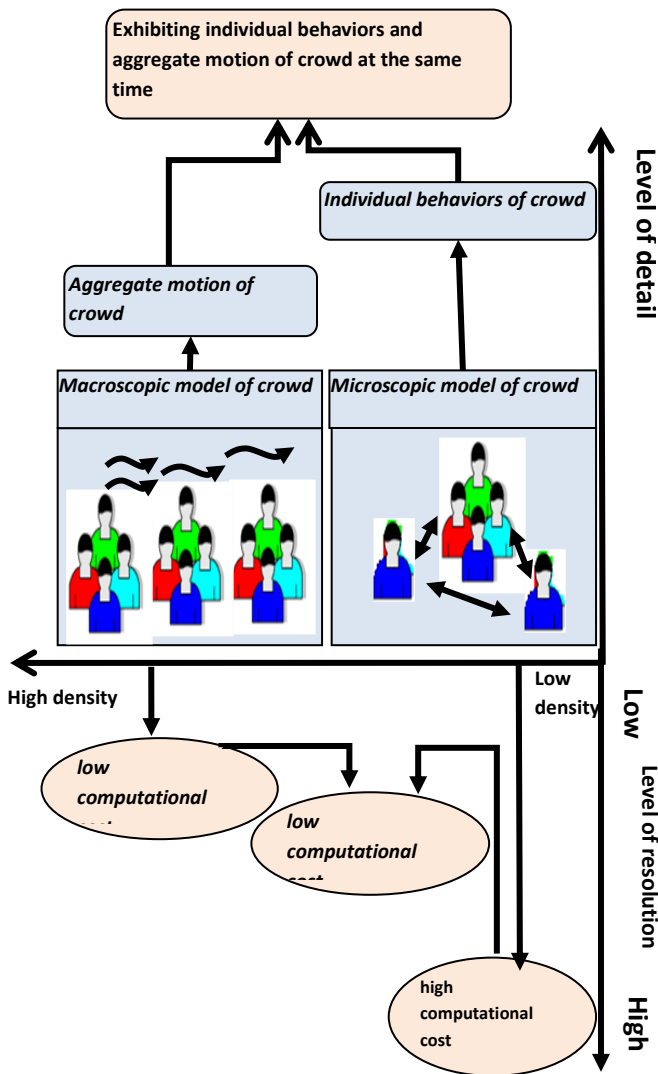


Fig. 1. Overview of hybrid model for real time crowd simulation

In the zones of high density of crowd the individual behavior is less critical, we assume to govern the crowd behaviors by a continuum approach to exhibit the aggregate motion and to describe behaviors similar to granular flows. On the other regions of low density, a realistic behavioral model is used to microscopically model the pedestrians, in these zones the standard representation of a crowd as a large group of pedestrian is used and individual behaviors is particularly relevant to the modeling process.

A challenging goal is to model these interactions in large crowd simulation systems at interactive rates. The areas of low density are exploited by a more realistic and accurate but expensive model, while computation time saved by using simpler and faster modeling approach in other regions. Our architecture also ensures that no visible disturbance is generated when switching from one model to another. To achieve this, we must be able to convert discrete pedestrian from microscopic simulation regions into the aggregate format necessary for continuum simulation (or vice versa).

IV. ENVIRONMENT MODEL

The environment in which the simulation takes place is the surrounding of the pedestrians, where they move along, interact and navigate to get from one location to another, typically, it includes walk-able areas, obstacles of different natures, and destination. Whereas, fixed obstacles can be defined as regions that no pedestrian can access, moving obstacles are other pedestrians occupying predefined space from the environment which is consequently not anymore available [34].

The first step in designing a crowd navigation system is to construct an efficient abstract representation of the virtual environment where the pedestrian can rapidly perform way-finding. We define a representation method which handles two types of structure data to clearly represent and to organize the topological relationship among the different geometrical areas of a large complex environment. This approach provides a well consistence resulting from the continuous interaction between two models of different level of detail.

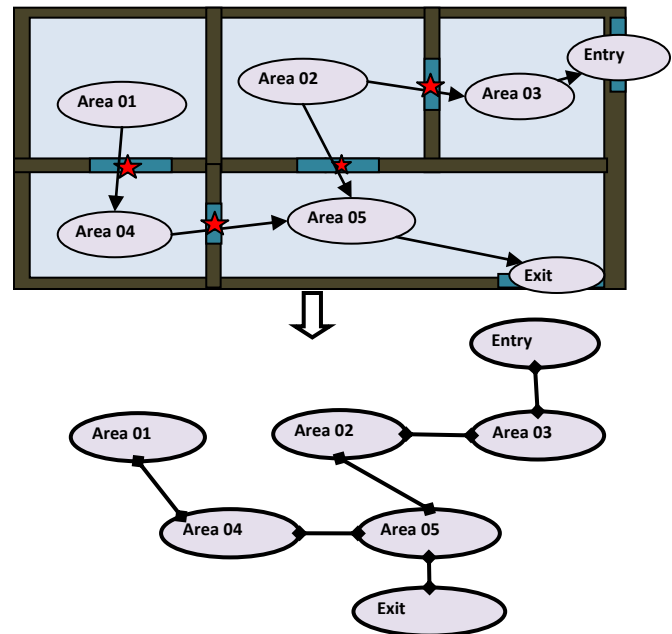


Fig. 2. Topological graph representation of the virtual environment

A. Topological graph

Usually, the virtual environment is defined by a 3D model to constitute a geometric representation of the real world. Such representation of the spatial data makes it difficult to handle by virtual pedestrian in order to find its own paths through the environment. The most way to facilitate this representation is to obtain the topological relation of the environment and its geographical areas captured in a graph based structure (Fig.2).

The topological graph uses nodes and edges to indicate the adjacency, connectivity, the inclusion and the intersection between the different parts of the environment, in which the node defines spatial areas and the possible path can be defined as edge. The internal spatial areas can be defined as a bounded volume in 3D space (such as a room, a corridor, a flight of

stairs or even an entire floor) with bottom flat that contains several objects inside it (e.g., ground, walls, benches).

B. Layered model for environment representation

The second form of space representation is to use the layer structure (Fig (3)). We identified three independent layers related to the model used for simulating the behavior of virtual crowd, each of which contains both static and dynamic data. Then we implemented the following layers for representing each spatial area in the environment:

- **Regional layer.** In this level, the whole walking space of one spatial area is divided into a number of unique continuous sub-areas. For generating these sub-areas, one main requirement must be valid which is: these subareas must be exhaustive; two different must not cross each other. This level is used to precise which model must be used for movement modeling, macroscopic model in the subarea with high density, and the microscopic in the other sub-regions.
- **Coarser layer.** The surface of sub-area, defined in the regional level, is assumed to be divided into cells; we do not limit ourselves to a maximum of one pedestrian per cell, in contrast, we consider each cell's size to be sufficiently large enough to contain at least 25 individuals of average size, where individuals placed on the same cell do not overlap.
- **Finer layer.** Every cell from the coarser layer is further divided into a uniform lattice of cells, each representing a portion of the simulated environment and comprising information about its current state, both in terms of physical occupation by an obstacle or by a pedestrian. The size of the subareas could be reduced to the average space occupied by a single pedestrian.

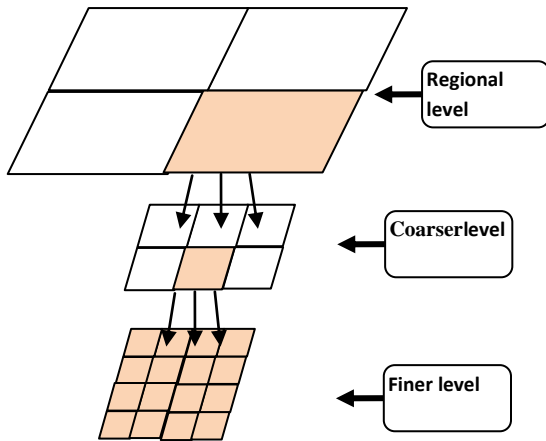


Fig. 3. Hierarchy model for environment representation

V. MACROSCOPIC MODEL

Macroscopic models deal with the whole crowd; they often use an analogy with fluid or gas dynamics to describe the evolution of crowd density and velocity in time by using partial differential equations [28, 18]. These models can be of value in simulating large-scale crowds and highly concentrated populations in spots such as stadiums, shopping malls, and subways. In the following sub-section, the

development of the continuum model for modeling the pedestrian behavior problem is discussed.

Throughout the development of this continuum approach of simulation, different equations have been proposed to model the pedestrian movement in order to use the continuum model as a tool of analysis.

We refer that pedestrians waking in a 2D continuous surrounding environment are capable to find a specific path describing by its minimal travelling cost to achieve the goal. This can be express by an eikonal type equation [38].

A. Governing equation

Similar to other physical systems, the conservation law appears to be more effective when it charged to describe the crowd dynamics by evaluating the velocity, density and flow of pedestrian. The conservation of mass for pedestrian flows is defined as follow [39]:

$$\rho_t(x, y, t) + \nabla \cdot F(\rho, x, y, t) = 0, \quad \forall(x, y) \in \Omega, t \in T \quad (1)$$

$$\rho(x, y, 0) = \rho_0(x, y), \quad \forall(x, y) \in \Omega \quad (2)$$

Where $\rho(x, y, t)$ (ped/m²) is the time-varying pedestrian density; Ω denotes a 2D continuous walking facility; T is the time horizon of analysis.

$$F(\rho, x, y, t) = (f_1(\rho, x, y, t), f_2(\rho, x, y, t))(\text{ped}/(\text{m} \cdot \text{s}^{-1}))$$

is the vector of the flux in the walking facility and f_1, f_2 represent the flow in x- and y-directions, respectively; $\|F(\rho, x, y, t)\| = \rho(x, y, t)U(\rho, x, y, t)$, where $U(\rho, x, y, t)$ (m/s) denotes the average walking speed of pedestrians at location (x,y) at time t which depend on the density and the free speed of pedestrian movement. Flow intensity or the flow-density relationship, $\|F(\rho, x, y, t)\|$, is defined as

$$\|F(\rho, x, y, t)\| = \sqrt{f_1^2(\rho, x, y, t) + f_2^2(\rho, x, y, t)}$$

B. Desired direction of crowd motion

It is assumed that a pedestrian flow at location (x,y) $\in \Omega$ prefers to reach its goal as fast as possible, which implies that it selects the shortest path to move along. This path minimizes its individual cost potential to the destination, based on the instantaneous information that is available at the time decision making [40].

$$C(\rho, x, y, t) = \frac{F(\rho, x, y, t)}{\|F(\rho, x, y, t)\|} + \nabla \Phi(x, y, t) = 0 \quad (3)$$

Let $\Phi(x, y, t)$ be cost potential which is used to describe the minimal travel cost of pedestrians from location (x,y) $\in \Omega$ to the destination, i.e. it consists to record a value for each cell in the simulation environment where (x,y) is the coordinate of the cell location. The value is assigned to 0 and positive infinity for cells in part of a goal exit and obstacle, respectively. Values for the other cells are assigned as positive values, which are determined during the model execution. $\Phi(x, y, t)$ is defined by the following Eikonal-type equation:

$$\begin{aligned} \|\nabla \Phi(x, y, t)\| &= C(\rho, x, y, t) \quad \text{in } \Omega \\ \Phi &= 0 \quad \text{on goal} \end{aligned} \quad (4)$$

$C(\rho, x, y, t)$ (s/m) is the generalized local cost function which is assumed to depend on the local operating conditions in the walking facilities, i.e., the tendency to avoid high-density regions. It takes the specific form [38, 39]:

$$C(\rho, x, y, t) = \frac{1}{U(\rho, x, y, t)} \quad (5)$$

C. Crowd speed selection

It is necessary to characterize how the magnitude of crowd velocity, i.e., the speed of crowd movement, changes with the crowd density. An exponential form of speed-density relationship is applied in the macroscopic model, as shown:

$$v(\rho) = v_{\max} e^{-\alpha \left(\frac{\rho}{\rho_{\max}}\right)^2} \quad (6)$$

where V_{\max} is a free flow speed, ρ_{\max} is a congestion density [39].

VI. MICROSCOPIC SOLUTION FOR CROWD SIMULATION

Our fundamental objective is to propose a behavioral approach, belonging to the microscopic philosophy, lies to reproduce the realistic navigation behaviors and the motion planning of crowd, so it also take into account the individual's detailed characteristics and behaviors of virtual pedestrian to provide a real time execution of coherent and interesting complex behaviors resulting from the evolution of system. This proposed solution should be able to simulate a large number of virtual pedestrian that behave like a large group of humans in the real world.

Obviously, the pedestrians are the strongly part of the simulation. A virtual pedestrian is defined as a fully autonomous entity which moves in a virtual environment with its own goals to achieve. It should look and behave in its own specific style as a real human in the life world, and be able to interact with the environment and its inhabitants in a seemingly intelligent way.

Our initial assumption is that the virtual pedestrian needs to be familiar with its surrounding because it has a mental map of the areas to be navigated and has a specific personality. During the navigation process, the pedestrian finds the shortest path and plans its movements to reach as specific destination as quickly as possible while avoiding bumping into other people, or tripping on an obstacle. As the pedestrian navigates, it has the capabilities of perceiving the virtual environment around them, analyzing environmental situations, and is exhibited natural behaviors. They should exhibit complex motions in large and dynamic environments. This process is repeated until it reaches the goal position. We will address all these aspects of a pedestrian to look natural.

Our solution to tackle all these issues consists to design a layered framework that is scalable and efficient to target the real time simulation for low to medium scale crowd of people, where each person corresponds to a simulated pedestrian endowed with its own individual behavior. Particularly, this layered architecture is also easily expandable to incorporate more and more broad range complex and realistic behaviors at will, by simply integrating one or more new layers. Figure 4 shows the conceptual design of the framework. The

framework consists of three levels, namely Physical layer, Behavioral layer, and Navigational layer. Every level contains a simple and/or complex highly independent task that may be executed by a pedestrian and has its own characteristics. This task will process inputs from the environment and produce as output an action for the pedestrian. Each level can be divided into modules which is associated in turn with more detailed and specific routines.

A. Physical Layer

The physical layer is the bottom level, it is comprises the perception and the action sub-modules.

- Perception module: is one which is responsible for collecting and acquiring sensory data by interpreting the virtual environment, it provides the virtual agent information about its surrounding space and makes it aware of the current situations for discuss the appropriate actions. In the simplest way, with this component, the pedestrian determines dynamically the current position, velocity and orientation of their neighbouring agents and the other visible elements depending on the agent's direction and field of view.
- Action module: This component including a walking sequence of animation which runs to achieve the locomotive tasks of pedestrians, then it transfers the behaviour determined by the behavioural level to the specific executions of the agent's effectors in the virtual space in order to move smoothly the pedestrian along the path that has been planned [27].

B. Behavioral Layer

Once the shortest path is computed by the navigational level, the next challenge is how the pedestrians follow, move and how evade each other naturally to achieve their goals. The proposed model deals with this problem by the behavioral level. The main role of this level is to address the realistic local movement of the individual pedestrians of the virtual crowd while preserving visually pleasing overall crowd behaviors. This is a challenging task, particularly in crowded scenarios with several hundred or thousands of entities where each one requires independently navigating in a decentralized way.

The behavioral layer uses [27] a microscopic approach of crowd simulation where the pedestrians are represented by a position in space, an orientation and a velocity. During each simulation phase, this level receives the information perceived by pedestrian from the physical level, and a set of way points representing the path calculated by navigational layer, and using it to decide the new behavior which is represented by determining the new location of pedestrian by modifying his orientation and/or his speed. This layer receives the information perceived by pedestrian from the physical level, and a set of way points representing the path calculated by navigational layer, and combines a forces model with behavioral rules to decide the new behavior which is represented by determining the new location of pedestrian by modifying his orientation and/or his speed. Finally this information is then sent to physical level in order to implement the walking animation.

C. Navigational Layer

The Navigational layer is responsible to represent the navigation process. Hence, the navigation process will allow to people to follow the known route or choose a new path based on changes in the environment and their psychological parameters. This process focuses to analyze the representation and to compute an optimal path between any two locations in the environment. In this context, navigation process could be either local or global [27].

- Local Path Planning means the path is done while the pedestrian is moving; this process is capable of producing a new route in response to environment changes, i.e. user inserts or removes one or set of obstacles in the environment, or a door appearing blocked which makes that path invalid or creates a bottleneck in some part of the desired path. Specifically, the Local Path Planning Process is used in the case where the entity and its sub-goal are in the same region.
- Global Path Planning requires that the environment be completely known. In this approach, an offline process (before the simulation is started) generates a complete path from the start point to the destination point, and then the route is pre-calculated and stored in the topological graph in order to achieve real-time interactive navigation. The Global Path Planning Process is executed if the pedestrian and his sub-goal are in two different zones.

Both these two processes send the shortest path in form a set of waypoints (attractor points) to the Behavior Mechanism to carry out the required motion to reach it.

VII. INTERACTION BETWEEN MODELS

Our approach proposed here for the design a hybrid crowd simulation model concentrates on the integration of two models that have different level of resolution in order to capture at the same time the micro and macro dynamics of human crowds. This type of simulation has the ability to divide the environment into multiple disjoint areas and to simultaneously execute these two models in different regions, by using a macroscopic modeling approach to simulate pedestrian flow in region of high density and a detailed microscopic model to simulate individual behaviors of pedestrian in the other regions. Investigating this integration makes clear two hard problems to be encountered:

First, assuring the consistency between the models is important for maintaining semantic continuity of results in terms of space (discrete/continuous), behaviors, and time, when concurrent interactions occur, if pedestrians pass from microscopic scale to macroscopic scale and vice-versa. Final issue consists to provide an efficient strategy for changing adaptively the simulation method in a specific region. It needs to identify the conditions in which a suitable modeling approach can be selected.

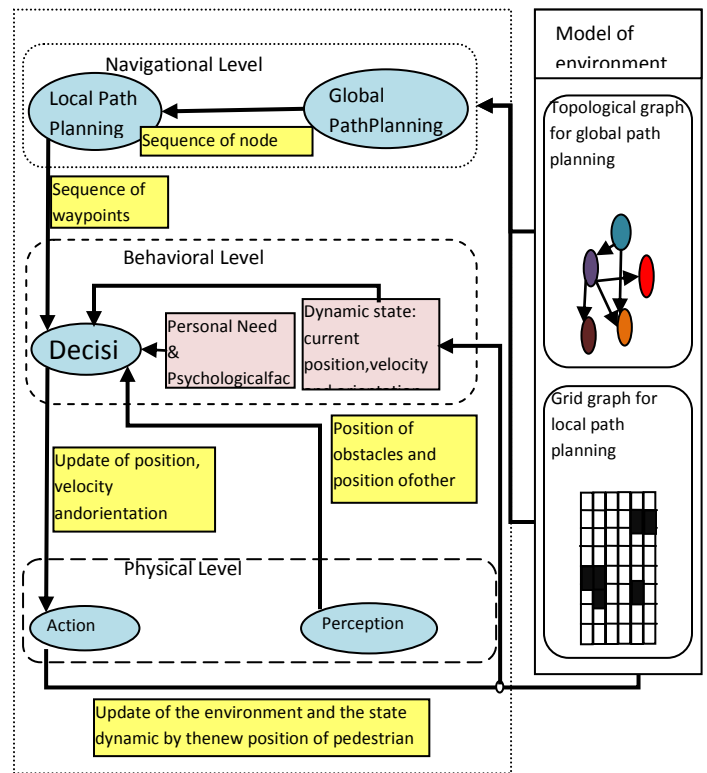


Fig. 4. Multi-layer architecture for modeling pedestrian movement, three levels are interesting physical layer, behavioral layer and navigational layer

A. Transition of pedestrians

As mentioned above, inconsistencies can arise in our hybridization solution when persons can be transformed from one model to another of different scale. To achieve this, an interface translating boundary condition is needed to define for moving those pedestrians from a continuous to a discrete modeling approach. We adapt this case by defining a boundary area adjacent to the macroscopic region; this bound is divided to cells, only the pedestrians in this area transform to the microscopic scale, i.e. if there are when pedestrians enter this area forcibly changing their representation levels, their positions, orientation and velocity will updated by the microscopic model.

There exist two basic communication operations between the two models: aggregation and disaggregation. The disaggregation refers to the process of generating the initial parameters for the microscopic model based on the result from the macroscopic model. Correspondingly, aggregation is the operation where the collects statistics from microscopic model and the parameters are generated in the format as required by the macroscopic model.

B. Trigger

Our hybrid approach of modeling consists to combine two models of different level of detail of simulation in a single framework; it attempts to provide a scalable and accurate method for the autonomous navigation process.

The whole continuous simulation environment is divided into multiple mutually regions, defined at initialization. Each region is governed by one of two different motion modeling approaches, either macroscopic model or microscopic model.

According to the density of crowd, the model used in a specific region is determined at initialization and modifiable at runtime. Then it is very important to present an efficient technique to define how the simulation changes dynamically from one model to another. Density is another fundamental component of pedestrian flow models. As the density of pedestrians' increases, pedestrians will have less space to overtake other slow pedestrians and eventually the average walking speed is slowed down. Usually when the pedestrian density is higher than 5~6persons/m², the average walking speed is so low that the crowd can hardly move any more.

- Switching Micro→Macro we calculate the density in each area. This operation occurs when the density in an area ruled by microscopic simulation is larger than a predefined threshold, then the system should trigger the execution of macroscopic model in this region.
- Switching Macro→Micro we calculate the density in each area, It consists to switch the simulation in a specific zone from a macroscopic to microscopic model; this is occurred when the density of this region is smaller than a threshold. Then the microscopic should execute.

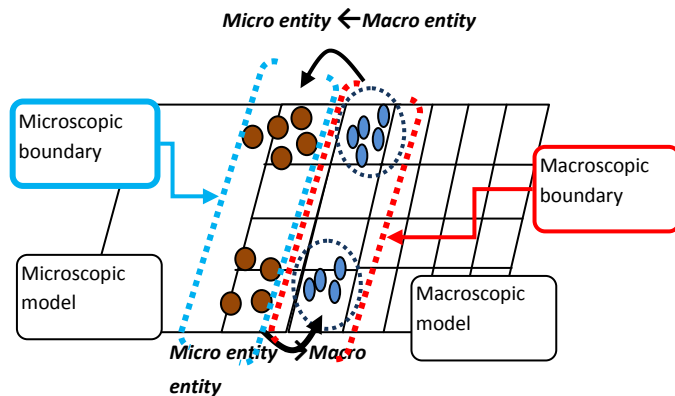


Fig. 5. Interaction between micro and macroscopic model

VIII. RESULTS

In this section we present and describe the experiments for the evaluation of our hybrid model for interactive visual simulation of large scale crowd of virtual pedestrian. Our solution model is specifically designed to support robust real time simulation of scenarios with thousands or even hundreds of thousands of pedestrians. It involves the combination of two significantly different types of modeling methodologies for taking the advantage of their complimentary features, in which a macroscopic model is applied where needed and a microscopic model where plausible.

In order to evaluate the effectiveness and the robustness of our multi-modeling approach presented in this paper, we have conducted to realize a number of simulations with different

initial distributions and conditions (mainly changing the density of pedestrian crowd in the environment) in a situation in which experiments focused at analyzing the impact of the density of crowd on the pedestrian behavior that could be handled was being investigated.

The objective of the experiments which we use is to show the proposed model performs well to produce results that closely simulate real human behaviors in these situations, and to study whether the proposed model can describe the qualitative dynamic properties of the pedestrian's movement under situations with three different level of density (low, medium and high density) in terms of number of pedestrians that could be handled with reasonable performance.

We tested our system on virtual complex environment, in order to produce realistic crowd behaviors in this type of space; the simulation environment itself should have the features (properties) of real life environment. We also believe the representation of the environment has an important influence on pedestrian navigation. In this section we demonstrate the application of the hybrid approach using the arbitrarily complex geometry. The structure has a free floor space area comprising of 7 irregular shaped rooms with two external exits. The pedestrians are initially distributed randomly over the area of the environment. The particular distribution of the density pedestrian was selected to ensure that during the simulation all two possible modeling approaches: Macroscopic model; and microscopic model; would be used. We performed a series of experiments in order to test the behaviors under study focusing on showing the results of the interaction of the two sub-models.

Fig.6 shows the simulation results of the pedestrian dynamic produced by the microscopic model which is selected to apply according to the pedestrian density calculated as the numbers of pedestrians existing in the restricted areas under consideration. This model is considered to be qualitatively more accurate than the macroscopic model. This experiment shows that when the number of virtual pedestrian is small (Fig.6 (a)), the microscopic modeling approach has been employed to simulate each pedestrian as an individually entity with its own its own personality, and its behavior which is determined by both the global and local movement. In this scenario, we demonstrated that a leader has a major influence on people especially in evacuation situation, in order to formulate the leader-follow behaviors. Fig.6 (b) shows this case, when, we can observed the red flow follow a leader which has a global view of the simulated environment, then he find the shortest path into the exit, but the blue flow has no leader, then he choose the shortest door which leads it to follow the longest path (Fig. 6 (c, d)).

During the second experiment (Fig.7), we noticed that pedestrian's density increases in the same subarea; the macroscopic is adopted to handle the pedestrian's behaviors within a crowd of high density. This model facilitates the construction of small groups of individuals that shows a slight cohesion and natural fragmentation into subgroups that might be simple and therefore much more compact.

Group phenomenon is an interesting area of research for pedestrian simulation, because it is very common in the every-

day life, people standing closer to its familiars and forming small groups. In panic situation, people relatively tend to gather together closer. In these situations, people are mostly linked by the (temporary) sharing of a common goal, and the overall group tends to maintain only weak compactness, with a following behavior between members [27]. In Fig. 6, the macroscopic used to cluster the pedestrians into a structured group by assuming a common goal, passing a direction and speed that applied to all of the members. As a member of a group, each pedestrian coordinates with others in the same group and show an aggregate motion as they move together.

The last simulation results in fig (8) show that in crowded situations, (pedestrian's density increases until it reaches a maximum value when situation becomes congested), one of the typical phenomena occurring in pedestrian flow is self-organization of lane phenomena. In the real life, pedestrians in a crowded area tend to self-organize into lanes in order to reach their destination faster and easier.

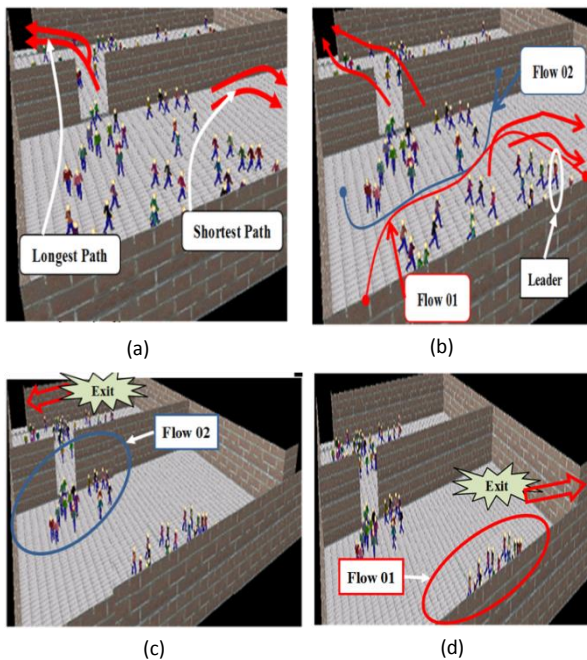


Fig. 6. The leader fined the shortest path, and the flow 1 follows it



Fig. 7. Creation of small pedestrian flow in normal situation, these pedestrian flows are formed when the pedestrian density is high by using the macroscopic model

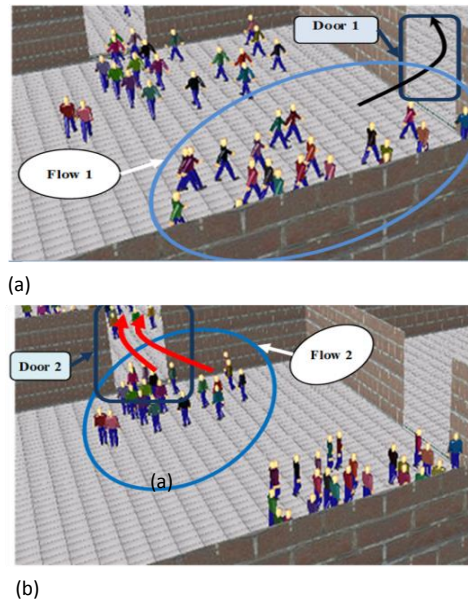


Fig. 8. Fig. 1. The small groups follow themselves for forming a flow

IX. CONCLUSION

Multi-approach modeling is proposed in this work as an adaptive simulation strategy for exhibiting the pedestrian crowd movements and its emergent behaviors in high density situation. Our method makes it possible to exploit advantages from both macroscopic and microscopic models. The two types of models work simultaneously in a single simulation system, and are executed over different mutually exclusive partitions.

It is important to notice that our resulting hybrid technique can automatically and dynamically select the suitable strategy; the dynamic switching between both models is ruled by the runtime simulation metric which is the crowd density in the partitions of the virtual environment. Our model also ensures that no visible disturbance is generated when adaptively change the simulation method in a region. The partitioning of the environment allows us to define transition zones where the two types of movement modeling approach must be interacted and crowd under one regime must be moved to the other.

We will continue to work on the proposed behavior model, which aims to be useful in different kinds of crowd simulation applications, our future work provides a development of a wide variety of social behaviors. These behaviors are managed for more accurate simulation results under various complex conditions by incorporating the more complex group structures and the interactions between the different types of pedestrians. Further work, the coupling of mesoscopic models with our model will develop to apply in the region with middle density.

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Data Mining and Intrusion Detection Systems

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Abstract—The rapid evolution of technology and the increased connectivity among its components, imposes new cyber-security challenges. To tackle this growing trend in computer attacks and respond threats, industry professionals and academics are joining forces in order to build Intrusion Detection Systems (IDS) that combine high accuracy with low complexity and time efficiency. The present article gives an overview of existing Intrusion Detection Systems (IDS) along with their main principles. Also this article argues whether data mining and its core feature which is knowledge discovery can help in creating Data mining based IDSs that can achieve higher accuracy to novel types of intrusion and demonstrate more robust behaviour compared to traditional IDSs.

Keywords—(Intrusion Detection; NSL-KDD; Machine Learning; Datasets; Classifiers; Feature Selection; Waikato Environment for Knowledge Analysis; Anomaly detection; Misuse detection; Data mining)

I. INTRODUCTION

As the years have passed by computer attacks have become less glamorous. Just having a computer or local network connected to the internet, heightens the risk of having perpetrators try to break in, installation of malicious tools and programs, and possibly systems that target machines on the internet in an attempt to remotely control them. The (GOA) team categorised the attacks encountered in 2014 discovering that 25% of the attacks were non-cyber threats followed by scan/probes/attempted access 19% and policy violation 17% [1]. This data is further acknowledged by the annual FBI/CSI survey which discovered that though virus based attacks occurred more frequently, attacks based on unauthorised access and denial of service attacks both internally as well as externally, increased drastically.

Recent exploits also suggest that the more sensitive the information that is held is, the higher the probability of being a target. Several Retailers, banks, public utilities and organizations have lost millions of customer data to attackers, losing money and damaging their brand image [2]. In some cases attackers steal sensitive information and attempt to blackmail companies by threatening to sell it to third parties [5]. In the second quarter of 2014, Code Spaces (source code company) was forced out of business after attackers deleted its client databases and backups. JP Morgan, Americas' largest bank, suffered a cyber-attack in 2014 that impacted 76 million members [3]. In 2014, Benesse, A Japanese Education Company for children suffered a major breach whereby a disgruntled former employee of a third-party partner disclosed up to 28 million customer accounts to advertisers [4]. Most notably the "Sony Pictures hack" best displayed how significant a companies' losses are in the aftermath of a

security breach. The network servers were temporarily shut down due to the hack [4]. Cybersecurity experts estimate that Sony lost up to \$100 million [5] [6]. Other companies under the Sony blanket fell victim to attacks [7]. To tackle this growing trend in computer attacks and respond threat, industry professionals and academics are joining forces in a bid to develop systems that monitor network traffic activity raising alerts for unpermitted activities. These systems are best described as Intrusion Detection Systems.

II. INTRUSION DETECTION SYSTEMS

A. Definition of Intrusion Detection

Heady et al. [8] describes an intrusion as a set of actions that make attempts to challenge the integrity, discretion or accessibility of a resource. Generally the practice of intrusion detection involves the tracking of important events which take place in a computer system and analysing them in order to detect the potential presence of intrusions [9]. Alessandri gives a more comprehensive definition of intrusion detection, describing it as a collection of practices and mechanisms used to detect errors that may lead to security failure with the use of anomaly and misuse detection and by diagnosing intrusions and attacks [8].

Correspondingly it may be added that an intrusion detection system is the practical implementation of intrusion detection principles and mechanisms over a network [8]. This is combination of software and/or hardware components that run on a host machine monitoring the activities of users and programs searching for possible insider threats on the host device and also inspecting network traffic of networks that are connected to the host, looking for outsider threats [8]. The objective of an IDS is to alert administrators of suspicious activities and in some cases even attempt to circumvent the attacks. The practices employed in IDSs' do differ from other security techniques such as firewalls, access control or encryption which aim to secure the computer system. With this being identified however it is strongly recommended that these security practices are used in conjunction with one another as this reinforces defence of a system and ensures that a much larger scope of a system is protected [9].

B. History of Intrusion Detection

Originally, Intrusion Detection (ID) was conducted manually by system administration. They were tasked with thoroughly monitoring each activity on a console identifying any anomalies. This early form of ID proved ineffective due to the errors it produced. Automated log file readers were then developed allowing quick searching for irregularities and unauthorised personnel [8]. It is worth noting that early versions of ID were owned by few organisations, computing

was not a widespread practice and the technological computing age had not been born [8]. The introduction of audit logs helped manifesting ID into a forensic technique; whereby administration collated information and only identified issues after incidents had already occurred and not during the process of an attack. Before the 90s' Intrusion detection was a form of post analysis, analysis of intrusions and changes in system structure were only identified long after the actual event. The processes were tedious, slow time consuming and presented potential of human error due to heavy involvement [11]. During the '80 to 90s' research was carried out in a bid to strengthen existing ID software. Some suggest that the breakthrough came in the 90s' as a result of the Intrusion Detection System proposed by Denning [12]. Researchers developed an IDS that reviewed audit data as it was produced. This advancement spawned the first version of real time IDSs' allowing for attack pre-emption through methods of real time response [10]. As the world entered the technological age, the market demand for IT security increased and IDS were further developed and made available to large organisations. New features were developed such as various new alert methods, updates to attack pattern definitions, dedicated user friendly interfaces and prevention techniques that automatically stopped attacks when identified [11].

With the focuses now shifting toward enhancing security measures, newer attack techniques continued to spawn from every corner of the web; most notably the Millennium bug and Morris worm. Due to this it became apparent to developers that in an ever changing environment one must always seek to improve and stay ahead as threats become more diverse in their methods to find new ways to penetrate systems.

C. Understanding taxonomy of IDS

A general definition of Taxonomy is the practice or principle of classification [9]. Taxonomy may serve several purposes in design. Firstly, it can describe the current global situation, assisting in refining complex situations and presenting it in a clearer measured approach (Description). Additionally using taxonomy to classify a number of objects, enables identification of missing objects early in design, which allows users to exploit the predictive qualities of a good taxonomy (Prediction). Lastly, a good taxonomy presents users with ideas, further explaining observed current occurrences (Explanation).

There have been many taxonomies presented for IDSs' and ID technologies, dating back as far as 1999. The first real recognised IDS taxonomy seems to be the one proposed by Debar et al. [18]. Since then many more taxonomies have been published, most notably is the one proposed by Axelsson [9] followed by another proposed by Halme and Bauer [19]. The identified taxonomies can be used in order to illustrate general relationships in IDSs'. Figure 1 which is illustrated below displays the revised version of the taxonomy previously proposed by Debar et al. in 1999, this version features additional criteria for classifications [39].

Debar explains the significance of understanding the system before creating an IDS and expanding on various mechanisms used in IDS to enable a structured approach in design [39].

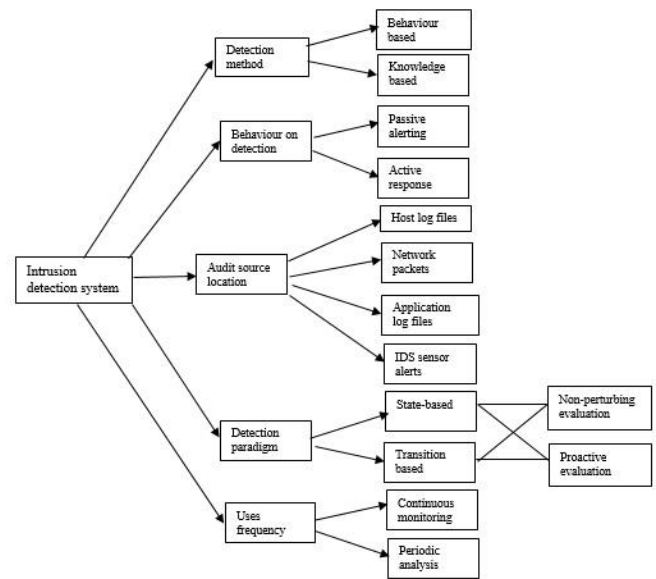


Fig. 1. Updated IDS taxonomy

It is believed that following this practice results in development of efficient Intrusion Detection Systems. There is no clear indication as to whether this is a justified approach to creating IDSs', however the sheer sum of researchers who incorporate this technique into their design may serve as evidence, suggesting the importance of taxonomy. The Halme and Bauer [19] taxonomy named "A taxonomy of Anti-Intrusion Techniques", focuses on unexplored methods in combating intrusion activities, it focusses on this rather than dealing with IDSs'.

The taxonomy reveals six anti-intrusion approaches prevention, pre-emption, deflection, deterrence, detection, and/or autonomously countered. Axelssons' taxonomy propose an enhanced approach, as it deals solely with the IDSs [14]. This commences with classification of the section principles, followed by the operational tasks of the IDS. The taxonomy aims to analyse current IDSs, which consequently allows progress in researching the chosen field enabling categorisation to help aid enhance knowledge of the field. In Figure 1 the taxonomy follows a series of steps, firstly the classification of the detection principle and then certain operational aspects of the intrusion detection system. In Figure 1, initiation starts through first identifying the different types of intrusive behaviours generated by an intruder. Further progressing to questioning suitable practices on how to observe these intrusions (intrusion sources), the repercussion of doing so and finally the outcome and decision.

D. Intrusion Detection System Categories

By further analysing Axelssons taxonomy, two methods of IDS may be discovered when viewed from another perspective as illustrated in Table 1 [17]. The first is building a taxonomy using principals of IDSs', where categorisation is based upon the following detection methods; Anomaly detection, Signature detection and Hybrid/compound detection. Following on from this the author establishes IDS classifications, based upon system characteristics, time of detection and response to detecting intrusions as presented in Table 1.

TABLE I. GENERAL IDS TAXONOMY

Anomaly	Self learning	Non-time series
		Time series
	Programmed	Descriptive stats
Default deny		
Signature	Programmed	State modelling
		Expert system
		String matching
		Simple rule-based
Signature inspired	Self learning	automatic feature selection

III. IDS PRINCIPLES

An intrusion detection system functions by determining whether a set of actions can be deemed as intrusion on a basis of one or more models of intrusion. A model describes a list of states or actions as good or bad (potential intrusion) [20]. These ID methods can be implemented into two different system categorisations. Anomaly detection system which identifies network traffic behaviour and misuse detection system which bases its detection on signatures or pattern matching, also described as knowledge based.

A. Anomaly Detection

An anomaly detection based system uses the normal profile of a system or user to determine its decision making process[10]. Development begins at the point at which the detector forms a judgement on behaviour that constitutes to normal for the observed object in question (application, user, resource usage etc.) and then a percentage of this activity may be flagged as suspicious and a preserved action is then taken [14]. This type of system is suited for the detection of previously unknown attacks as it detects most intrusions without having acquired prior knowledge of the intrusion [20,40,41]. However, issues still remain as it fails to expand on details surrounding the particular intrusion, (fault diagnosis), in addition this system is noted to return high false positive rates [10]. These detection methods can be used to define signatures for misuse detectors and when merged with a misuse system can form a hybrid system [12].

1) Self Learning systems

When using self learning IDS, no underlying information is made available to them. The IDS typically learns by observing the traffic and creating a model for each system's fundamental process [14]. Self-learning IDS may use "non-time series" or "times series" approaches to formulate a model for the normal behaviour of the system. Time series is the more complex technique as it requires time to be taken into account; examples of these techniques the Hidden Markov Model (HMM) and Artificial Neural Network (ANN).

2) Programmed

In this classification, an expert programmes the system to detect certain irregular events. The system does not learn on its own and requires an expert to form the normal behaviour profiles of the system; then later deciding actions to be considered abnormal that trigger an alert. Thus, this is the user of the system who defines the normal operation. Axelsson states that programmed IDSs use two techniques firstly, descriptive statistics, whereby the system builds a profile of normal statistical behaviour by the parameters of the system

and then gathering descriptive statistics on selection of parameters. Secondly is default deny, this is where the class plainly states each status whereby the system functions in or around a realm deemed safe and secure; deviations from this state will flag as intrusions [14].

Halme and Bauer [19] further categorise anomaly detection systems through system specification and profiling. Determining the system components and the behaviours to capture and monitor, allows for different classes to appear. Classes such as Threshold Monitoring, user profiling, group profiling, static work profiling and adaptive rule based profiling are discussed.

B. Misuse Detection

Authors in [11] and [12] demonstrate Misuse Detection Systems commonly referred to as signature based detection systems, because they work by using patterns of recognised attacks or known critical points in a system to find and match known intrusions. The decisions made, are formed on the basis of knowledge of a model, dealing with the intrusion process and what is to be tracked in the observed system. Misuse detection offers greater accuracy and can efficiently detect variations of recognised attacks. Furthermore, such IDS also offer more meaningful intrusion diagnostics when an alarm is triggered by detailing diagnostic information about the cause of an alarm [8]. However these detection systems also have some pitfalls as they lacks ability in detecting new attacks and signatures that are unavailable [10]. Contrary to this it must be noted that many commercial systems employ misuse detection systems such as Cisco which employs knowledge-based systems.

1) Programmed

In this approach the system is programmed from the offset with a clear decision rule set. This rule set, offers simplicity as it contains coding of expected responses in the event of an intrusion. Another variation of programmed misuse IDS is the State Modelling method, where the intrusion is coded as a number of different states; each of them must exist within the observed space to be determined as an intrusion. These methods are in fact time series models. Two subclasses exist within the method, State transition which states the chain that is negotiated from beginning to end and petri-net. Both subclasses establish a petri-net, the structure of this system is similar to that of a tree and several states can be satisfied in any order no matter where they may occur in the model.

The expert system class is an intelligent system employed to evaluate the security state of the system, when assigned rules that describe intrusive actions. Forward chaining, production tools are often used as they are appropriate when dealing with systems, where new data are continually entered into the system. String matching method is also used for matching case sensitive characters in text, which are exchanged among systems. Simple rule-based systems are simplified versions of an expert system. These tend to execute quicker as they are not as advanced [14]. To conclude, Axelsson suggests in his latest revision of his taxonomy that current research must be redeployed in studying the effectiveness of intrusion detection and how to handle attacks against intrusion detection systems themselves.

Countless research has been conducted in an attempt to answer this question; with many quality contributions noted in this area, however there is still plenty of space to improve areas in IDS development and fulfilling the proposal made by Axelsson. Based on expert opinions there is a general consensus regarding the current state of network IDS. Many organizations are opting into purchasing signature based intrusion detection systems, due to the fact that they require less supervision, offer more automation and consume less time in setting features; therefore there is a belief that chances of human error are reduced. Furthermore, its widely stated that the majority of these organisations will employ IDSs' that are not suited to their system needs as they simply pick the biggest brands, which may offer simplicity but on the same time they are left without an understanding of how to use these systems. Many experts state that issues still remain in identifying new forms of intrusion and in order to stay ahead, the cyber security industry must continue to develop IDS and organisations train key staff on how to use these devices rather than relying solely on automation.

IV. DATA MINING

This section of this article will aim to combat the concerns raised by Axelsson as well as other experts, a review of existing literature is undertaken with surveys on current datasets in IDS, effective use of classifier algorithms, identifying relevant fields for feature selection and suitable ranking systems to test performance characteristics when a test is undertaken.

Computers have been identified as one of the sole orchestrators in building a platform to move technology enhancements. This has also had an impact on network traffic monitoring solutions, with huge volumes of data generated, some of which being heterogeneous and from different origins and travelling across devices at high speeds. Subsequently all of these factors makes it difficult to produce accurate analysis of data in a timely manor [12]. Data mining is identified as a solution to handling the analysis of data due to its adaptability and validity and it is now used extensively used for network security purposes [13].

Authors in [32] describe how intrusion detection systems categorise network traffic as either an anomaly or normal. Data mining is employed into an intrusion detection system as a method of extracting the huge volumes of data that exist in network traffic for further analysis [14]. As an application of machine learning, data mining holds a very significant position in intrusion detection, presenting methods of predicting future patterns based on past experiences [15].

However, in the same way significant researchers suggest that major challenges lie within intrusion detection and evidence demonstrates the difficulties in current data mining tools, such as high False Alarm Ratio (FAR), and low Detection Ratio (DR). Further development have been suggested in current data mining tools as questions have been raised about the quality of tools implemented [16]. Witten states that to be able to answer current questions surrounding data mining one must grasp the concept of learning while working with data mining [19]. Witten defines Data mining as a topic that involves learning in a practical and non-theoretical

sense. In a way the author signifies that learning within data mining is essentially steered by the ability to think whilst also having purpose. The researcher concludes that learning without purpose is simply training and not a practice of data mining.

Knowledge discovery in databases (KDD) is a term most frequently used interchangeably with data mining and it is defined as the application of a scientific method to data mining [18]. The typical process over KDD model includes methodologies for extracting, preparing data and making decisions about actions once mining has taken place. Maimon identifies that the process commonly has 4 to 12 steps [18].

Step 1, The idea is to begin with the goal identification task, understanding a particular domain, anguish knowledge discovery is required. Step 2, The next stage is to identify a target dataset, an initial set of data for analysis. Step 3, the pre-processing of data, the use of available resources to move noisy data and decide how to deal with redundant data values and so on. Step 4, the transformation of data, the addition or elimination of attributes and instances from the target datasets, decisions on matters of normalisation, combination and smoothing of data. Step 5, here the most appropriate features for representing data is built using data algorithms. Step 6, analysis of the outputs from the previous step, determining the usefulness of the discovery and deciding whether to change steps prior to 5, possibly using different attributes to achieve different results. Step 7, if the knowledge is deemed suited it is applied and incorporated directly as a solution to the problem [18].

The following scenario described by Witten gives an understanding of the possible data mining application into current practices; the combination of knowledge discovery stages identified by Maimon enable illustration of the learning definition that Witten explains about data mining [18]. The first three stages of knowledge discovery can be noted within the scenario. The problem domain is firstly identified, subsequently data for analysis and characteristics is specified then desired features are defined. Witten illustrates this concept by using the following example; human in vitro fertilisation involves collecting many eggs from a womans ovaries, once fertilised several embryos are produced. Some are selected and transferred to the womans uterus. Challenges here lay in the subject of identifying the 'best embryos' to use, and the most likely to survive. Selection is based upon the 60 features of an embryo, characterisations such as follicle, sperm sample, morphology etc. This large number of features creates an issue amongst embryologists in assessing them all concurrently while also correlating historical data to determine if an embryo was likely to result in a child or not. Data mining and machine learning algorithms are used to solve the identified issue. The practice of data mining has been applied to many fields, such as sales, healthcare, medical, finance, multimedia and most importantly intrusion detection [17].

It can be concluded that Data mining, offers more than simply finding data and applying algorithms over it. Both seemingly accredited the lack of efficiency identified within data mining to lack of understanding among researchers with Witten suggesting that several publications have erroneously used data mining procedures [16]. Running many algorithms

over a particular dataset and writing results, claiming one method or machine learning algorithm over another with little understanding of the nature of the dataset; poor understanding of the learning algorithm and no deliberation of the statistical importance in results [16,17] are some representative paradigms.

V. APPLYING DATA MINING ALGORITHMS TO INTRUSION DETECTION

The growth of data mining methods has consequently brought forth a wide range of algorithms drawn from areas as pattern recognition, machine learning and database analysis. There are many types of algorithms that may be used to mine audit data. Lazar identifies data algorithms as a set of heuristics and designs between data mining models [19]. In effect, the author suggests that for a model to be formulated, the algorithms must start by analysing the data type provided, in order to find particular trends and patterns. These results of analysis are later used by the algorithm for defining optimal parameters to create the selected mining model. The parameters are applied across the dataset, together with selected patterns and detailed statistics [13].

Scholars, Manish and Hadi conducted an investigation into network traffic analysis and prediction techniques following this they established a list of currently used data mining techniques. The authors use Table 2 to present the most commonly used data mining techniques.

TABLE II. DATA MINING TECHNIQUES

Data mining techniques
Clustering
Classification
Hybrid
Association
Other methods

Numerous studies indicate that classification techniques and clustering are by far the most widely used data mining techniques. The hybrid technique is considered shortly after together with the Association technique [15,13,20].

Manish and Hadi [13] stating that clustering is the process of splitting data into clusters based upon the features of the data. This clustering partitions data into groups of similar objects. Each member within the cluster is similar to one another [13]. Witten adds by describing clustering as an unsupervised learning method primarily used when training the normality model for anomaly detection and situations where little knowledge of the attack class is required while training. Wahono further expands on the functionality of techniques by expanding on methods of grouping together into clusters using distance functions. In addition several clustering, classification algorithms are identified, however the most widely used seemed to be the k-means classification [20].

Classification and prediction, as seen in Table 2 are described as the most popular mining techniques; allowing for extraction of models, describing important data classes and aiding in predicting future trends [13]. Wahono expands on this stating that it classifies data into metrics-based classification such as normal or abnormal in intrusion detection systems.

More importantly he states that classification maps data items to one of many predefined categories. The classifier's output can be used to predict a model that may forecast future trends, when sufficient normal and abnormal behaviour is gathered in audit data. The classification algorithm may be able to predict new unseen data classifying it by using pre-existing information. In addition, the author identifies some of the most widely used approaches in data classification; Bayesian classification, decision tree induction, neural network and statistical learning [20].

Manish and Hadi, briefly summarise the contrast between clustering algorithms and classification algorithms; classification algorithms require knowledge in both normal and known attack data in order to separate classes during detection [13]. Manish and Hadi determines that the Association technique discovers anomalies by using association rule algorithms, suggesting that the best applications for using the technique is; finding as many related defects to the detected defect within data, evaluating results during inspections and analysing reasons for anomalies recurring within data. Wahano argues that this technique is better suited to handling forensic analysis and not real time attacks, suggesting that the process is time consuming and would not benefit network analysis when scanning the system. The hybrid technique is a combination of two or more approaches for analysis of network traffic. The hybrid model achieves good results in the analysis of network traffic. We present various hybrid model techniques that are investigated by researchers for network traffic analysis.

Wahono briefly describes hybrid models stating that they are the combination of two or more approaches that may be utilised when analysing network traffic. Further highlighting that these approaches are generally new developments in data mining, offering new solutions in network analysis, however they are under-utilised and many of these techniques are difficult to be implemented. Notably, Haratian states that rapid development in data mining has introduced a wide range of attribute-value conditions [22]. These values often occur together in given sets of data and may help develop an understanding when determining relationships between the fields in database records. The author uses a market transaction basket example to illustrate how the method groups relevant data. Through analysis of the confidence and rule support figures within data, a system can judge the possibility of an action occurring, for example the age of a customer and the income they produce per year, the research suggest how likely it is for the individual to purchase a dvd player with the use of statistical analysis to understand the associations that may occur.

VI. MODEL OF IDS

This section will examine the components required to construct a model of IDS. First it uncovers datasets, and the finally classifiers. It is of utmost importance to understand the contents of a dataset, and the purpose of the attacks featured in order to help design and build appropriate tools [16].

VII. INTRUSION DETECTION DATA SETS

Amudha et al. [21] presents a diagram which illustrates the knowledge discovery process (Figure 2). This process notably

aids in creation of a model for intrusion detection system. At the first stage the dataset is chosen and pre-processing techniques are used to clean data.

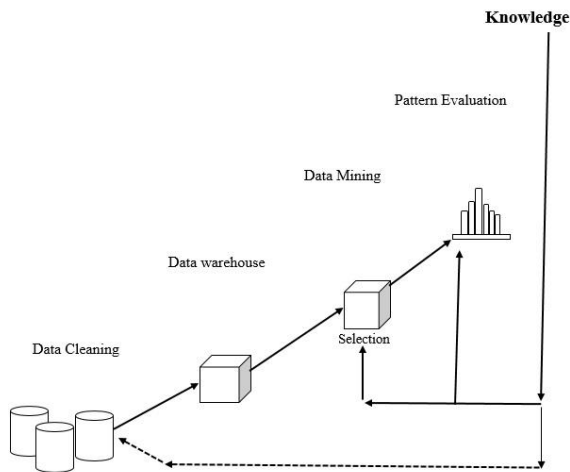


Fig. 2. Steps in knowledge discovery

Extensive research shows that the following datasets were commonly used for investigations by recognising that the KDD dataset, is widely known among scholars in the network analysis community, offering a large scope of records in the previous DARPA dataset, with up to 4,900,000 training instances, 41 features, 24 training and testing attacks with a further 14 types. The dataset offers more information compared to the DARPA dataset. However, Bajaj and Arora state that the KDD dataset is outdated, suggesting that the NSL-KDD dataset is the most suited for current network analysis. They state that KDD 99 dataset suffers with redundant data which often lead to biased detection of attacks, highlighting more frequency in DOS and probe attack. This lead to failures in classifying features appropriately and most records cannot be classified and are misrepresented in most of the cases. The author states that if the KDD, datasets are used, investigations will likely present results that do not represent real network situations.

Neethu also states that the KDD dataset is outdated and due to technological advances can no longer provide accuracy for evaluation. However the lack of alternatives is the reason the dataset is still in use. Bajaj and Arora identify The NSL-KDD as the most appropriate network analysis dataset as it manages to eliminate identical records in testing data and redundant instances in training data. This means that, classifiers are more likely to be able to categorise data if records are matched correctly. The author states that this factor can affect the accuracy of a classifier producing better results than the KDD Dataset. In contrast, Neethu believes that the NSL-KDD dataset contains some redundant instances and the dataset cannot be used for the correct training of models.

Neethu suggests that the most appropriate dataset is the ADFA-LD dataset, introduced in 2013. Normal training data holds up to 833 traces and also 10 attacks in attack data, this dataset is believed to have a closer resemblance between attack and normal data, than KDD security datasets. However the scholar does state that many researchers have struggled to utilise the dataset as features are not best described and it is

difficult for many developers to understand its functionality. This adds confusion when creating effective models for intrusion detection. The author, however believes that further experimentation in the dataset is required to demonstrate its effectiveness in network analysis [23] [24].

Amongst the majority of researchers, there seemed to be no preference in use of datasets within intrusion detection. There is a vast amount of datasets available on the internet and as Witten states the choice of a specific dataset should be accompanied by an understanding of the problem that one wishes to address, an understanding of classifiers that may be implemented over it and a way to effectively carryout an investigation [16].

VIII. EXTRACTING FEATURES

Feature reduction and selection are commonly used in current intrusion detection publication. The methods are often used interchangeably to indicate specific points. Feature reduction/extraction is the process of finding new subspaces with less dimensions than the original feature space [34]. Further expansion to explain feature selection is that the features established by feature selection must always be a subset of an original set of features, in contrast feature reduction reduces dimensions combining the linear combination of the original set and establishing new synthetic features, the least important features are discarded [34]. Independent component analysis

A. Feature Reduction

Feature reduction is finding a new subspace which has less dimensions than the original feature space. The following commonly used methods for feature reduction are presented, Linear Discriminant Analysis (LDA), Principal Component Analysis (PCA), Uncorrelated Linear Discriminant Analysis (ULDA), Independent component analysis (ICA) [5]. With regards to the pre-processing technique of Feature selection, it seems that many invocative hybrid techniques are being tested in response to the current outcry to improve issues of efficiency and accuracy discovered in current Datasets. Wahba et al. [25] developed a multiclass classification to aid in producing modern intrusion detection models with improved efficiency and accuracy. The aim was to merge different classifiers together to produce better results. The authors highlight the significant benefits of using multiclass IDS backed with recent research suggestions and investigations. The author further identifies that the overall performance of the model degrades during implementation. This is caused by attempts to fully merge classification patterns where features do not match, this results with redundancy in data [25]. The author proposes a technique to reduce irrelevant features and improve the performance of the model.

Tesfahun and Bhaskari [26] reveal that due to the erratic characteristics of intrusion detection, there is huge disproportion between the classes in the NSL-KDD dataset making it difficult, to apply machine learning effectively in the area of intrusion detection. The synthetic minority over sampling technique is applied to the training data set in an attempt to deal with class imbalance. Feature selection method based on information gain is used to create the reduced feature

subset. The classifier used as random forests algorithm with SMOTE and IG. The author states random forests classifiers were chosen over decision tree classifiers due to the fact that random forests algorithms can run on large datasets, have the ability to handle nominal data and do not over fit, classification is conducted through votes of string. The NSL-KDD datasets is chosen as benchmark due to research acknowledging that, KDD 99 calls a huge number of redundant records. Therefore preventing the identification of minority classes such as U2R and R2L. NSL-KDD dataset was used for test purposes following investigations the number of features were reduced to 22, the detection rate for U2R increased and time to build reduced.

Dhafian et al. [27] Investigates current literature surrounding classification techniques and methods of intrusion detection. Reviewing current IDS approaches using the following datasets; DARPA, KDD 99, NSL, KDD, Kyoto 2006 + and CAIDA. Findings suggest that NSL-KDD performed best overall once trained against specified classifiers. The authors conclude by suggesting consideration must be taken during developing of classification techniques identifying optimal dataset that is a rich the recent attacks and which features are selected without confusion, unnecessary overhead and time-consuming selection.

Desale and Ade [28] propose a Genetic Algorithm based Feature Selection Approach for Effective Intrusion Detection System. The genetic algorithm is used to search method when selecting features from the full NSL-KDD dataset. The mathematical intersection principle is used selecting features that appear in every experiment. The investigation is carried out on both test and training data set, proposed approach is measured against the popular approaches, namely the Correlation Feature Selection (CFS), Information gain (IG), Correlation Attribute Eval (CAE) and the effect on the performance of the Naive Bayes and J48T algorithm classifiers are measured. The resulting information determined that the proposed model selected the minimum features from the dataset, which improved the classifier, accuracy of the Naive Bayes classifier while also reducing time, complexity.

Chabathula et al. [29] propose Principal Component Analysis approach Using Machine for feature reduction and effective selection. The authors adopts a different approach to network analysis in order to reduce data features. Header fields of incoming packets are analysed in vectors, these serve as import for the PCA algorithm. The System is designed in two phases. The training and testing phase are conducted over the full NSL-KDD dataset. PCA produces features which are deemed weak. The test records are compared with the base profile during training phase and then the confusion matrix is used over the classification algorithms to determine performance. Results in the accuracy of detection time of each algorithm is measured. The following classifiers are tested upon within the investigation SVM, KNN, J48 tree, random Forest tree, classification, Adaboost, nearest neighbour, naive Bayes classifier and voting features classification. Two experiments take place using each classifier, one with PCA applied to it and without PCA. If an attack is present an alarm is sounded. Results display performance of classifiers in terms of detection time, testing accuracy and speed. They indicate

that tree-based classifiers such as J48 and Random Forest notably outperformed other classifiers in accuracy and speed. With linear growth up until the 18th dimension when it steadily increased. Voting features classification had a lower detection rate with PCA than without, the naive Bayes problematic classifier displayed better results with PCA than without it. Detection rates noted in SVM, KNN, J48 tree, random forest tree, Adaboost, nearest neighbour, were almost similar in both with PCA and without PCA.

Chauhan and Bahl [30] propose performance Analysis of Dimension Reduction Techniques with Classifier Combination for Intrusion Detection System. A Review of current dimension reduction techniques, search methods, attribute evaluators and classifiers was conducted. Different combinations feature selection and feature classification algorithms were applied to the datasets to detect intrusion. Results show that there was an increase in classification accuracy from 52% to 96% of PCA analysis. Scholars suggest that classification with a good accuracy results in a reduction in completion time and effective outputs.

B. Feature Selection

Feature selection is described as a method whereby specific features are selected from a set of features, which have a high discrimination capability between class labels. It is required to reduce the irrelevant features by eliminating them by using some methods. The method may be wrapper based or filter based or combination of them [29]. Feature selection is used to select sets with minimum length while ensuring maximum classification accuracy attributes of most importance and increase performance speed by reducing the irrelevant features through elimination [29].

Ganapathy et al. [31] review current feature selection and classification techniques for IDS, surveying intelligent techniques for developing IDSs then developing a new IDS using two proposed algorithms. Once an understanding was developed, a test scenario was formulated whereby a subset of KDD consisting of 10% of its records was used to test the algorithms. The researchers found that Modified Mutual Feature Based are more flexible during feature selection than Gradual feature removal as they use mutual information rather than relying on predetermined clustering when removing features. CRF based feature selection methods can handle uncertainty effectively and the wrapper based methods uses a decision tree to remove subsets of features. Analysis of these methods highlights that mutual information and information gain ratio provide the best methods of feature selection as they can be used to perform tuple reduction and attribute selection. After reviewing linear programming methods for detecting U2R attacks. They determine that a layer approach adopted through a Least Squares Support vector Machine will offers solutions of a linear equation to conquer trivial SVM methods, simplifying, detection of normal or attack data and improving accuracy and detection time. Furthermore they add that a neuro-tree classifier may also be used as it offers effective classification when optimal features are provided to it. A classification technique named IREMSVM is then proposed from current intelligent agent -based multiclass SVM, the new feature uses information gain ratio and attribute selection to effectively compute the feature set in a timely manner. Two

new algorithms, the IREMSVM, IAEMSVM and SVM were then tested against a full KDD dataset using all feature and one with selected features. Accuracy was analysed through comparing Probe, DOS and other attacks as the proposed algorithm uses constrain checking in classification. The conclusion was that Accuracy was higher in IREMSVM than in SVM or IAEMSVM.

Zargari and Voorhis [32] examine significant features in anomaly detection systems with an aim to apply them to data mining techniques. Identifying some current challenges of obtaining a comprehensive feature set and establishing a system that eradicates redundant and recurring data from the KDD 99 dataset while also keeping the feature set to a minimal size. Rough set theory dependency was used to identify the most discriminating features of each class. Feature 21 and 22 in the KDD dataset were found not to have any significance in intrusion detection (FTP session and hot login). A further five features were identified to have a small significance in intrusion detection. These were su attempted, number of file creation operations, is guest login, dst host error rate. To produce a distinctive report finding the features and characteristics of the intrusion detection that offers more attacks and distribution of attacks as compared to KDD dataset. Corrected KDD-dataset was used, in order to discover the features and characteristics of the intrusions plus, whether anomaly detection can be improved by using this dataset from a statistical point of view. It is important to mention that different to other studies, the Corrected KDD-dataset was analysed here instead of the KDD-dataset. The Corrected KDD-dataset contains more attacks and the distribution of attacks is different comparing to the distribution of attacks in the KDD-dataset. A subset of features was later proposed to help decrease dimensions of KDD and compare to subset features through data mining techniques. The proposed features were later tested on NSL –KDD and demonstrated higher detection rates for proposed features. The work may require live analysis before we can be sure that it would function correctly.

Aparicio-Navarro et al. [33] finds three scenarios in which correctly labelled datasets are required. Stating that when using unsupervised IDS there is a need for labelled datasets to be trained. When the nature of an analysed data set must be recognised to evaluate efficiency of IDS when detecting an intrusion. Finally using feature selection that only works if processed datasets are labelled. Finding the flaws in current practices of labelling datasets states that collection of labelled datasets from real-world networks is impossible as many datasets are labelled through off-line forensic analysis, which is impractical as it does not allow real-time implementation. The author, develops an approach that automatically generates labelled traffic datasets with unsupervised anomaly based IDS. The resulting labelled dataset are subsets of the original unlabelled dataset. The remaining dataset may contain valuable information and so is kept so an administrator may add or remove data as they see fit. The newly labels dataset are processed using genetic algorithm (GA) approach, that performs the feature selection. This GA is implemented to automatically provide the metrics to generate appropriate intrusion detection results while reducing the risk of

misclassification. The Ffitness is identified to have an important role in implementation of the technique allowing fine tuning of outcomes through maximising DR minimising FPR number of metrics or other parameters. Relating to Yasmien and Jyoti the author acknowledges the measurement of efficiency of IDS, also stating important aspects of evaluation lie in the DR, FPR, FNR, which provide quantifiable evidence of effectiveness in ID. For an IDS to be evaluated in those terms, the nature of analysed information must be normal. On the other hand, knowing the nature of this analysed information is not required during the intrusion detection process and is only necessary in evaluating IDS efficiency.

Zhang and Wang [34] investigate current feature selection techniques in a bid to build an effective solution. The author uses for commonly used in selection methods to elucidate features that are least important in the dataset. IG, ReliefF, GR, ChiSquare are used over the full NSL-KDD dataset to identify 20 important attributes. These 20 features are applied to the proposed Bayesian network classification model for feature selection. The proposed model calculates the most useful features from the 20 further reducing the figure. A investigation is conducted to compare and accuracy of commonly used feature selection methods as well as the results of the proposed method. The authors conduct tests over the benchmark dataset NSL-KDD with all of its records in the training set and 10 fold cross validation for training and testing. A comparison between the filter and wrapper approach are made with the wrapper being chosen to the fact that it evaluate features by performance of the classifier. The subset results in the best performing classifiers being selected. Bayesian networks are accepted of a model to the widespread belief that it is suited in working under uncertainty.

Relan and Patil [35] study effective ways of feature selection, comparing two algorithms. The C4.5 , decision tree algorithm and the C4.5 algorithms pruning and testing proposed features over the KDD 99 and NSL kDD dataset to test and train the classifier algorithm. After identifying decision tree technique as a logical method with advantages and extracting features and rules. The authors identify that to train machine algorithms historical data is necessary, and since the KDD dataset holds up to 4,94,020 records they choose NSL KDD to train and test the dataset. The authors decide on reducing the features in the training data. The author acknowledges the use of C4.5, which uses information gain and splitting criteria to deal with continuous and discrete attributes and uses pruning to ensure that over fitting the decision tree does not occur. When training the authors alternated between selecting the 10 fold cross validation technique and the partitioning methods, randomly choosing the percentage of the dataset to use for training and testing. The two classification algorithms were tested upon the two different datasets c4.5 decision tree and c4.5 with pruning only discrete value attributes were considered during classification. The time required for testing was less than the training classifier and the results showed that the c4.5 with pruning algorithm performed better than the C4.5 algorithm. After training the classifier KDDCup 99 and NSL-KDD test data is tested against both c4.5 decision tree and c4.5 decision tree

with pruning. During C4.5 decision tree algorithm the author considered all the 41 attributes of KDDCup 99 dataset while at the time of C4.5 with pruning only discrete value attributes like protocol_type, Service, flag, land, logged_in, is_host_login, is_guest_login and class are considered during classification. The performance is measured in terms of classifier accuracy, percentage of true positive, percentage of false positive and time required for testing. The time required for testing is less as compared to training the classifier. The results generated by both the classifiers are compared with each other as shown below.

IX. DATA MINING TOOLS

There are various datamining tools that may be used to conduct investigations. In Zupans publication the author highlights the significance in choosing the correct data mining, by first theoretically matching the correct classifier technique to the field where it would work [36]. Understanding characteristics such as fundamentals operations, the way it works phases of work then, doing the same with current data mining tools to identify patterns and ensure appropriate functionality and features are installed to create a model. Once this has been completed the author can begin to use the chosen methods in the medical biometrics field. The investigation helped illustrate the effectiveness of The WEKA tool in diagnosing Leukaemia. Many researchers share the same views expressed by the author, with several data mining investigations carried out notably through the WEKA tool rather than other popular data mining tools, such as rapid miner, KNIME and so on[36].

Most notably, they are a number of publications that study algorithms on the DARPA dataset, the KDD cup dataset and the NSL – KDD dataset which used the WEKA environment. When testing upon Knowledge discovery databases it is noted by Jagtap that WEKA supports many standard data mining tasks such as data integration, selection transformation and evaluation [37]. Knowledge extraction is a key process for businesses tools such as WEKA that allow for clear, operations or all levels of users is significant [37]. Furthermore, the report by the Pharmine company states that WEKA achieved the highest performance in accuracy amongst the data mining tools [36] [37]. Weka also offers some functionality that other tools do not, such as the ability to run up to six classifiers on all datasets, handling multi-class datasets which other tools continue to struggle with tools [36] [37]. Frank states that this effectively means that complex critical algorithms may be experimented on in this complimenting environment allowing innovative flexible research while decreasing technological limitations in research [38].

X. CONCLUSION

In recent years there has been a large interest in identifying the best feature set attributes for IDS classifiers. With the growing number of intrusions reported there is cause for creating accurate IDSs with low percentages of false positives. Data mining based IDSs have demonstrated higher accuracy, to novel types of intrusion and robust behaviour. Furthermore, it has been noted that intrusion detection must keep up with the sheer size, speed and dynamics which modern networks are expected to operate on. Many have tackled the issue in research

papers with various datasets, namely the Knowledge Database Discovery cup 99. Fewer attempts have been made to adapt this test into the NSL-KDD dataset. NSL-KDD is noted to be one of the best representations of network traffic in the current time frame. The need for more sophisticated and adaptive IDS systems remain and Industry Professionals and academics continue to develop and present novel methods that can cope with new more sophisticated attacks.

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Dynamic Crypto Algorithm for Real-Time Applications DCA-RTA, Key Shifting

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Abstract—The need for fast and attack resistance crypto algorithm is challenging issue in the era of the revolution in the information and communication technologies. The previous works presented by the authors “Dynamic Crypto Algorithm for Real-Time Applications DCA_RT A”, still need more enhancements to bring up the DCA_RT A into acceptable security level. In this work, the author added more enhancements on the Transformation-Table that is generated by the Initial-Table IT, which affects the overall encryption/decryption process. The new TT generation proven to be less correlated with the IT than using the previous TT generation processes. The simulated result indicates more randomness in the TT, which means better attack resistance algorithm. More room for algorithm enhancements is still needed.

Keywords—Dynamic crypto algorithm; real time applications; shared key generation; symmetric key encryption

I. INTRODUCTION

The Internet and related Internet services are rapidly changing, in the next era of computing things will be outside the realm of the traditional desktops. Many of the objects surrounding us will be on the network, in which Information and Communication Technologies (ICT) are invisibly embedded in the environment surrounding us [1]. The fast momentum in modern computing toward the Internet of Things (IoT), wherein, different objects like home appliances, processors, TVs, cars, furniture, goods and others, blend seamlessly with the environment around us. [2], the shift toward using mobile computing, mobile applications, smartphones, smart systems and cloud computing. This for sure results in new challenges to the ICT and enforces researchers to find creative solutions to fulfil the increasing demands of higher performance, low latency, more bandwidth and acceptable level of security, or what we call it the network Quality of Service QoS.

One of the main challenges to the (IoT) and the Future Internet (FI), is how to employ the concept of Internet anytime, anywhere and any service without compromising the security and QoS in resource constrained technologies [3]. Most of the proposed security solutions consider privacy, which is the main concern for Internet users. The current Internet privacy implementations mainly employ the existing known security standards, and it needs improvements [4]. Cryptology is the science that aims to provide information security in the digital environment. Cryptology details can be found in the Handbook of Applied Cryptography by [5].

A lot of improvements and (QoS) enhancements are needed to the current Internet privacy, these enhancements should focus on creative solutions that is able to maintain acceptable level of privacy without affecting the processing time. The researchers of this work, have been working since the first release of the proposed algorithm, to add more enhancements over the previous work shown in [6] [7] [8] [9] [10]. Two conference papers, one journal paper and two master students were working on the algorithm, we believe, there are many enhancements areas could be added to the original proposed algorithm, the enhancements for sure will end up with a solid algorithm that can be used in the (IoT/FI). The promising results achieved out of the previous works, provide the researchers with strong confidence of the algorithm success.

The experiment results show a faster encryption/decryption algorithm with minimal processing delay compared to the well-known algorithms, the algorithm still needs a lot of improvements to be robust and durable.

The rest of the paper is structured as follows. Section II presents general cryptographic background, a brief definition of different types of cryptographic techniques were shown. Sections III show the importance and motivation behind the work, the need for fast, secure and attack resistance symmetric algorithm is discussed. Section IV introduces the enhancements proposed by the researchers on the previously published work, the enhancements are supposed to solve some of the short comes of the previous work. Section V presents the main contribution of this paper, namely, the shifting process on the Transformation Table (TT) to eliminate the table deficiencies. Finally, section VI concludes the paper and outlines the future work.

II. BACKGROUND

A. Some Definitions

Cryptology is the study of secret codes, it involves two main branches: cryptography; concerned with writing of a plain text messages using secret code to produce a decrypted message and the creation of these methods, while cryptanalysis; concerned reading encrypted message by breaking the secret code. [11].

Cryptosystem is the method of securing communication. The sender encrypts (or enciphers) a message using an encryption algorithm together with a secret key. This produces a ciphertext which is sent to the recipient. The recipient, who

also possesses a key, receives the ciphertext and decrypts (or decipher) using the key to recover the original message, called the plaintext. [11].

B. Block Cipher

“An n-bit block cipher divides the plaintext into blocks of n-bits, and encrypts one block at a time with a fixed key-dependent invertible transformation. In practice, the block size n is often 64 or 128 bits. A block cipher is a function $E: P \times K \rightarrow C$, with P, K and C the sets of all possible plaintext blocks, keys and ciphertext blocks. $M = C = \{0,1\}^n$; and $K = \{0,1\}^k$. For every key $K \in K$, the encryption function $E(.,K) = E_K(.)$ is invertible and its inverse is denoted $D(.,K) = D_K(.)$. In a secure block cipher it is infeasible to determine K from a number of plaintext/ciphertext pairs faster than trying all possible K, and the permutation on all n-bit words denoted by $E_K(.)$ should be indistinguishable from a random permutation” [12].

C. Cryptanalysis Techniques

Block (or Symmetric) Ciphers are subjected to some popular cryptanalysis techniques. These techniques are differential cryptanalysis, linear cryptanalysis and algebraic cryptanalysis [12]. Any encryption technique should be cryptanalysis resistance.

III. RESEARCH MOTIVATION

The need for fast, secure and resistant to attacks cryptography technique is highly required, nowadays many secure encryption techniques used in RTA (for example 3-DES and AES) provide acceptable security level, but the problem becomes clear when QoS is major requirement [10]. The proposed algorithm, once it is gets completed is expect to be provide noticeably high security level for many reasons; to mention some, it is fast, secure, use variable key size, can change the system key frequently with no extra time overhead, flexible and can be used in the IoT/FI regardless of using desktop, laptop, smartphone or handheld devices.

The problem of having an algorithm that is resistant to Cryptanalysis, Differential Power Analysis (DPA) and Simple Power Analysis attacks (SPA) need to be investigated more. The researchers think that, the proposed algorithm is equipped with an acceptable level of cryptanalysis attacks (the proof needs more investigations), but the DPA and SPA are still major challenges to the algorithm. Resistance to cryptanalysis is not sufficient to have secure cryptosystem [13], where, cryptanalysis attacks since vulnerabilities can raise from other layer of implementations. Cryptanalysis can test cryptographic algorithms in isolation using differential or linear cryptanalysis exploiting statistical characteristics of the algorithm, DPA and SPA can exploit vulnerability of other components and parts of the algorithm rather than statistical and mathematical parts of the system [13].

In [14], the researchers proposed to use conventional encryptions to solve the problem of long processing time instead of modern encryptions, the claimed that “Some of the conventional encryption techniques are very weak and brute force attack and traditional cryptanalysis can be used to easily determine the plain text from encrypted text” [14].

This work is consider a block cipher, but doesn't share the common characteristics and definitions of the block cipher, so it is a novel idea of symmetric encryptions. And the promising results that we had in [10], armed the researches with solid background to proceed further enhancing the algorithm to get a trusted secure and fast one.

IV. RELATED WORK

This work is a contribution to the ongoing research on the original work [6]; our objectives were to build a concrete solid algorithm that can be used in the (IoT/FI) as trusted privacy solution, which is characterized by being transparent to the public, secure and faster than the known algorithms.

The work presented here, tries to achieve better encryption/decryption performance while keeping overhead processing to minimal levels and preserving an acceptable security level. The proposed algorithm is not complicated, so, the operations performed, the key selection process, the key size, the plaintext size, the key insertion process, the encryption and decryption processes make it simple and secure technique.

The proposed algorithm consists of three components, the Index Generation Process (IGP), Encryption Process (EP) and Decryption Process (DP) component. The IGP is common between EP and DP. The steps briefly describe the algorithm:

1) *The IGP is used in both the EP and DP; where the Initiate Table (IT) is shared and announced, it uses a random Shared Value to generate the Transformation Table (TT), and the Table of Indexes (TI) is generated by randomizing the table content.*

2) *The Encryption/Decryption Process EP/DP, followed by Key Insertion process, the result of the final step is the Ciphertext. The inputs to the EP are the plaintext, system key and the table of indexes. The System Key is randomly generated by the user; the key is 1024-bit size. The Scrambled text; is the result of the XoR operation performed over the plaintext and the key. The system Key is inserted inside the scrambled text (XoRed table) according to the value of the Index values.*

3) *The Decryption Process DP starts upon receiving the Ciphertext, then the algorithm extracts the Key out of the Ciphertext, this is called Key-Recovery (KR); the extraction process is the inverse of the key insertion process used in the EP, once the key is retrieved, the DP is performed by XoRing the scrambled message and the system key to obtain the original plaintext.*

The original work was described in [6], [8] and [7] the Dynamic Cryptographic Algorithm for Real Time Applications (DCA-RTA). It consists of three main processes, the Encryption Process (EP), the Decryption Process (DP) and the Index Generation Process (IGP).

The EP, takes the *Plaintext* (P) and the *System-Key* as input, then it performs XoR operations to produce the *Scrambled-Text*, then the EP inserts the *System-Key* inside the *Scrambled-Text* according to the *Table-of-Indexes* (from the IGP process), then the *Cipher-Text* which contains the *system-*

key inside are send to the receiver, who is responsible to extract the key from the received *Cipher-Text* and decrypts the message see Encryption Process (EP) below.

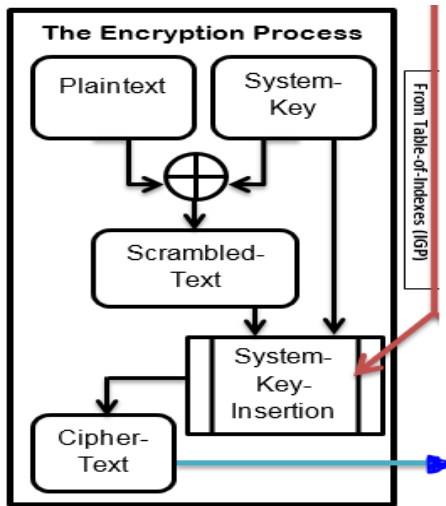


Fig. 1. Encryption Process (EP), [9]

The DP receives the *Cipher-Text* from the (EP) step, it consults the *Table-of-Transformation* to extract the *System-Key* from the received *Cipher-Text*, and then it performs XoR operation between the recovered *System-Key* and the *Scrambled-Text* to get the *Plaintext* (P), see Decryption Process (DP), [9] below.

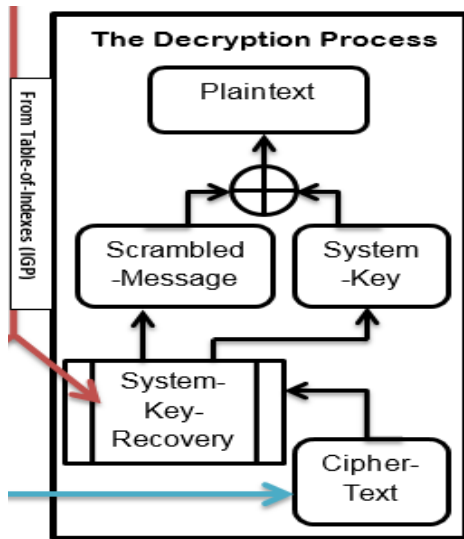


Fig. 2. Decryption Process (DP), [9]

The Index Generation Process (IGP), starts with a fixed announced $(16 \times 16)_{16}$ table, then a randomly generated *Shared-Value* is used to mix up the initial table to produce *Transformation-Table*, then the table is used as a *Table-of-Indexes*, which is used in both (EP) and (DP) processes. See Index Generation Process (IGP), [9].

The complete architecture of the algorithm is summarized in

Complete DCA-RTA Architecture Algorithm, [9]

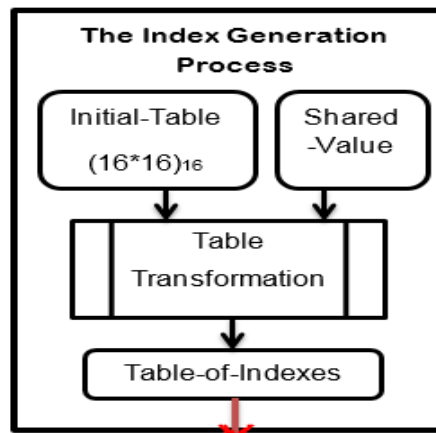


Fig. 3. Index Generation Process (IGP), [9]

In [6], the complete architecture was proposed as described above, the researchers achieved promising results, and the DCA-RTA algorithm was compared with the Advanced Encryption Standard (AES), where DCA-RTA found to be in average ten times faster than AES in Encryption/Decryption processes. In spite of the promising results, the algorithm at that time was seeking for proof of concepts, and finally it works and gave positive results.

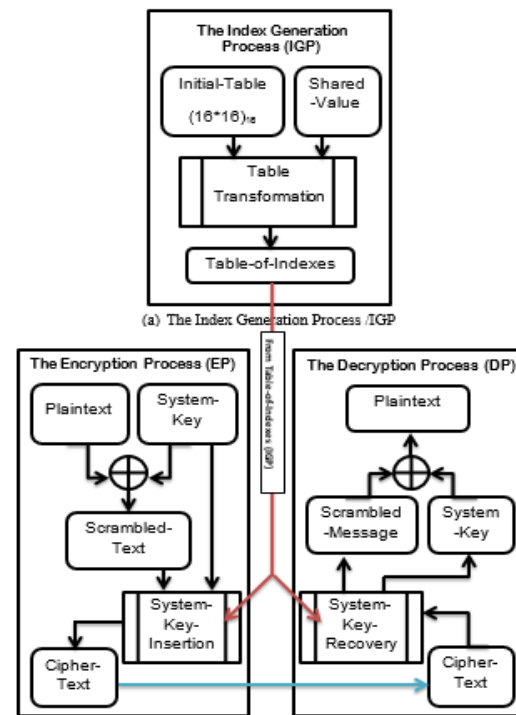


Fig. 4. Complete DCA-RTA Architecture Algorithm, [9]

In [8], the researchers proposed a ten digit shared-key value and the use of different sequences of Indexes to insert the *System-Key* inside the *Scrambled-Text* table that represents the *Cipher-Text* to be send.

In [7], the researchers implements the suggested enhancements on the DCA-RTA, and they improve the use of the *Table of Indexes*, in this work, no significant

enhancements were achieved, the aim of the work is to prove the correctness of the enhanced algorithm.

In [9], major enhancements on the original DCA-RTA were achieved, the researchers believe that, the initial algorithm parameters should base on scientific background, for this reason, the following parameters were tackled:

1) *The System-Key-Insertion sub-process needs more tests, to prove the algorithm requirements according to the modern symmetric algorithms requirements.*

2) *The best Shared-Value length needs to be examined and set; since it was set to 10-digits without enough proves.*

3) *The best Plain-Text size should be determined.*

4) *Table-Transformation generation needs more intelligent processing in order to get rid of repeated patterns or segmentations problems.*

Abeer A. Al-Omari, studied the algorithm from different perspectives and proposed some modifications and enhancements on the original DCA-RTA. From the performance perspective, she compared her work using the default key size (128 bytes) with AES under standard key size used in AES (16, 24 and 32 bytes), she found that her approach outperforms AES more than seven times for both encryption and decryption processes.

Abeer has added the following enhancements on the original DCA-RTA, the *Table-Transformation* building process now based on swapping one row with one column based on the *Shared-Value*. The *System-Key-Insertion*, was suffering from bad segmentation, or the *System-Key* was not distributed efficiently inside the *Plain-Text*, Abeer enhanced the *System-Key-Insertion* process by inserting the *System-Key* inside the *Scrambled-Text* from both sides. Abeer found that the proper *key-size* of 128 byte is enough to produce the required level of performance and complexity, Abeer also defined the plain-text size to be *190-bytes* long, this size guarantees the insertion process will always be within the *Plain-Text* boundaries.

V. RESEARCH IMPROVEMENTS

A. Reasons for Improvements

An encrypted message must not reveal any information about its origin, so the cryptosystem must make it look as random as possible [15]. In [9] The *Table-Transformation* (TT) is used to produce the *Table-of-Indexes* (TI) which affects the overall security of the proposed encryption algorithm. The researchers proposed five methods to produce the (TI); the methods were Circular Shifts, Swap Cells, Columns Row Swapping (CRSw), same Column Row Swapping or Row Column Swapping (RCSw). The researchers found that the (RCSw) is the best method could be used to produce the (TI). The work presented in [9], suggested randomizing the Initial Table (IT) by swapping one complete row with one complete column, the idea is to add more complexity and randomization on the resulting Transformation Table (TT). The Row Columns Swapping process (RCSw) used in [7] was excluded in [9] since it produced a segmented

(TI). The swapping method (RCSw) was considered the best method for the following reasons:

- RCSw was proven to be the best method among other five methods used.
- Using 10-digit Shared-Value (SV), and no need to go beyond this value.
- RCSw using 10-digit Shared-Value produces more segments than the other proposed methods. (More segments mean more cells displacements).

In this work, the researchers believe, more enhancements can be added to the (RCSw) process to get rid of the deficiencies appear on the Transformation Table (TT). The resulted (TT) suffers from the stationary of some cells, they didn't move, also, the resulted (TT) is still segmented, where group of cells were moved for the same distance. By combining the (RCSw) and gets benefits of the idea of the method used in [7] and [9], we could find an optimal method that can eliminate stationary cells and cells segmentations. The optimal method that is free of stationary cells and has less cells segmentations might help to defend the algorithm against cryptanalysis and brute force attacks.

B. The Proposed Improvements

Although, the Row Column Swapping (RCSw) process presented in [9] was used; the proposed enhancement in this work, suggests not only to shift some selective rows and columns, but to shift the entire table including all rows and columns. The researchers believe shifting the whole table contents even for at least one cell distance will add more complexity and randomness over the TT and the TI as well.

The original proposed algorithm consists of many steps; one of them (which are used here) is to generate the *Table-Transformation* (TT) out of the *Initial-Table* (IT), the generation process based on the *random Shared-Value* as an input to produce the *Table-of-Indexes* that is used by *System-Key-Insertion* and *System-Key-Recovery*. The *System-Key-Insertion* inserts the *System-Key* inside the *Scrambled-Text*. The insertion process employs the *Table-of-Indexes* to decide where the *System-Key* is to be inserted, the extracted index is a two-digit length extracted out of the *Table-of-Indexes*.

The new improved process is basically based on the RCSw process, the row column shifting process depends on the value of the last two digits of the Shared-Value. The new method is called Row Colum Shifting (RCSf).

After performing the RCSw process, the whole columns are circularly shifted to the right according to the value of $(n-1)$ digit of the Shared-Value, also the whole rows are circularly shifted down according to the value of the (n^{th}) digit of the Shared-Value. The shifting processes the Columns Right Circular Shifts (CRCS) and the Rows Down Circular Shifts (RDCS), guarantee the movement of each individual cell of the IT.

Our hypothesis says, Columns Right Circular Shifts (CRCS) and Rows Down Circular Shifts (RDCS) on the whole table (IT), will produce more randomize (TT) than using (RCSw) only.

The new method formula representation is $RCSf = (RCSw \text{ followed by } CRCS \text{ and } RDCS)$; Row Column Shifting is the process of Column Right Circular Shifts followed by Row Down Circular Shifts

C. The Implementation

To validate the correctness of the proposed method, we performed the tests on the same (IT) using two transformation methods RCSw and RCSf methods. We run both methods using seven different Shared-Value lengths (10, 20, 30, 40, 80, 100 and 1000) digits respectively, each Shared-Value was run 30 times, the total number of runs was $(7 \text{ Shared-Values} * 30 \text{ runs} * 2 \text{ methods}) = 420 \text{ runs}$. After the completing the 30 runs of each individual Shared-Value length, we calculated the new cells locations, then, we ascendingly ordered the resulting TT based on the number of cell displacement, we noticed that, some cells didn't move while others were moved to 1, 2, 3 up to 29 cells, the cells movement ranged from (0 to 29), then we indexed each cell in the new TT table starting from 1 to 256 (the total number of cells are 256 cells). For each Shared-Value runs (30 runs), we calculated the average cells displacement, cells variances, variances between groups and within groups and correlations for each method

D. Research Analysis

RCSw was the best of other 4 methods specially in generating the will-mixed-table [], Method-1 could not generate the will-mixed-table but it had one positive advantage over than RCSw, all the cells of table-of-indexes resulted from Method-1 were changed their location while in RCSw some cells were not changing their initial location specially when using shore Shared-Value length (e.g. 10 or 20).

Since RCSw was the best and Method-1 had one advantage over RCSw, we decided to improve RCSw by taking advantage of Method-1.

The new method was called RCSf, by comparing the Table-of-Indexes resulted from two methods we find that:

- RCSf is better than RCSw when using short Shared-Value-Length, (Fig.5) shows that when using Shared-Value-Length 10, all cells were displaced and changed their location in RCSf, while about 66 cells were not displaced in RCSw, (Fig. 6) shows the resulted tables for both RCSw and RCSf when Shared-Value-Length = 20, it's obvious that increasing Shared-Value-Length improve results of RCSw in terms of number of displaced cells.
- From the correlation table (Table 1), it is obvious that the Shared-Value length = 10, in RCSW indicates very strong correlation between the TT and IT tables, while RCSF shows week correlation between the tables.
- It is notes that while extending the Shared-Value to longer length size, RCSF overweight RCSW obviously for the rest of the Shared-Value lengths.
- The negative correlation in RCSF indicates the randomness of the resulted TT using Shred-Value more than 30 digits length.

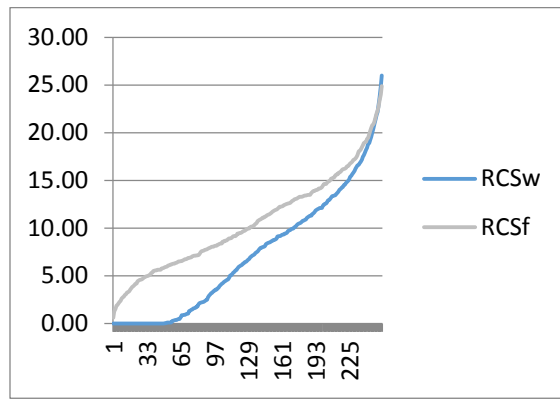


Fig. 5. Shared-Value Length 10

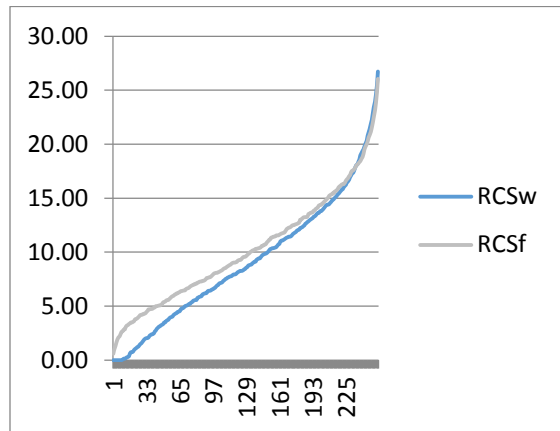


Fig. 6. Shared-Value Length 20

TABLE I. CORRELATION TABLE

Shared Value Length	RCSW	RCSF
10	0.430432021	0.031555991
20	0.149981165	0.00909278
30	0.112991412	-0.029247134
40	0.040089742	-0.008268769
80	-0.007915713	-0.024461118
100	0.002503195	-0.012060659
1000	0.025229386	0.006841883

- The overall assessment of the resulted TT tables by both methods (RCSw and RCSf), definitely indicates the adoption of the new method RCSF.

VI. CONCLUSION

Row Column Swapping followed by Shifting on the Initial Table IT to produce the Transformation Table TT, based on the random values generated by the Shared-Value, even with a relatively small value length (i.e. 10 digits length) produces randomized table, which makes it harder to the cryptanalysis attacks to break the algorithm.

The authors recommend to use Shared-Key value length not less than 30 digits length, in this case the correlation between the IT and TT is weak, here, the attacker will face attack resistance algorithm.

VII. FUTURE WORK

More enhancements on the algorithm are exists, where researchers can add substitution techniques, transformation methods, determine the best time to change the Shared-Value or the encryption key, the key exchanging scheme and the best message size to maintain less processing delay.

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Human Object Tracking in Nonsubsampled Contourlet Domain

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Abstract—The intelligent systems are becoming more important in life. Moving objects tracking is one of the tasks of intelligent systems. This paper proposes the algorithm to track the object in the street. The proposed method uses the amplitude of zernike moment on nonsubsampled contourlet transform to track object depending on context awareness. The algorithm has also been processed successfully such cases as the new object detection, object detection obscured after they reappeared, detecting and tracking objects which successfully intertwined and then separated again. The proposed method tested on a standard large dataset like PEST dataset, CAVIAR dataset and SUN dataset. The author has compared the results with the other recent methods. Experimental results of the proposed method performed well compared to the other methods.

Keywords—object tracking; zernike moment; nonsubsampled contourlet transform; context awareness; extracting features

I. INTRODUCTION

The intelligent systems are becoming more important in life. Building an intelligent surveillance system can be split into four main challenges: moving object detection, object classification, object tracking and behavior recognition. Moving object tracking is one of the tasks of intelligent systems. Object tracking can be applied in many place such as security surveillance systems for airports, train stations, schools, or buildings of government etc.; or traffic control systems like automatic traffic signal systems, street-crossing safety systems, traffic density statistics systems etc.

In past time, there are many researchers who proposed the methods to track moving objects. Most of these methods are divided into four groups such as contour-based [1], region-based [2], feature-based [3] and model-based [4] algorithms. Moving object detection used common techniques such as: background subtraction, statistical models, temporal differencing and optical flow [11, 12]. Algorithm based on background subtraction utilizes the current image to compare it with the background image and detect the moving scene. Most methods of background subtraction are median filter, mean filter, temporal median filter, Kalman filter, sequential kernel density approximation and Eigen backgrounds [11, 13]. The mean-shift algorithm using colors has been used to track the objects in video. This method has improved tracking results [5, 6, 7]. However, the implementations of them for object blur are complex. The object tracking algorithms using feature are performed based on the point, shape and contour in many domains [8, 9, 10]. Johnsen [14] gave the model to track

objects through static camera with motion detection technique using background subtraction in which the background generated using filters approximated Median Filter. After detecting the moving area, he uses two-pass connected component labeling for identifying and selecting the motion area to make for object tracking step. Next, the model boundary objects are separated using standard RGB color and use the Kalman filter to predict the next location of the object. Finally, using apriori assignment combined with Euclidean metric distance, Bhattacharya to track objects. The drawback of this method is to require the appropriate reference background image. If there are changes of light compared to the reference background image, the current area will be considered as the motion that should not be able to detect moving objects. The object tracking is hard work. This task has many challenges.

In this paper, the author proposes a method to implement for human tracking based on their contour. The proposed method uses the amplitude of zernike moment on NonSubsampled Contourlet Transform (NSCT) to track object depending on context awareness. The algorithm has also been processed successfully such cases as the new object detection, object detection obscured after they reappeared, detecting and tracking objects which successfully intertwined and then separated again. The proposed method tested on a standard large dataset like PEST (Performance Evaluation of Tracking and Surveillance) dataset, CAVIAR (Context Aware Vision using Image-based Active Recognition) dataset and SUN (Scene UNDERstanding) dataset. The author has compared the results with the other recent methods. Experimental results of the proposed method performed well compared to the other methods such as Kernel Filter method [14], Particle Filter method [15], curvelet method [16] and contourlet method [17]. The rest of this paper is organized as follows: in section 2, the author described the basic of NSCT, zernike moment and its advantages for human object tracking. And details of the proposed method for human object tracking are presented in section 3. The result and conclusion of the paper are presented in section 4 and 5 respectively.

II. SELECT A NEW GENERATION WAVELET TRANSFORM FOR TRACKING

The Discrete Wavelet Transform (DWT) provides a fast, local, sparse, multiresolution analysis of real-world signals and images. Although DWT is a powerful tool for signal and image analysis, it has three serious disadvantages: shift

sensitivity, poor directionality and lack of phase information. To improve these drawbacks, the new generation wavelet transforms such as the ridgelets, curvelets, contourlet transform and NSCT have been proposed.

Curvelets and ridgelets take the form of basic elements which exhibit very high directional sensitivity and are highly anisotropic. In two-dimensions, for instance, curvelets are localized not only in position (the spatial domain) and scale (the frequency domain), but also in orientation. Unlike wavelet transforms, the ridgelet transform processes data by first computing integrals over lines with all kinds of orientations and locations. The curvelet transform, like the wavelet transform, is a multiscale transform, with frame elements indexed by scale and location parameters. Unlike the wavelet transform, it has directional parameters, and curvelet pyramid contain elements with a very high degree of directional specificity. In addition, the curvelet transform is based on a certain anisotropic scaling principle which is quite different from the isotropic scaling of wavelets. However, curvelet also has two drawbacks: first, not optimal for sparse approximation of curve beyond C^2 singularities and second, highly redundant [18].

Contourlet transform [19] is built from a discrete domain first, then extend to the continuous domain, it has lower redundancy and a faster discrete implementation version than curvelet. But contourlet is just multidirectional and multi-scale but not shift - invariant, which causes pseudo-Gibbs phenomena visible on the decoded image by high compression ratio. NSCT [20] brings shift - invariance for contourlet with the trade-off of more redundancy. As NSCT is used for contour detection, this redundancy is not a drawback but even gives better results.

The important feature of the proposal bases on object's contour and NSCT is chosen to extract the object's contour. NSCT belongs to the family of the new generation wavelet transform. The NSCT development through its main predecessors may start at wavelet transform [21] as in [22]. The properties of NSCT like multi-scale, multi-direction, shift-invariant make it very suitable for contour detection.

A. NonSubsampled contourlet transform

NonSubsampled Pyramid (NSP) is similar to Laplacian pyramid. This algorithm uses two-channel nonsubsampled filter banks and decomposition is realized with level $j = 3$ stages. The filters of subsequent stages are the result of upsampling filters of the first stage. The subbands are divided into high-pass and low-pass filter. The region of low-pass filter is $[-\frac{\pi}{2j}, \frac{\pi}{2j}]^2$, and the complement of the low-pass filter is the high-pass filter which is solved by the region $[-\frac{\pi}{2j-1}, \frac{\pi}{2j-1}]^2 \setminus [-\frac{\pi}{2j}, \frac{\pi}{2j}]^2$.

Directional Filter Bank (DFB) eliminates the down-samplers and up-samplers in each two-channel filter bank that is NonSubsampled Directional Filter Bank (NSDFB). NSCT is the process that combines NSP and NSDFB. Ping [23] also asserts that NSCT is implemented by two major steps: when multi-scale image multi-scale decomposition is NSP filter to be used in a low-pass subband and a band-pass subband. Then,

each level of band pass subbands directional decomposition of band-pass subbands will continue with NSDFB. And image decomposition process is complete with two steps: NSP and NSDFB.

B. Zernike Moment (ZM)

ZM [24] was firstly introduced by Teague [25] to overcome the shortcomings of information redundancy present in geometric moments [25]. ZM can represent the properties of an image with no redundancy or overlap of information between the moments [26]. Due to these characteristics, ZM has been utilized as a feature set in different applications such as object classification, shape analysis, content based image retrieval etc.

ZM owns the following properties [24, 27, 28]:

- (i) ZM is rotation, translation and scale invariant [29].
- (ii) ZM is robust to noise and minor variation in shape.
- (iii) Since the basis of ZM is orthogonal, therefore they have minimum information redundancy.
- (iv) ZM can characterize the global shape of pattern. Lower order moments represent the global shape pattern and higher order moment represents the detail.
- (v) An image can be better described by a small set of its ZM than any other types of moments [24].

ZM is a set of complex polynomial which forms a complete orthogonal set over the interior of the unit circle of $x^2 + y^2 \leq 1$ [30, 31]. These polynomials are of the form,

$$V_{mn}(x, y) = V_{mn}(r, \theta) = R_{mn}(r) \cdot \exp(jn\theta) \quad (1)$$

where m is positive integer and n is integer subject to constraints $m-n$ even and $|n| \leq m$, r is the length of vector from the origin to pixel (x, y) and θ is the angle between vector r and x -axis in counter clock wise direction, $R_{mn}(r)$ is the zernike radial polynomials in (r, θ) polar coordinates and defined as

$$R_{mn}(r) = \sum_{s=0}^{\lfloor \frac{m-|n|}{2} \rfloor} \frac{(-1)^s (m-s)! r^{m-2s}}{s! \left(\frac{m+|n|}{2} - s\right)! \left(\frac{m-|n|}{2} - s\right)!} \quad (2)$$

here $R_{m,-n}(r) = R_{mn}(r)$

The above-mentioned polynomial in equation (2) is orthogonal and satisfies the orthogonality principle:

$$\iint_{x^2+y^2 \leq 1} V_{mn}(x, y) \cdot V_{pq}(x, y) dx dy = \frac{\pi}{n+1} \delta_{mp} \delta_{nq} \quad (3)$$

where, $\delta_{ab} = 1$ for $(a = b)$ and $\delta_{ab} = 0$ otherwise, is the Kronecker symbol.

ZM is the projection of image function $I(x, y)$ onto these orthogonal basis functions. The orthogonality condition simplifies the representation of the original image because generated moments are independent [24, 32].

The ZM of order m with repetition n for a continuous image function $I(x, y)$ that vanishes outside the unit circle is

$$Z_{nm} = \frac{m+1}{\pi} \iint_{x^2+y^2 \leq 1} I(x, y) [V_{nm}(r, \theta)] dx dy \quad (4)$$

In case of digital image, the integral are replaced by summation, given as follows

$$Z_{nm} = \frac{m+1}{\pi} \sum_x \sum_y I(x, y) V_{nm}(r, \theta), \quad x^2 + y^2 \leq 1 \quad (5)$$

ZM has its own properties such as: shift-invariant [33], translation invariant [28], rotation invariant [29] etc. So if the author use the combination of NSCT and ZM features in one methodology then the author can expect for more accurate object tracking results.

III. OBJECT TRACKING BASED ON NSCT COMBINED WITH ZERNIKE MOMENT

In this section, the author proposed the method for object tracking based on NSCT combined with ZM depending on context awareness. The overall of the proposed method is to present as figure 1.

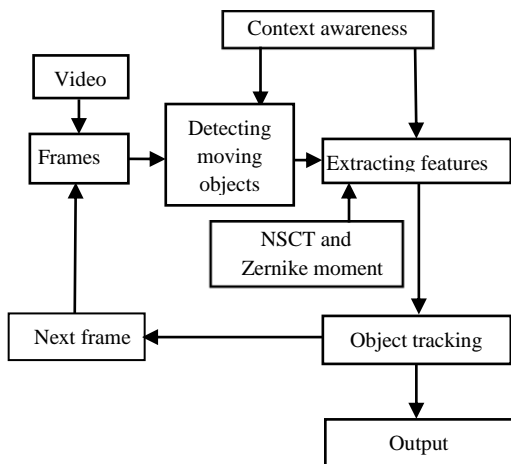


Fig. 1. The overall of proposed method

The proposed method has three main stages: moving object detection, features extraction and object tracking. Firstly, the input data is videos that have the same serial frames. The author detects moving objects. Secondly, the extracting features of object. In this stage, the author uses NSCT and zernike combined with context awareness to extract features. Finally, object tracking. The author uses a contour object for object tracking from frame to frame.

A. Moving object detection

Most of the moving object detection methods take two consecutive input images and return the locations where differences are identified. The motion of an object can cause these differences.

A video sequence contains a series of frames. Each frame can be considered as an image. The common approach for detection of objects consists of three steps: background modeling, foreground detection and data validation. The

author assumes there are only two modes for each pixel in a single frame: background and foreground. The basic of background subtraction method is to compare the frame background with a threshold T which the author is pre-defined. If the difference of a pixel is smaller than T , then it is background, otherwise, it is foreground. To detect objects, the NSCT coefficients and their statistical values were extracted as the features of object images. The author defines a discrete warped NSCT which goes across the region boundaries based on context awareness. The author computes the image sample values in each region of the partition and also describes its implementation together with the inverse resampling. A warped NSCT with a sub-band filtering along the flow lines is implemented. At the boundaries, warped NSCT still have two vanishing moments. The NSCT coefficients of a discrete image are computed with a filter bank. This method reduces computation time significantly by utilizing the characteristics of high correlation between adjacent frames. Because the data are highly correlated pixels in each frame, there are two possibilities for the NSCT element of the consecutive frame will be equal.

The algorithm uses a diagram to check the repetitive element NSCT between two consecutive frames depending on the context awareness in video, thereby reducing the frequency of calculation of the NSCT calculation. The results have showed that this method improved significantly reduces the computation time, and it goes beyond real-time requirements.

B. Feature Extraction

Most previous definitions of context are available that context awareness look at who's, where's, when's and what's of entities and use this information to determine why the situation is occurring. Here, the author's definition of context is:

“Context is any information that can be used to characterize the situation of an image such as: pixel, noise, strong edge, weak edge in image that is considered relevant to the interaction between pixels and pixels, including noise, weak and strong edge themselves.”

In video processing, if a piece of information can be used to characterize the situation of a participant in an interaction, then that information is context. Contextual information can be stored in feature maps on themselves. Contextual information is collected over a large part of the scene.

As [18], the author will be calculated two values as a feature vector: Aspect Ratio (AR) and zernike NSCT value for each bounding boxed object. AR is defined by:

$$AR = \text{width of bounding box} / \text{weight of the bounding box} \quad (6)$$

The bounding box of human usually has less in width than height, and the opposite for car. This feature may fail in cases that human sits down and his height is just a half of the standing pose or poses with two raising arms may increase the width of the bounding box and break the assumption. For the car, that is the perspective of the camera. If the camera view is in the same line of a moving car, it will show the longer

dimension in height not width. But in many of the outdoor cases, the assumption is true. The author chooses AR feature because it is simple, fast and its nature is completely different so that the union of failed case set of them is smaller.

Zernike NSCT value is a value that represents the contour property of objects and used to differentiate between a human and a car. It is calculated as the amplitude of zernike moment on contour binary images of the bounding boxed object as followings: first, the bounding boxed object image is contour detected by applying NSCT [18, 20] decomposition on it. The n levels parameter is [0, 1, 3]. This parameter is has 3 numbers, which means using 3 pyramidal levels (from coarser to finer scale). The first number – zero means at level 1 of pyramid, the level of directional filter bank decomposition will be 2 exponent zero to 1. It is the contour image received which is synthesized from two next levels [18]. The second level of pyramid is 2 exponents 1 to 2. It's $\pi/2^1$ so the author may see it like the two images with horizontal and vertical ways of energy. Similarly, the third level is $\pi/2^3$, which is 8 images with rotation of energy. The synthesis of from the third through the second and to the first level gives us contour image with energy at all ways keeping. The number of levels and direction number at each level is chosen to be computationally efficient and good enough to reflex the contour of object. In my experiment, NSCT is quite slow, but three levels are also enough to do the job. Not all contour images are perfectly detected but it is good enough for the classification result. Detailed implementation of NSCT can be found in [20].

Second, the contour image is converted to the binary image based on threshold which is just simply the mean of the pixel values in image. This is just a preparatory step for zernike, which is done due to [18, 34] that the binary image with just contour point is faster and more accurate than the original image for zernike.

Third, the binary image is passed through zernike moment to get amplitude [18, 20]. Zernike moment is rotationally invariant, so it is suitable for characterizing contour of object, which may be changed because of the various activities and this property reduces effort in training many poses which are just the rotations of another. Detailed implementation of zernike and its amplitude can be found in [20].

C. Object tracking

Tracking task is to identify an object in the current frame and the previous frame. There are many cases in the tracking task such as new objects appear, the object is obscured or disappear, and the object intertwines and then splits again. The tracking method is used optical flow [35] on the points of the contour of the object, to track the object through each frame.

To solve the problem of objects tracking, the author defines an equivalent coefficient $p(m, m')$ between an object of the frame previous $R_{prev}(m)$ and the object of the current frame $R_{cur}(m')$ as follows [35]:

$$\{p_1, \dots, p_{n_i}\} \in S_i \quad (i = 1, \dots, m) \quad (7)$$

$$\{S_1, \dots, S_m\} \in R_{prev}(m) \quad (8)$$

and

$$\{p'_1, \dots, p'_{n'_j}\} \in S'_j \quad (j = 1, \dots, m') \quad (9)$$

$$\{S'_1, \dots, S'_{m'}\} \in R_{cur}(m') \quad (10)$$

The equivalent coefficient $p(m, m')$ is defined as:

$$p(m, m') = \frac{\sum_{i=1}^m T_i}{m} \quad (11)$$

where $T_i = \begin{cases} 1 & \text{if } S_i \text{ compatible with at least one } S'_j \\ 0 & \text{otherwise} \end{cases}$

Two contour S_j and S'_j called incompatible if the equivalent coefficient $q(S_i, S'_j)$ met the following conditions:

$$q(S_i, S'_j) = \frac{\sum_{k=1}^{n_i} X_k}{n_i} > \beta \quad (12)$$

where, β is the threshold compatibility of the two contour with $0 \leq \beta \leq 1$ and $X_k = 1$ if $p_k(x, y) \in S_i$ satisfy (13) and (14), otherwise $X_k = 0$.

$$\min_{p_i \in S_j} \{|x + u_k - x'_i|\} < \alpha_u \quad (13)$$

$$\min_{p_i \in S_j} \{|y + v_k - y'_i|\} < \alpha_v \quad (14)$$

where, (u_k, v_k) is the velocity vector of point $p_k(x, y)$ obtained from the results of optical flow; (x'_i, y'_i) is the coordinates of points $p'_i \in S'_j$ and α_u, α_v are the threshold constant [35].

After calculating the equivalent coefficient $p(m, m')$, two objects $R_{prev}(m)$ and $R_{cur}(m')$ are considered duplicate if

$$p(m, m') > \Omega \quad (15)$$

where $0 \leq \Omega \leq 1$ is threshold constant.

The step of the tracking period as:

Initial:
 C_i is the contour of the i^{th} object in the previous frame.
 C_j is the contour of objects in the current frame number j .
 $p(C_i, C_j)$ is the coefficient equivalents of C_i contour and C_j contour.
 Step 1: take contour C_i and set $n = 0$
 Step 2: take contour C_j
 if $p(C_i, C_j)$ satisfied of (15) then
 $n = n + 1$;
 else
 if contour j is final contour then
 if ($n > 1$)
 C_i is separated into n object (C_j)
 goto step 3
 if $n = 1$ then
 C_i coincides with C_j
 if $n = 0$ then
 if C_j outside frame
 delete C_j
 else

```

        Cj is obscured
    else
        goto step 2
Step 3: if contour ith is final contour then
    Stop
else
    goto step 1
    
```

IV. EXPERIMENTS AND RESULTS

In this section, the author applies the procedure described in section 3 and achieved a superior performance in my experiments as demonstrated in this section. For performance evaluation, author compares the results of the proposed method with the methods such as Kernel filter method [14], Particle filter method [15], curvelet method [16] and contourlet method [17].

The author uses the standard dataset PETS dataset [36], CAVIAR dataset [37], SUN dataset [38], etc to experiment and evaluate. These video datasets are the video datasets in computer vision field. Hard thresholding is applied to the coefficients after decomposition in NSCT domain. For the tracking part, the author determines the object in each frame of the video. The object area is determined in the first frame; the tracking algorithm needs to track the object from frame to frame. Here, the author reports the results on some video clips. Figure 2 shows some scenes in PETS dataset. The experimental approach is as follows.

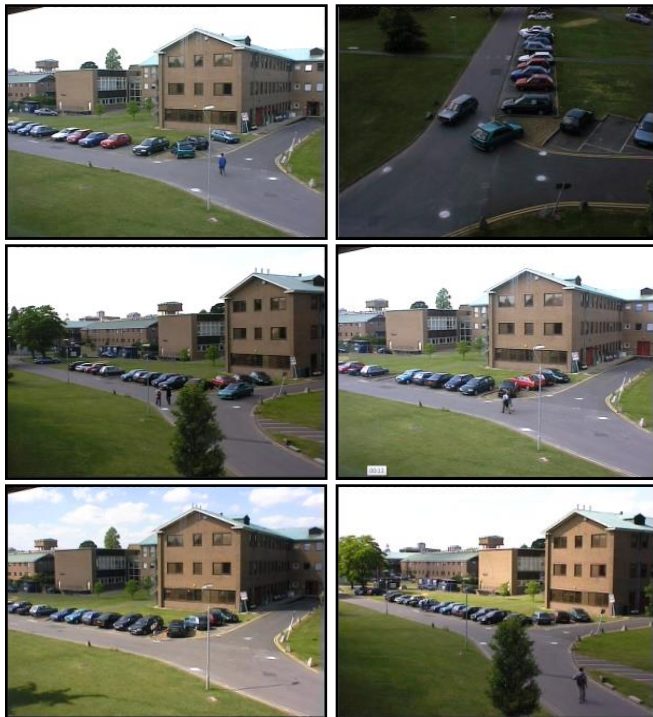


Fig. 2. Scenes in test dataset

The experiments are person video clips with the frame size 288 by 352. The proposed method processes this video clips at 28 frames/second. The author has experimented on the video up to 5000 frames. Here, the author reports the results up to 400 frames. Some results achieved as shown in figure 3.

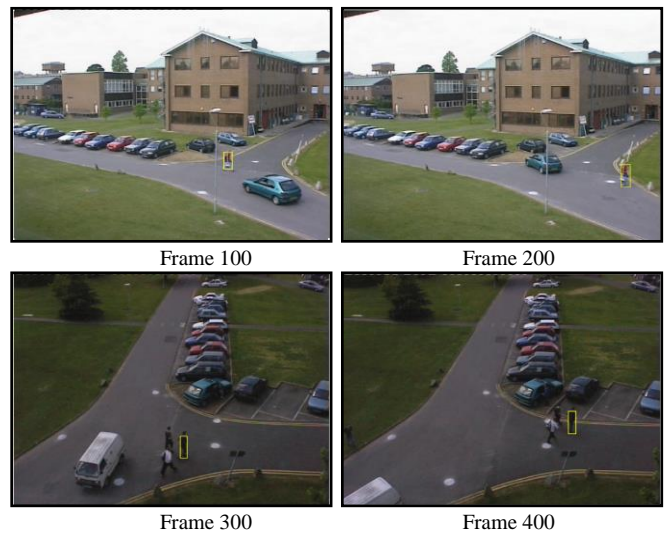


Fig. 3. Tracking in human video clips up to 400 frames

The other of the experiments is person video clips with the frame size 288 by 352. The proposed method processes this video clips at 25 frames/second. The author has experimented on the video up to 2000 frames. Here, the author reports the results up to 250 frames. Some results achieved as shown in figure 4.

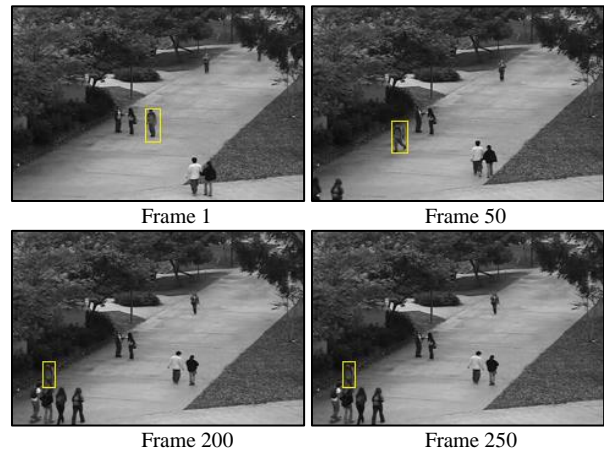


Fig. 4. Tracking in human video train clips up to 250 frames

In these figures, the author observes that the proposed method performs well.

TABLE I. COMPARING THE OBJECT TRACKING ERROR OF THE PROPOSED METHOD AND OTHER METHODS

Frame Number	Kernel method [14]	Partial method [15]	Curvelet tracking [16]	Contourlet Tracking [17]	Proposed method
100					
150					
200	Error				
250					
300		Error	Error		
350				Error	

Frame Number	Kernel method [14]	Partial method [15]	Curvelet tracking [16]	Contourlet Tracking [17]	Proposed method
400	Error				
450					
500		Error			
550					
600	Error	Error	Error	Error	Error
650					
700	Error				
750		Error	Error		
800					
850	Error				
900	Error			Error	
950					
1000		Error			

Table 1 compares the human tracking error between the proposed method and other methods. The error in this case means that the tracking algorithm could not track the correct object. In the table 1, the author sees that the number of errors of the proposed method is less than other methods.

Now, the author uses the Euclidean distance values to perform the evaluation of tracking. The Euclidean distance between the computed centroid (x_c, y_c) of tracked human window and the actual centroid (x_A, y_A) is followed:

$$K_c = \sqrt{(x_A - x_c)^2 + (y_A - y_c)^2} \quad (16)$$

Figure 5 shows the Euclidean distance of the proposed method and other tracking algorithms. It is clear that the proposed method has the least Euclidean distance between the centroid of tracked bounding box and the actual centroid in comparison with other methods. The ground truth values of centroid of the object were shown by the x-axis.

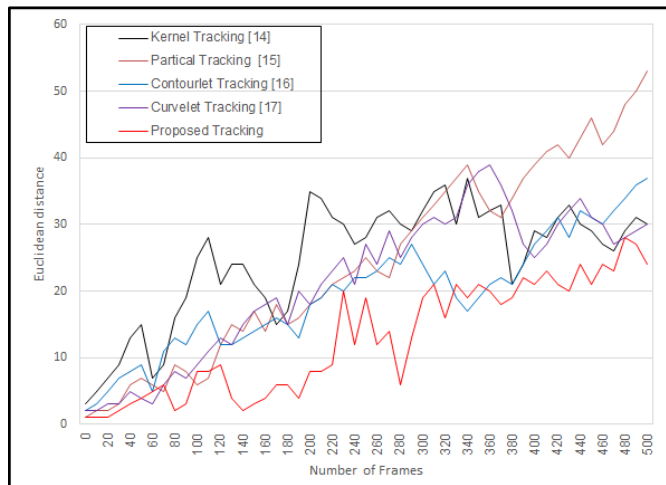


Fig. 5. The Euclidean distance of the proposed method and other tracking algorithms

V. CONCLUSION

Object tracking is an important feature of intelligent vision systems. Tracking has many applications in intelligent surveillance systems such as in traffic system: counting the vehicles, measuring vehicle's velocity, and then it can detect faults about the velocity of vehicles automatically, etc. These tasks are hard because they depend on the context and environment. The results of the monitoring objects tremendously impact the analysis phase and identification of behavior. This paper proposes a new method to track the human based on their contour. The proposed method uses the amplitude of zernike moment on NSCT to track object depending on context awareness. The algorithm has also been processed which successfully such cases as the new object detection, object detection obscured after they reappeared, detecting and tracking objects successfully intertwined and then separated again. In the future work, the author should compare equivalent geometry contours of the object shape. This equivalent does not depend on the direction of motion, or the location of the object. So the objects are still tracked properly in the above case.

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Metrics for Event Driven Software

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Abstract—The evaluation of Graphical User Interface has significant role to improve its quality. Very few metrics exist for the evaluation of Graphical User Interface. The purpose of metrics is to obtain better measurements in terms of risk management, reliability forecast, project scheduling, and cost repression. In this paper structural complexity metrics is proposed for the evaluation of Graphical User Interface. Structural complexity of Graphical User Interface is considered as an indicator of complexity. The goal of identifying structural complexity is to measure the GUI testability. In this testability evaluation the process of measuring the complexity of the user interface from testing perspective is proposed. For the GUI evaluation and calculating structural complexity an assessment process is designed which is based on types of events. A fuzzy model is developed to evaluate the structural complexity of GUI. This model takes five types of events as input and return structural complexity of GUI as output. Further a relationship is established between structural complexity and testability of event driven software. Proposed model is evaluated with four different applications. It is evident from the results that higher the complexities lower the testability of application.

Keywords—Graphical User Interface; Structural Complexity; Testability; Fuzzy model

I. INTRODUCTION

Graphical User interfaces have special characteristics that differentiate them from the rest of the software code. These applications have many challenges due to its event driven nature and infinite input domain. This event driven nature presents a challenge to testing because there are a large number of possible event sequences that users can invoke through a user interface. It is important to assess the quality of software. Software testability is one of the quality metric for software applications and ISO has defined software testability as a functionality and it defines functionality as “the collection of characteristics of software that bear on the effort required to authenticate the software produced”.

The testability of software is determined by factors such as:

- Controllability
- Observability
- Built in Test Capability
- Understandability
- Complexity

Complexity is one of the important factors to assess the testability of software. It is concluded from the literature

survey that typical software metrics for complexity do not identify complex GUI. Various structural complexity metrics are reported in literature survey. Most of the papers defined structural complexity in terms of visual objects, size, distribution and position and tree structure they are as follows:

- 1) Number of controls in an interface [1]
- 2) The longest sequence of different controls that is defined to perform a specific task. In terms of the GUI XML tree, it is depth of the tree.
- 3) Maximum number of choices for a user at any moment while using that interface.
- 4) Time required performing certain events in a GUI.
- 5) Tree structure metric that defines complex GUI when majority of its controls is toward the top and less complex GUI when most of the control is at bottom [2].
- 6) Tree path count calculates the number of leaf nodes in a tree [3].
- 7) Tree depth is calculated by the total number of choices in the tree is divided by the tree depth [3].

Users interact with a GUI by performing events on some widgets, like a button click, menu open, and dragging an icon. All interaction of GUI and a user is with the help of these events [4][6]. As defined in above metrics all choices presented to the user will be in the form of events. Maximum tree depth will be longest sequences of events. Therefore a new metric can be defined in terms of all types of events a GUI is consisting. It was shown from the Alsmadi I. (2011) research that LOC metric is irrelevant for GUI because it cannot be identified that how much code is GUI oriented. GUI events are classified on the basis of their response to the system on selection. Their classification is as follows [11].:-

Restricted-focus events,

Restricted-focus events open a modal window of GUI. For example, “Set Language” is a restricted focus event.

Unrestricted-focus events

Unrestricted-focus events open modeless windows. For example, Replace in MS WordPad is an unrestricted focus event.

Termination events

Termination events are used to close modal windows. Examples of Termination events include Ok, Exit and Cancel. There are other types of events that GUI contains, and they do not open or close windows but make other GUI events available. Menus that contain several events are open by these events.

Menu-open

Menus can be open by using Menu-open events. They expand the menus so that the set of GUI events would be available to the user. Menu-open events need not interact with the underlying software. Unrestricted-focus events open a window that has to be explicitly terminated, whereas menu open events has no such restriction. The most common example of menu-open events is generated by menus that open pull-down menus. Common example includes File, Edit, and Format. All other remaining events in the GUI are used to interact with the underlying software [8].

System-interaction

System-interaction events interact with the underlying software to perform some action; common examples include the Copy event used for copying objects to the clipboard [13]. This classification of events is used to compute structural complexity of GUI based software. To compute the structural complexity an assessment process is defined. An overview of Assessment Process is presented in the next section.

II. STRUCTURAL COMPLEXITY ASSESSMENT PROCESS

Based on types of events an assessment process for structural complexity is designed. This process is outlined next:

1) *Input to the process is GUI application. The Application under Test (AUT) is identified which essentially means identification of the locations of source files and any library modules needed to compile/build the AUT. This is known as baseline AUT.*

2) *To extract GUI components GUITAR (GUI Testing framework) is used, which generates GUI Structure (using a module called the GUI ripper) by automatically traversing all the windows of the GUI.*

3) *Event calculator computes count of each type of events. It is a kind of parser which takes GUI structure as input and return count of type of event.*

4) *Each event is assigned a weight value.*

5) *Normalization of data: Softmax scaling method is used to map all values between 0 and 1. Softmax scaling is based on the logistic function given in equ. (1):*

$$y = 1 / (1 + e^{-x}) \quad (1.1)$$

Where y is the normalized value and x is the original value.

The logistic function transforms the original range of $[-\infty, \infty]$ to $[0, 1]$ and also has a linear part on the transform. The values of the variables must be modified before using the logistic function in order to get a desired response.

This is achieved by using the following transform

$$x' = (x - \bar{x}) / (\lambda(\sigma / 2\pi)) \quad (1.2)$$

where \bar{x} is the mean of x , σ is the standard deviation, and λ is the size of the desired linear response.

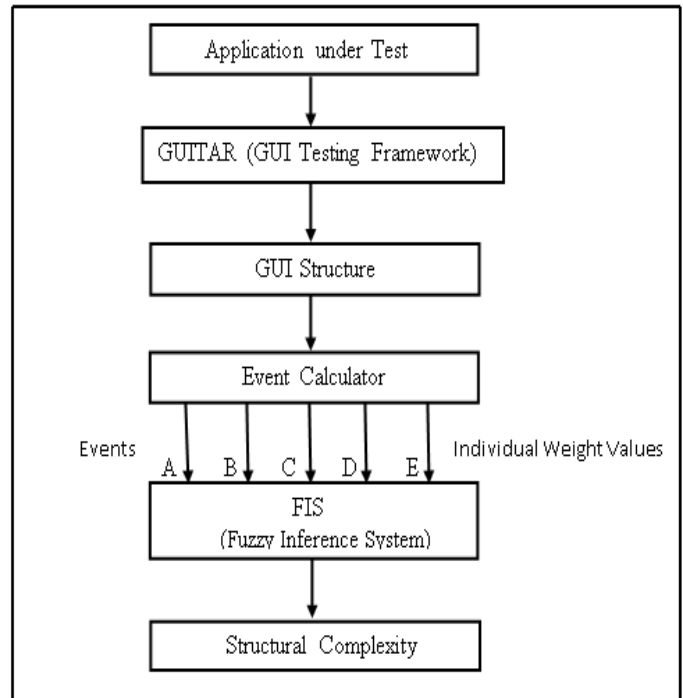


Fig. 1. Structural Complexity Assessment Process

Once the number of events is calculated their weight value is multiplied to each count.

6) *Fuzzy Inference System (FIS) is the process of formulating the mapping from an input space to output space [13]. A fuzzy model is proposed with five inputs, namely Unrestricted Focus Events, Menu Open Events, Termination Events, System Interaction Events and Restricted Focus Events. Figure 1.2 shows the fuzzy model. The proposed model consists of five inputs and provides a crisp value of structural complexity using Rule Base. In this model Mamdani's fuzzy inference method is used as shown in figure 3.*

After the fuzzification process, there is a fuzzy set for each output variable that needs defuzzification. The input for the defuzzification process is a fuzzy set (the aggregate output fuzzy set) and the output is singleton number. Further centroid method is used for defuzzification.

Events are classified in five types and further they are divided into three states (linguistic variables) i.e. low, medium and high.

The input variable Unrestricted Focus Events has been divided into three levels i.e. Low, medium and high.

Similarly other four inputs are divided into three states i.e. low, medium and high. The output variable complexity is also classified as low, medium and high.

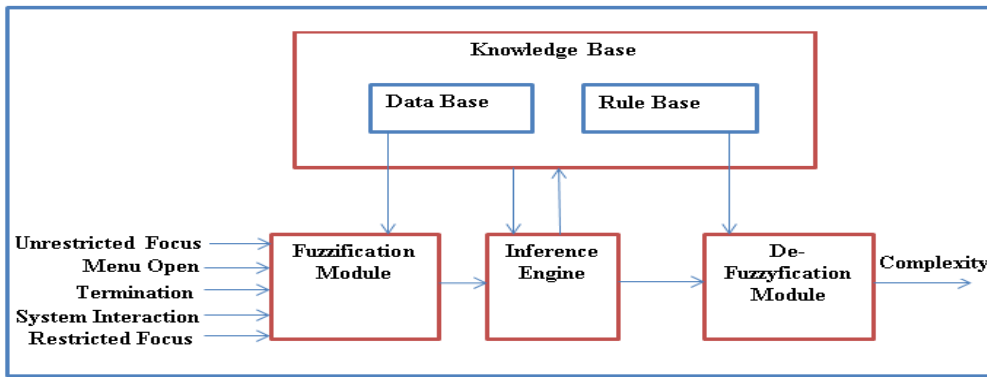


Fig. 2. Fuzzy model for Structural Complexity Assessment

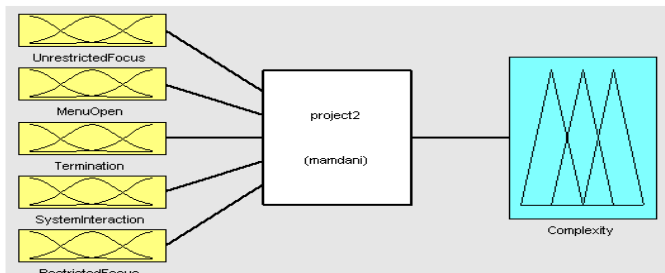


Fig. 3. Fuzzy Inference System: Complexity Assessment Model

A. Rule base and evaluation process

When input data is fuzzified, processing is carried out in fuzzy domain. The model integrates the effects of multiple factors Unrestricted Focus Events, Menu Open Events, Termination Events, System Interaction Events and Restricted Focus Events into a single measurable parameter that will define the structural complexity of test case, based on the following knowledge/rule base. The rule base can further be advanced by creating more ranges (fuzzy sets) for the input variables. All inputs and outputs are fuzzified as shown in figure 1.4. All possible combinations of inputs were considered that will create 3^5 i.e. 243 sets. The structural complexity for all 243 combinations is classified as low, medium and high by expert judgment. This indicates to formulation of 243 rules for the fuzzy model and some of the rules are presented below:

- 1) If value assigned for Unrestricted Focus Events is high, Menu Open Events is high, Termination Events is high, System Interaction Events is high and Restricted Focus Events is high then structural complexity is high.
- 2) If value assigned for Unrestricted Focus Events is high, Menu Open Events is high, Termination Events is high, System Interaction Events is high and Restricted Focus Events is medium then complexity is high.
- 3) If value assigned for Unrestricted Focus Events is high, Menu Open Events is high, Termination Events is high, System Interaction Events is high and Restricted Focus Events is low then complexity is high.

? If value assigned for Unrestricted Focus Events high, Menu Open Events is medium, Termination Events is high, System Interaction Events is high and Restricted Focus Events is high then complexity is high.

.....

243. If value assigned for Unrestricted Focus Events very low, Menu Open Events is low, Termination Events is very low, System Interaction Events is low and Restricted Focus Events is low then complexity is low.

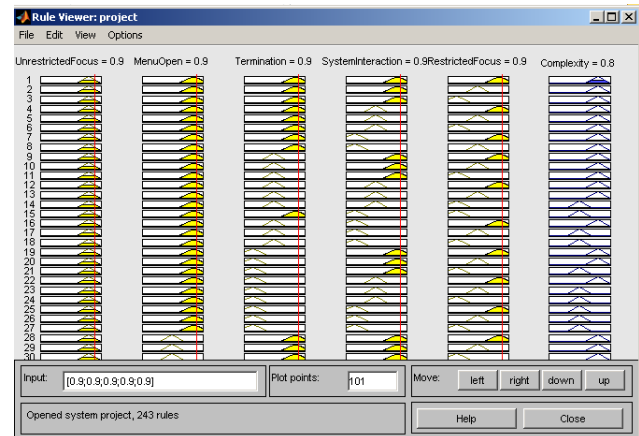


Fig. 4. Rule Viewer for the Complexity Assessment Model

All 243 rules are inserted and rule base is created. A rule is fired based on the particular set of inputs. In this model Mamdani style inference has been used.

The output of GUI complexity has been observed using rule viewer for particular set of inputs using MATLAB fuzzy Tool Box as shown in figure 4.

Defuzzification

After getting the fuzzified output as specified in previous section, these values are defuzzified to get the crisp value of the output variable priority. Transformation of the output from fuzzy domain to crisp domain is called defuzzification. In this model defuzzification is done using centre of gravity (COG) method. The final crisp value for COG is 0.8. Final value of complexity assessment for given input is 0.8, which lies in a complex GUI category.

For example if following values are considered as inputs to the model: Unrestricted Focus Events =0.9, Menu Open Events=0.9, Termination Events=0.9, System Interaction Events=0.9 and Restricted Focus Events=0.9.

III. EVALUATION OF FUZZY BASED COMPLEXITY ASSESSMENT METHOD

For the evaluation of fuzzy based test case prioritization method software artifacts are taken from Event Driven Software Lab (Dept. of Computer Science, University of Maryland, USA) established in 2005. These applications are part of an opensource office suite which has been considered in many research. This application suite is known as TerpOffice3 and includes TerpWord (a small word-processor), TerpSpreadSheet (a spreadsheet application), TerpPaint (an image editing/ manipulation program) and TerpPresent (a presentation tool). These applications are implemented using Java. These applications are fairly large with complex GUIs nearly as the size of MS WordPad. These applications do not have any complex underlying “business logic.” This property of applications makes them perfect subject applications for GUI research.

TABLE I. EVENTS AND NORMALIZED VALUES FOR TERPPAINT 3.0

Event type	TerpPaint 3.0	Weighted Value	Value Transformation	Logistic Normalization
System Interaction	589	58.9	6.0607	0.9977
Termination	0	0	-8.8630	0.0001
Non Restricted Focus	1	10	-6.3293	0.0018
Restricted Focus	11	55	5.0725	0.9938
Menu Open	51	51	4.0590	0.9830
Mean	-	34.98	-	-
Standard deviation	-	24.80809545	-	-

For all these application value of termination event is zero. So this value is not considered. Count of all events is coming in different range so different weight value is assigned to make these values come in same range. Softmax scaling is used to map all values in the range of 0 to 1. For all applications these values are calculated and they are shown in the table 1 to 4 for all applications. Table 1 shows events and normalized values for TerpPaint 3.0.

Table 2 shows event and normalized value for TerpPaint 3.0. For this application number of menu open events is very high as compared to other applications.

Table 3 represents Events and Normalized values for Terp SpreadSheet 1.0. In this application total numbers of events are small as compared to other application.

TABLE II. EVENTS AND NORMALIZED VALUES FOR TERPPRESENT 3.0

Event type	Terp Present 3.0	Weighted Value	Value Transformation	Logistic Normalization
SYSTEM INTERACTION	682	68.2	3.1606	0.9593
TERMINATION	0	0	-6.7453	0.0012
NON RESTRICTED FOCUS	0	0	-6.7453	0.0012
RESTRICTED FOCUS	10	50	0.5171	0.6265
MENU OPEN	114	114	9.8130	0.9999
Mean	-	46.44	-	-
Standard deviation	-	43.27556354	-	-

TABLE III. EVENTS AND NORMALIZED VALUES FOR TERPSPREADSHEET 3.0

Event type	TerpSpread Sheet 3.0	Weighted Value	Value Transformation	Logistic Normalization
SYSTEM INTERACTION	401	40.1	5.2760	0.9949
TERMINATION	0	0	-7.4079	0.0006
NON RESTRICTED FOCUS	0	0	-7.4079	0.0006
RESTRICTED FOCUS	6	30	2.0813	0.8891
MENU OPEN	47	47	7.4585	0.9994
Mean	-	23.42	-	-
Standard deviation	-	19.87223188	-	-

Table 4 shows events and normalized values for TerpWord 3.0. this application is fairly large application in terms of number of events as compared to other applications.

TABLE IV. EVENTS AND NORMALIZED VALUES FOR TERPWORD 3.0

Event type	TerpWord 3.0	Weighted Value	Value Transformation	Logistic Normalization
SYSTEM INTERACTION	1628	162.8	9.7407	0.9999
TERMINATION	0	0	-5.6195	0.0036
NON RESTRICTED FOCUS	1	10	-4.6760	0.0092
RESTRICTED FOCUS	23	115	5.2308	0.9947
MENU OPEN	10	10	-4.6760	0.0092
Mean	-	59.56	-	-
Standard deviation	-	66.62112578	-	-

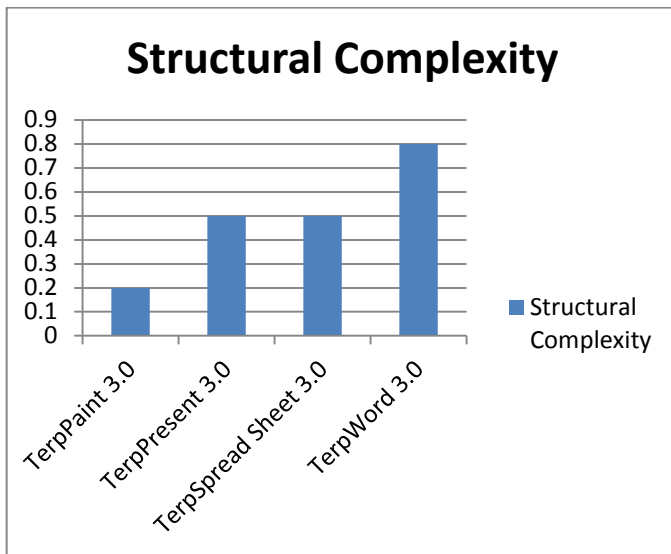


Fig. 5. Structural complexities of Applications under test

All normalized values shown in table 1 to table 4 are given as input to fuzzy model and their structural complexity is calculated and shown in figure 5.

Further fault seeding has been done on all applications and 200 faults are injected in each applications and different version of applications are created and tested with GUITAR[5].

Number of faults identified by each application is shown in figure 6.

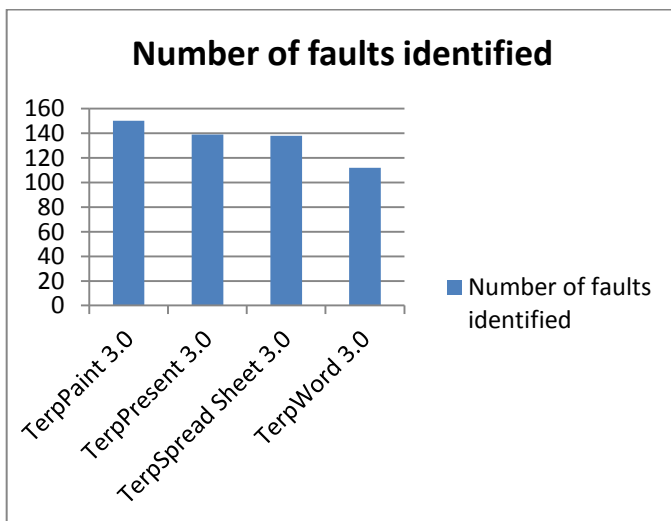


Fig. 6. Number of faults identified by each application

IV. CONCLUSION

Number of identified faults represents testability of software, where TerpPaint shows highest Testability and its structural complexity is lowest. Testability of TerpWord is

lowest and its complexity is highest. Structural complexity of TerpPresent and TerpSpreadsheet is same and their fault revealing capability is almost same. Hence it is evident from the results that higher the complexity lower the testability of application.

For evaluation of GUI structural complexity metric is defined which is based on types of events. A fuzzy model is developed for GUI evaluation. Its results are evaluated with 4 different applications. A relationship is further established with testability with this metric.

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A Novel Adaptive Grey Verhulst Model for Network Security Situation Prediction

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Abstract—Recently, researchers have shown an increased interest in predicting the situation of incoming security situation for organization's network. Many prediction models have been produced for this purpose, but many of these models have various limitations in practical applications. In addition, literature shows that far too little attention has been paid in utilizing the grey Verhulst model predicting network security situation although it has demonstrated satisfactory results in other fields. By considering the nature of intrusion attacks and shortcomings of traditional grey Verhulst model, this paper puts forward an adaptive grey Verhulst model with adjustable generation sequence to improve the prediction accuracy. The proposed model employs the combination methods of Trapezoidal rule and Simpson's 1/3rd rule to obtain the background value in grey differential equation which will directly influence the forecast result. In order to verify the performance of the proposed model, benchmarked datasets, DARPA 1999 and 2000 have been used to highlight the efficacy of the proposed model. The results show that the proposed adaptive grey Verhulst surpassed GM(1,1) and traditional grey Verhulst in forecasting incoming security situation in a network.

Keywords—Grey Theory; Network Security Situation Prediction; Adaptive Grey Verhulst Model; Adjustable Generation Sequence; Prediction Accuracy

I. INTRODUCTION

Internet has become an impeccable necessity in our life providing services such as information sharing, communication, social interaction and etc. The acceleration of countries modernization and proliferation of mobile devices utilization has boost up the Internet users to reach 3.17 billion in 2015 [1]. The growth of Internet is further driven by new technologies such as cloud computing and Internet of Things (IoT). However, the immense popularity of the Internet and prevalent use of online services has made Internet a breeding ground for malware and cyber criminals. New security challenges are emerging while people are enjoying to sharing their resources borderlessly. In 2014, Symantec has encountered a 23% and 40% increase in data breaches and phishing attacks respectively compared to previous year [2]. This alarming situation brings serious challenges to network security worldwide.

Prevention is better than detection and recovery. Due to the rising number of the threats, network security communities nowadays crave to know the incoming security situation in their network before any precaution taken. Unfortunately, countering attacks in an Intrusion Prevention System (IPS)

with a complete list of responses is insufficient. Surprisingly, in a study done by University of South Wales in 2013 on nine big-brand IPS systems, they found that seven out of them failed to detect and prevent up to 49% of attacks that target vulnerabilities especially in web-based application [3]. Therefore, predicting the incoming security situation in an entire network is desired to facilitate IPS to be more intelligent in in aspect of preventing the problem from growing and in returning the system to a healthy mode. Coincidentally, security situation prediction capability was considered as one of the main components in situation awareness when the concept has been introduced by Endsley [4] to the world. The idea then has been first adapted in the cyberspace by Tim Bass [5] with 3-hierarchical phases network security situation awareness (NSSA) which consists of event detection, current security situation assessment and future security situation prediction.

Recently, the governments, enterprises and other stakeholders started to adapt the concept of NSSA and seek some appropriate strategies especially in predicting incoming security situation in their network before any incident occurred. For instance, in Germany, the National Cyber Response Centre is responsible to alert the crisis management staff whenever the cyber security situation reaches the level of an imminent or already occurred crisis [6]. Meanwhile, in Malaysia, the National Cyber Security Policy addressed that there is a need to develop effective cyber security incident reporting mechanisms which capable of disseminating vulnerability advisories and threat warning in a timely manner in order to strengthen the National Computer Emergency Response Teams (CERTs) in monitoring the situation of critical national information infrastructure [7]. From the efforts of aforementioned countries in their strategic planning, it obviously brings us a significant motion that future network security situation prediction is very much in demand at the top level of cyber security strategic plan.

The rest of this paper is structured as follows: the authors first discuss some limitation of existing prediction models. Then, the author present a novel adaptive grey Verhulst model with the approach of calculating its adjustable generation sequence in the following section. Next, the authors demonstrate the grey prediction models with benchmarked datasets, The Defense Advanced Research Project Agency (DARPA) 1999 and 2000 (LLS DDOS1.0 and LLS DDOS 2.0.2). To verify the performance of the proposed model, the authors compare the accuracy of prediction result of our model

with traditional GM(1,1) and grey Verhulst models from the aspects of their Mean Absolute Percentage Error (MAPE) and Root Mean Square Deviation (RMSD). Finally, the authors summarize our work with a conclusion.

II. LIMITATION OF EXISTING PREDICTION MODELS

A considerable amount of work has been published on designing network security situation prediction models. These studies can be categorized into three groups, i.e. Machine Learning, Markov Model and Grey Theory [8]. Prediction based on machine learning such as neural network and support vector system is commonly used in situation prediction due to its high convergence rate and strong fault tolerance capacity. But it requires a large amount of training data to gain the appropriate parameters and establish self-learning neurons. Furthermore, the method is unsuitable for small-scaled data as less input information will slower the convergence. Markov model, on the other hand, is also to be used to perform the prediction in various time series such as series of network situation. Nevertheless, the model is complex and difficult to build due to its difficulties in making assumption on all possible states and transitions especially in a network which is highly heterogeneous in nature. Since data pertaining to network situation may be inconsistent and incomplete, grey theory especially First-order One-variable grey model (GM(1,1)) has been widely used to provide better prediction in short-term forecasting with small sample data without any training required. Regrettably, the method is only limited to linear time series and it is not suitable for non-stationary random sequence. Apparently, the generation sequence with mean is only limited to small time interval and it depresses the model precision with delay error. In fact, Grey Verhulst, a type of small sample predicting model in Grey Theory which able to forecast the situation with single peak of data sequence. However, the model failed to apply some related influencing factors which will degrade its performance [9]. Observing the chronology of an intrusion attack as illustrated in Figure 1, the authors argue that an adaptive Grey Verhulst model with its adjustable generation sequence is best suited to predict the incoming network security situation which behaves as a non-linear time series.

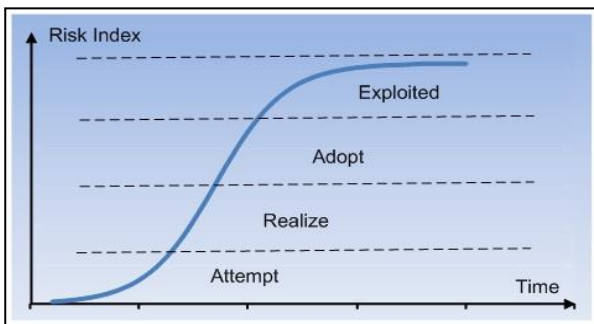


Fig. 1. The chronology of an intrusion attack

III. PROPOSED ADAPTIVE GREY VERHULST MODEL

GM(1,1) and Grey Verhulst are theories to deal with indeterminate and incomplete system with their superiority in small sample. Nonetheless, they have similar problem in overshoots which caused by the non-monotonic time series

data [10]. In addition, the generated sequence also make the prediction generate the advance or delay error which will depress the model precision [11]. Hence, this paper attempts to show that adaptive determination of grey parameters in grey Verhulst model is able to guarantee the precision. The adjustable generation sequence in this adaptive grey Verhulst model is not only suitable to forecast a stochastic time series such as incoming network security situation but also to handle multiple-peak situation variation which is inherent in network behavior [8].

In order to predict the incoming network security situation, a sequence of current and historical assessment of network security situation is used as input to the model. Figure 2 depicts the process flow of adaptive grey Verhulst model.

First, a sequence of network security situation assessment, $X^{(0)}$ is channeled into the model and a new sequence of accumulated data is built by applying the 1-Accumulated Generating Operation (1-AGO).

$$X^{(0)} = \{x^{(0)}(1), x^{(0)}(2), \dots, x^{(0)}(n)\}$$

$$X^{(1)} = \{x^{(1)}(1), x^{(1)}(2), \dots, x^{(1)}(n)\}$$

where

$$x^{(1)}(t) = \sum_{i=1}^t x^{(0)}(i), \quad X^{(0)}(i) \geq 0, \quad t = 1, 2, \dots, n.$$

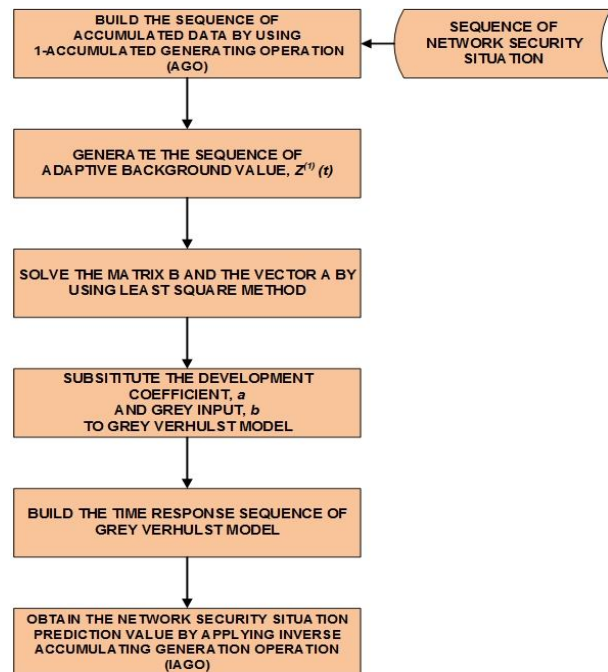


Fig. 2. Process flow of adaptive grey Verhulst model

Next, a sequence of adaptive background value, $Z^{(1)}(t)$, is generated by considering the consecutive (before and after) neighbors of $x^{(1)}$. The detail of calculating $Z^{(1)}(t)$ is explained in the following section.

$$Z^{(1)}(t) = \{z^{(1)}(1), z^{(1)}(2), \dots, z^{(1)}(n)\}$$

where

$$z^{(1)}(t) = x^{(1)}(t-1) + \frac{1}{6}x^{(0)}(t-1) - \frac{1}{6}x^{(0)}(t-2) + \frac{1}{2}x^{(0)}(t),$$

and $t = 3, 4, \dots, n$.

The value of $z^{(1)}(t)$ is substituted into Grey Verhulst model below.

$$x^{(0)}(t) + az^{(1)}(t) = b[z^{(1)}(t)]^2$$

where a is development coefficient which its size reflects the growth rate of the sequence $X^{(0)}$ and b is the role of vector which is grey input in Grey Verhulst model. After that, the equations are rearranged into matrix form $Y = B\hat{a}$ with

$$A = \begin{bmatrix} a \\ b \end{bmatrix} = (B^T B)^{-1} B^T Y$$

the parameter matrix and the matrixes of B and Y as below.

$$B = \begin{bmatrix} -z^{(1)}(2) & (z^{(1)}(2))^2 \\ -z^{(1)}(3) & (z^{(1)}(3))^2 \\ \vdots & \vdots \\ -z^{(1)}(n) & (z^{(1)}(n))^2 \end{bmatrix} \quad Y = \begin{bmatrix} x^{(0)}(2) \\ x^{(0)}(3) \\ \vdots \\ x^{(0)}(n) \end{bmatrix}$$

In order to find the value of a and b , the matrix B and vector A have been solved by using matrix method, $A = (B^T B)^{-1} B^T Y$. The value of a and b can be obtained through the formulas below:

$$a = \frac{DH - GE}{FG - D^2} \quad b = \frac{FH - DE}{FG - D^2}$$

where

$$D = \sum_{t=3}^n [z^{(1)}(t)]^3, \quad E = \sum_{t=3}^n [z^{(1)}(t)x^{(0)}(t)],$$

$$F = \sum_{t=3}^n [z^{(1)}(t)]^2, \quad G = \sum_{t=3}^n [z^{(1)}(t)]^4,$$

$$H = \sum_{t=3}^n [z^{(1)}(t)]^2 x^{(0)}(t)$$

With the value of a and b , the predicted time response sequence of Grey Verhulst model, $x_g^{(1)}(t+1)$ is calculated by substituting them into the solution of Verhulst model.

$$x_g^{(1)}(t+1) = \frac{ax^{(0)}(1)}{bx^{(0)}(1) + (a - bx^{(0)}(1))e^{at}}$$

where $x^{(0)}(1) = x^{(1)}(1)$. Finally, to obtain the prediction

value of next network security situation, $x_g^{(0)}(t+1)$, the Inverse Accumulating Generation Operation (IAGO) has been

applied. As $x^{(1)}(t+1) = x^{(1)}(t) + x^{(0)}(t+1)$, the

$x_g^{(0)}(t+1)$ can be determined by using the formula below.

$$x_g^{(0)}(t+1) = x_g^{(1)}(t+1) - x_g^{(1)}(t)$$

and

$$x_g^{(0)}(1) = x^{(1)}(1) = x^{(0)}(1)$$

where $t = 2, 3, \dots, n$.

IV. ADAPTIVE BACKGROUND VALUE GENERATION

Background value, $z^{(1)}(t)$ is a crucial factor that influences the adoption of grey theories and their forecasting result. The value of developing coefficient, a and the precision of the model will be affected by different background values [12].

In traditional grey Verhulst, the grey differential equation can be written as

$$x^{(0)}(t) + az^{(1)}(t) = b(z^{(1)}(t))^2 \quad (1)$$

where

$$z^{(1)}(t) = \alpha x^{(1)}(t) + (1 - \alpha)x^{(1)}(t-1)$$

Observed from the differential equation, background value has direct influences on the precision of the Grey Verhulst. Its value is determined by α which range $0 \leq \alpha \leq 1$. On the basic of traditional grey theories, α is always set as 0.5 to equalize the importance of each data [10, 13, 14]. In this context, the ignorance of data characteristic has produced more prediction errors [12, 15]. Thus, to improve the performance of grey theories especially in grey Verhulst, the error term resulted from the background value generation have to be eliminated. In other words, finding a suitable background value for the model is an essential subject to improve the prediction accuracy.

Based on [16], the most suitable background value should be located in between $x^{(1)}(t-1)$ and $x^{(1)}(t)$ as illustrated in figure 3. Due to the developing coefficient will direct affect the background value, thus the newer data should be emphasized by assigning a larger value of α [12]. In fact, setting the value of α is a process to search the optimal solutions within the value space. The time series dataset should be regarded as several different populations [17]. Hence, the value of α should be adaptable at each timescale with different adjustable background values as depicted in figure 4.

The possible error which might degrade the precision of grey Verhulst can be identified prior to its elimination. In grey Verhulst, the whitening equation of grey Verhulst model is written as

$$\frac{dx^{(1)}(t)}{dt} + ax^{(1)}(t) = b[x^{(1)}(t)]^2 \quad (2)$$

By integrating both side of equation (2),

$$\int_{t-1}^t \frac{dx^{(1)}(t)}{dt} dt + a \int_{t-1}^t x^{(1)}(t) dt = b \int_{t-1}^t [x^{(1)}(t)]^2 dt \quad (3)$$

where

$$\int_{t-1}^t \frac{dx^{(1)}(t)}{dt} dt = x^{(1)}(t) - x^{(1)}(t-1) = x^{(0)}(t)$$

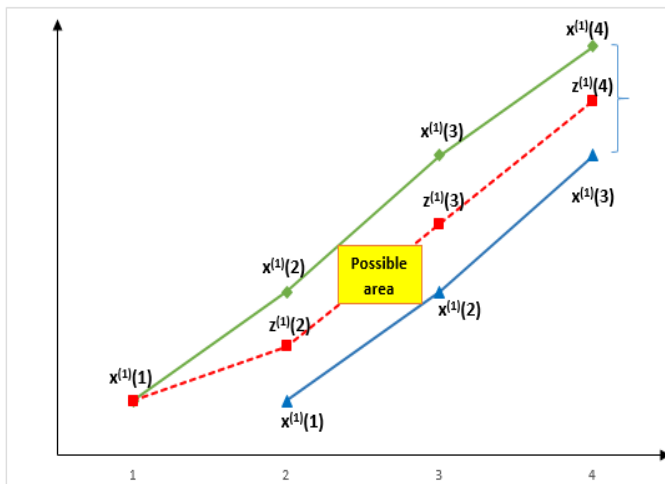


Fig. 3. Possible area of background values

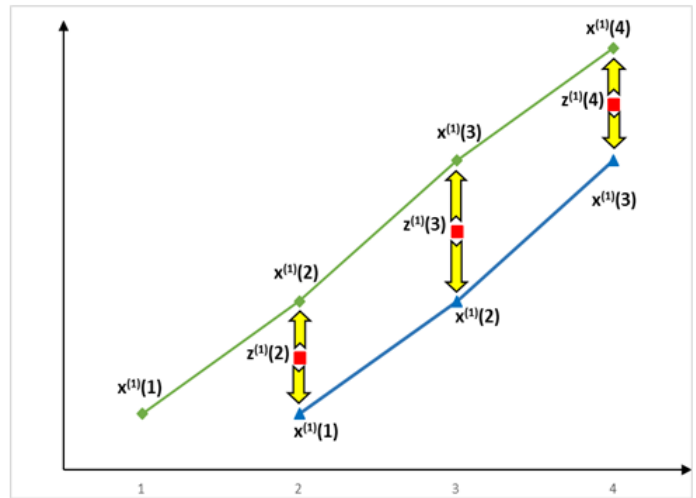


Fig. 4. Distribution of the adjustable background values

Thus, equation (3) can be written as

$$x^{(0)}(t) + a \int_{t-1}^t x^{(1)}(t) dt = b \int_{t-1}^t [x^{(1)}(t)]^2 dt \quad (4)$$

By comparing the equation (1) and (4), the background value can be determined as below.

$$z^{(1)}(t) = \int_{t-1}^t x^{(1)}(t) dt \quad (5)$$

From the equation (5), the error is exist if there has an inequality equation as follows.

$$\int_{t-1}^t x^{(1)}(t) dt \neq \alpha x^{(1)}(t) + (1-\alpha)x^{(1)}(t-1)$$

where

$$z^{(1)}(t) = \alpha x^{(1)}(t) + (1-\alpha)x^{(1)}(t-1)$$

Therefore, to eliminate the error, the background value must be equal to the integration of $x^{(1)}(t)$ from two consecutive time interval $t-1$ to t .

$$\int_{t-1}^t x^{(1)}(t) dt \quad (6)$$

Indeed, equation (6) represents an area under a graph function as presented in figure 5.

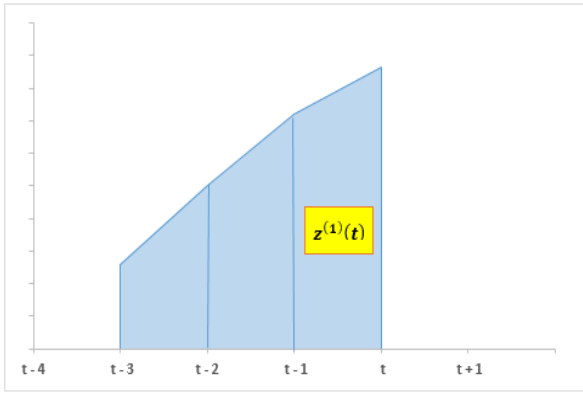


Fig. 5. Area under a graph function

Due to the curve function is unknown, in our proposed adaptive grey Verhulst model, the background value, $z^{(1)}(t)$ is calculated by the combination methods of Trapezoidal rule and Simpson's 1/3rd rule. These rules are used to determine the area under a graph without knowing its function. Trapezoidal rule is based on approximating the integrand by a first order polynomial and then integrating the polynomial in the interval of integration while Simpson's 1/3rd rule is an extension of Trapezoidal rule where the integrand is approximated by a second order polynomial. From figure 5, the area under the curve from time interval $t-3$ to t can be determined as below.

$$\int_{t-3}^t x^{(1)}(t) dt = \int_{t-3}^{t-1} x^{(1)}(t) dt + \int_{t-1}^t x^{(1)}(t) dt \quad (7)$$

Rearrange the equation (7),

$$\int_{t-1}^t x^{(1)}(t) dt = \int_{t-3}^t x^{(1)}(t) dt - \int_{t-3}^{t-1} x^{(1)}(t) dt \quad (8)$$

By applying the Trapezoidal rule and Simpson's 1/3rd rule, the equation (8) can be further simplified as follows.

$$\int_{t-1}^t x^{(1)}(t) dt = \frac{t-(t-3)}{2(3)} [x^{(1)}(t-3) + 2(x^{(1)}(t-2) + x^{(1)}(t-1) + x^{(1)}(t))] - \frac{(t-1)-(t-3)}{6} [x^{(1)}(t-3) + 4x^{(1)}(t-2) + x^{(1)}(t-1)]$$

Finally, the background value, $z^{(1)}(t)$ can be obtained through the equation below.

$$z^{(1)}(t) = x^{(1)}(t-1) + \frac{1}{6} x^{(0)}(t-1) - \frac{1}{6} x^{(0)}(t-2) + \frac{1}{2} x^{(0)}(t) \quad (9)$$

Where $t = 3, 4, \dots, n$.

V. CASE STUDY AND RESULTS

The DARPA/MIT Lincoln Lab evaluation datasets 1999 and 2000 have been published and widely used in evaluating the performance of prediction models [18-22] recently. In these datasets, there are various attacks found and there can be categorised into five main classes namely, Probe, Denial of Service (DoS), Remote to Local (R2L) and User to Remote (U2R) and the Data attacks [23].

In order to verify the performance of our proposed adaptive grey Verhulst in predicting network security situation, these three benchmarked datasets, DARPA 1999 and 2000 (LLS DDOS 1.0 and LLS DDOS 2.0.2) have been used in our model as well as traditional GM(1,1) and grey Verhulst models. These datasets were divided into several time-slots based on hours or minutes, and be evaluated by using entropy-based network security situation assessment approach in [24]. The values of situation assessment for each time slots have been used as input for the prediction models to forecast the next network security situation. Table 1, 2 and 3 present the prediction results from each dataset for the three models of grey theory aforementioned.

From the computational results aforementioned, Mean Absolute Percentage Error (MAPE) and Root Mean Square Deviation (RMSD) are used as evaluation metrics to determine the performance of prediction models in term of its accuracy. MAPE is a measure of accuracy of a method for constructing fitted time series values in statistics especially in trend estimation while RMSD is frequently used to measure the differences between the values predicted by a model and the values actually observed. The numerical results show that adaptive grey Verhulst model has attained average 93.3% of prediction accuracy while GM(1,1) and traditional grey Verhulst models has only achieved 87.3% and 92.0% respectively. Compared to both traditional GM(1,1) and grey Verhulst model, the lower MAPE and RMSD values produced have further prove the proposed prediction model is more reliable in forecasting incoming security situation in a network.

TABLE I. PREDICTION RESULT FOR DARPA 1999

Time /h	Real Value	GREY PREDICTION METHODS					
		Traditional GM(1,1)		Traditional Grey Verhulst		Adaptive Grey Verhulst	
		Predicted Value	RPE (%)	Predicted Value	RPE (%)	Predicted Value	RPE (%)
13	0.2206	0.2641	19.75	0.2467	11.86	0.2557	15.92
14	0.1871	0.2751	47.01	0.2183	16.70	0.2349	25.57
15	0.2056	0.2864	39.32	0.1853	9.89	0.2071	0.75
16	0.2151	0.2983	38.68	0.1517	29.49	0.1761	18.14
17	0.1806	0.3106	72.05	0.1206	33.23	0.1451	19.64
18	0.1706	0.3235	89.67	0.0936	45.12	0.1165	31.69
19	0.2056	0.3369	63.86	0.0714	65.29	0.0916	55.43
20	0.2160	0.3508	62.41	0.0537	75.16	0.0709	67.17
21	0.1956	0.3654	86.78	0.0399	79.59	0.0542	72.30
22	0.1835	0.3805	107.35	0.0295	83.94	0.0410	77.65
MAPE (%)		62.69		45.03		38.43	
RMSD		0.13		0.10		0.09	

TABLE II. PREDICTION RESULT FOR DARPA 2000 – LLS DDOS 1.0

Time /15 min	Real Value	GREY PREDICTION METHODS					
		Traditional GM(1,1)		Traditional Grey Verhulst		Adaptive Grey Verhulst	
		Predicted Value	RPE (%)	Predicted Value	RPE (%)	Predicted Value	RPE (%)
9	0.1848	0.2458	33.00	0.2281	23.42	0.2290	23.91
10	0.1961	0.2795	42.58	0.1926	1.77	0.1993	1.64
11	0.1415	0.3179	124.63	0.1423	0.59	0.1519	7.34
12	0.1773	0.3615	103.95	0.0952	46.29	0.1045	41.04
MAPE (%)		76.04		18.02		18.48	
RMSD		0.14		0.05		0.04	

TABLE III. PREDICTION RESULT FOR DARPA 2000 – LLS DDOS 2.0.2

Time /10 min	Real Value	GREY PREDICTION METHODS					
		Traditional GM(1,1)		Traditional Grey Verhulst		Adaptive Grey Verhulst	
		Predicted Value	RPE (%)	Predicted Value	RPE (%)	Predicted Value	RPE (%)
7	0.0705	0.1051	49.06	0.0971	37.72	0.1015	43.94
8	0.0541	0.1142	111.01	0.0569	5.09	0.0636	17.58
9	0.0651	0.1240	90.46	0.0273	58.13	0.0323	50.33
10	0.0751	0.1347	79.36	0.0118	84.26	0.0472	37.17
MAPE (%)		82.47		46.30		37.25	
RMSD		0.05		0.04		0.03	

VI. CONCLUSION

In conclusion, this paper presents a novel adaptive grey Verhulst prediction model with its adjustable generation sequence. The authors have shown that the prediction accuracy of the proposed adaptive grey Verhulst was much better compared to GM(1,1) and traditional grey Verhulst models due to its remarkable features in forecasting next security situation in the network. With the capability to handle multiple-peaks situation and able to provide higher accuracy of prediction to network administrator, the proposed adaptive grey Verhulst is very well-suited to predict incoming network security situation of an organization. In the future, the author propose to investigate the prediction error of the adaptive grey Verhulst and design a residual prediction algorithm to complement it in order to improve the predictive accuracy in forecasting incoming network security situation.

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Adaptive Neuro-Fuzzy Inference Systems for Modeling Greenhouse Climate

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Abstract—The objective of this work was to solve the problem of non linear time variant multi-input multi-output of greenhouse internal climate for tomato seedlings. Artificial intelligent approaches including neural networks and fuzzy inference have been used widely to model expert behavior. In this paper we proposed the Adaptive Neuro-Fuzzy Inference Systems (ANFIS) as methodology to synthesize a robust greenhouse climate model for prediction of air temperature, air humidity, CO2 concentration and internal radiation during seedlings growth. A set of ten input meteorological and control actuators parameters that have a major impact on the greenhouse climate was chosen to represent the growing process of tomato plants. In this contribution we discussed the construction of an ANFIS system that seeks to provide a linguistic model for the estimation of greenhouse climate from the meteorological data and control actuators during 48 days of seedlings growth embedded in the trained neural network and optimized using the back propagation and the least square algorithm with 500 iterations. The simulation results have shown the efficiency of the proposed model.

Keywords—Greenhouse climate; Modeling; ANFIS; Neuro-Fuzzy

I. INTRODUCTION

In recent decades, a considerable effort was devoted to develop adequate greenhouse climate and crop models, for driving simulation, control and managing [1-2]. The objective in a greenhouse climate control is to further improve the environmental conditions of plants growth in order to optimize the production process [3]. The dynamic behavior of the internal microclimate of the greenhouse is a combination of physical processes involving energy transfer (radiation and heat) and mass balance (water vapor fluxes and CO2 concentration) [4]. These processes depend on the outside environmental conditions, structure of the greenhouse, type and state of the crop and on the effect of the control actuators (typically ventilating and heating to modify inside temperature and humidity conditions, shading and artificial light to change internal radiation, CO2 injection to influence photosynthesis and fogging/cooling for humidity enrichment). The practical goal of this work is to model the greenhouse air temperature,

air humidity, CO2 concentration and internal radiation using Adaptive Neuro-Fuzzy Inference Systems from real data in order to predict the behavior inside the greenhouse. The main advantages of using automated climate control are energy conservation, better productivity, and reduced human intervention [5].

Greenhouses are considered as complex processes. In fact, they are nonlinear, multi-input multi-output (MIMO) systems which present time-varying behaviors, and they are subject to relevant disturbances depending generally on meteorological conditions. All these make it difficult to describe a greenhouse with analytic models and to control them with classical controllers [6-7].

Many conventional methods for controlling a greenhouse climate are not effective since they are based on either on-off control methods, or PID approaches. This results in a loss of energy, labor, and productivity [8]. To maintain a steady climate, a more complex control system must be used [5]. The necessities for climate control the energy consumption or maximize economic profit have instigated many researchers in this area. Today, there are various papers dealing with modeling, short term climate control, and long horizon control devised either to minimize. In [6], a model of a greenhouse using the energy balance has been presented. The proposed model is then used to carry out a simulation on the greenhouse climate (temperature and humidity) with optimal control for part of a day. In [9] the author has proposed a greenhouse model including the crop transpiration. They then made a comparison between optimal and predictive control on the considered greenhouse for part of a day. In [10] the authors have described the application of model predictive control (MPC) for temperature regulation in agricultural processes (a greenhouse). In [11], the authors have proposed the application of fuzzy logic to identify and control some multi-dimensional systems. They describe a method to reduce the complexity of a fuzzy controller and they show an application on a real system (a greenhouse). In [12], a recurrent neural network based on an Elman structure [13-14] is trained to emulate the direct dynamics of the greenhouse. In [15] the construction of fuzzy systems by fuzzy c-means for modeling

a greenhouse climate is described then the comparison with adaptive neuro-fuzzy inference system (ANFIS) and neural networks have presented. In [16] the authors have described the Greenhouse Design and Control using the adaptive neuro-fuzzy inference system.

Modern control techniques have been developed in various branches [17-18]. During the last two decades, considerable effort was devoted to develop adequate greenhouse climate and crop models, for simulation, control and management purposes [19-20]. A proper model for a greenhouse climate is an essential tool for its control [21-22]. The model can be designed in two ways. One method is based on the physical laws involved in the process and the other on the analysis of the input-output data of the process. In the first method, the thermodynamic properties of the greenhouse system are employed. However, the parameters of the equations are time variant and weather-dependent, so it is difficult to obtain accurate mathematical models of the greenhouse climate. The second approach is based on the theory of system identification [21]. Conventional methods based on system identification such as ARX approaches can not correctly model the nonlinear behavior of greenhouse climate. Intelligent methods seem to be the most proper choices for the modeling of this type of systems [3]. Because of the properties of universal approximation, they can model nonlinear systems with trained data by arbitrary fitness.

In contrast with a neural network identifier, a fuzzy identifier has some essential advantages which are described in the following. Due to its capability to handle both numerical data and linguistic information, it is feasible to apply fuzzy logic system for greenhouse climate modeling and then provide prediction for choosing optimal controlling decision. The growing process of tomato plants inside the greenhouse was modeled in this paper using the ANFIS system to predict the effect of meteorological variables and control actuators on air temperature, air humidity, CO₂ concentration and internal radiation inside the greenhouse. Specifically the relationship between the sensors signals and internal climate components is first captured via a neural network and is subsequently reflected in linguistic form with the help of a fuzzy logic based algorithm. It uses training examples as input and constructs the fuzzy if-then rules and the membership functions of the fuzzy sets involved in these rules as output. After training the estimator, its performance was tested under various internal climate conditions. Test data sets collected from a wide range of internal climate conditions (meteorological data and control actuators during 48 days of seedlings growth) were applied to the estimator for evaluating the magnitude of air temperature, air humidity, CO₂ concentration and internal radiation inside the greenhouse. The present paper describes simulation results of an ANFIS system that seeks to provide a linguistic model optimized by back-propagation and the least square algorithm for predicting the greenhouse climate.

II. EXPERIMENTAL SET-UP

The recordings were made during the period of February 20th to April 7th, 2015 on the site of BENOMOR, nursery of

tomatoes seedlings, Guelma (Algeria), in order to prepare them for the tomatoes season. The greenhouse in which all the experimental recordings were carried out is a plastic greenhouse multi chapels, multi inflatable wall, 3600 m³ of volume and 1000 m² of surface. Their major axis is parallel to the East-West direction. The roof and the side walls are polythene. This greenhouse with its climatic vectors of input-outputs is identified as a climate model represented in the "Fig. 1".

To carry out our work, we divided the 48 days of data file into three parts, the first part include the values of first 16 days and we used it as training data, the second part for the checking and the third for testing.

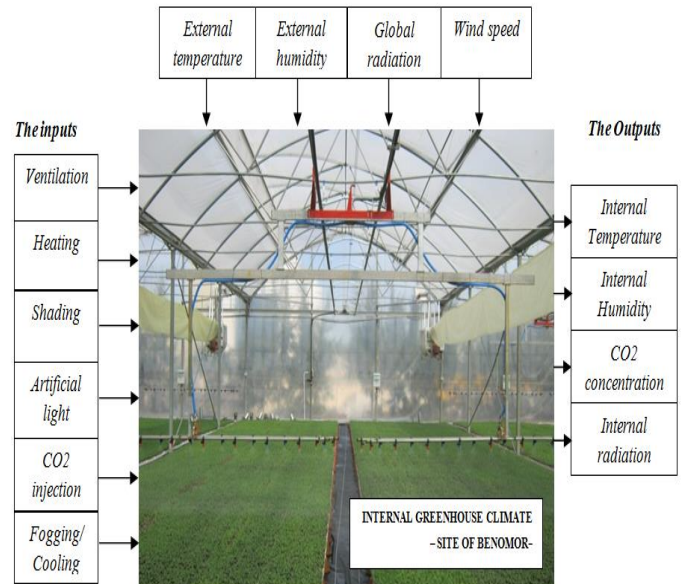


Fig. 1. Schematic of greenhouse climate

III. ANFIS PREDICTIVE ARCHITECTURE

Using a given input/output data set, the ANFIS method constructs a fuzzy inference system (FIS) whose membership function parameters are tuned (adjusted) using either a backpropagation algorithm alone, or in combination with a least squares type of method. This allows fuzzy systems to learn from the data they are modeling. FIS Structure is a network-type structure similar to that of a neural network, which maps inputs through input membership functions and associated parameters, and then through output membership functions and associated parameters to outputs [23].

In our case ANFIS is a four-layer neural network that simulates the working principle of a fuzzy inference system. The linguistic nodes in layers one and four represent the input and output linguistic variables, respectively. Nodes in layers two are term nodes acting as membership functions for input variables. Each neuron in the third layer represents one fuzzy rule, with input connections representing preconditions of the rule and the output connection representing consequences of the rules. Initially, all these layers are fully connected, representing all possible rules.

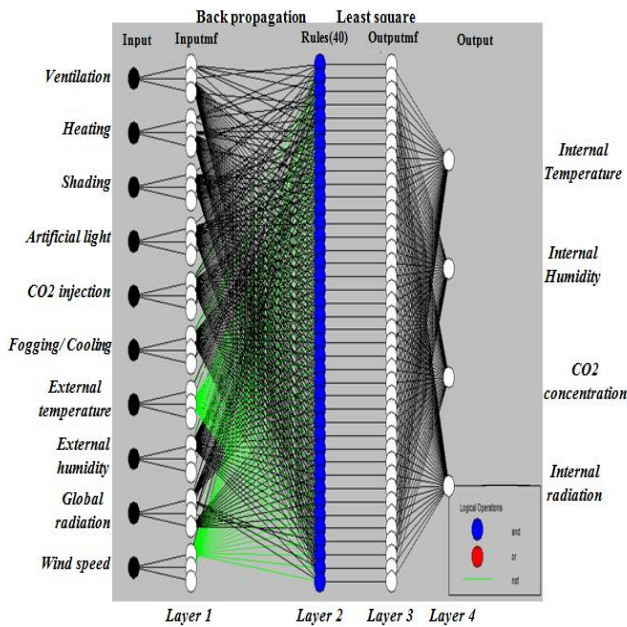


Fig. 2. ANFIS model structure of greenhouse climate

Ten feature variables, ventilation, heating, shading, artificial light, CO₂ injection, fogging/cooling, external temperature, external humidity, global radiation and wind speed, are selected as inputs of the ANFIS. Three trapezoidal membership functions are assigned to each linguistic variable. The suggested ANFIS model is shown in “Fig. 2”.

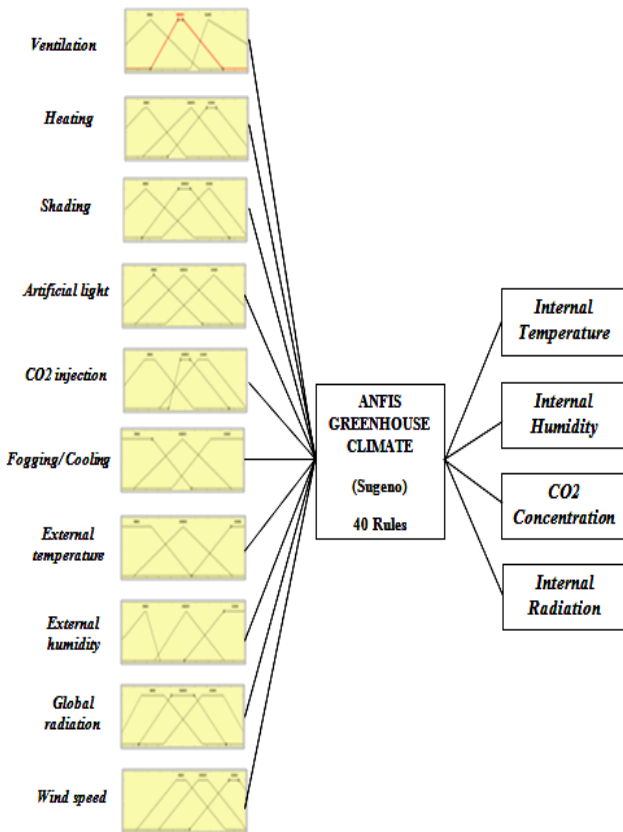


Fig. 3. Fuzzy rule architecture of the trapezoidal membership function

It shows the fuzzy rule architecture of ANFIS when the trapezoidal membership function is adopted. The architecture consists of 40 fuzzy rules. During preliminary experiments the proposed architecture proved to be sufficiently capable of extracting greenhouse climate model from the control actuators and meteorological data. “Fig. 4”, shows the flow chart for predicting the internal climate via ANFIS.

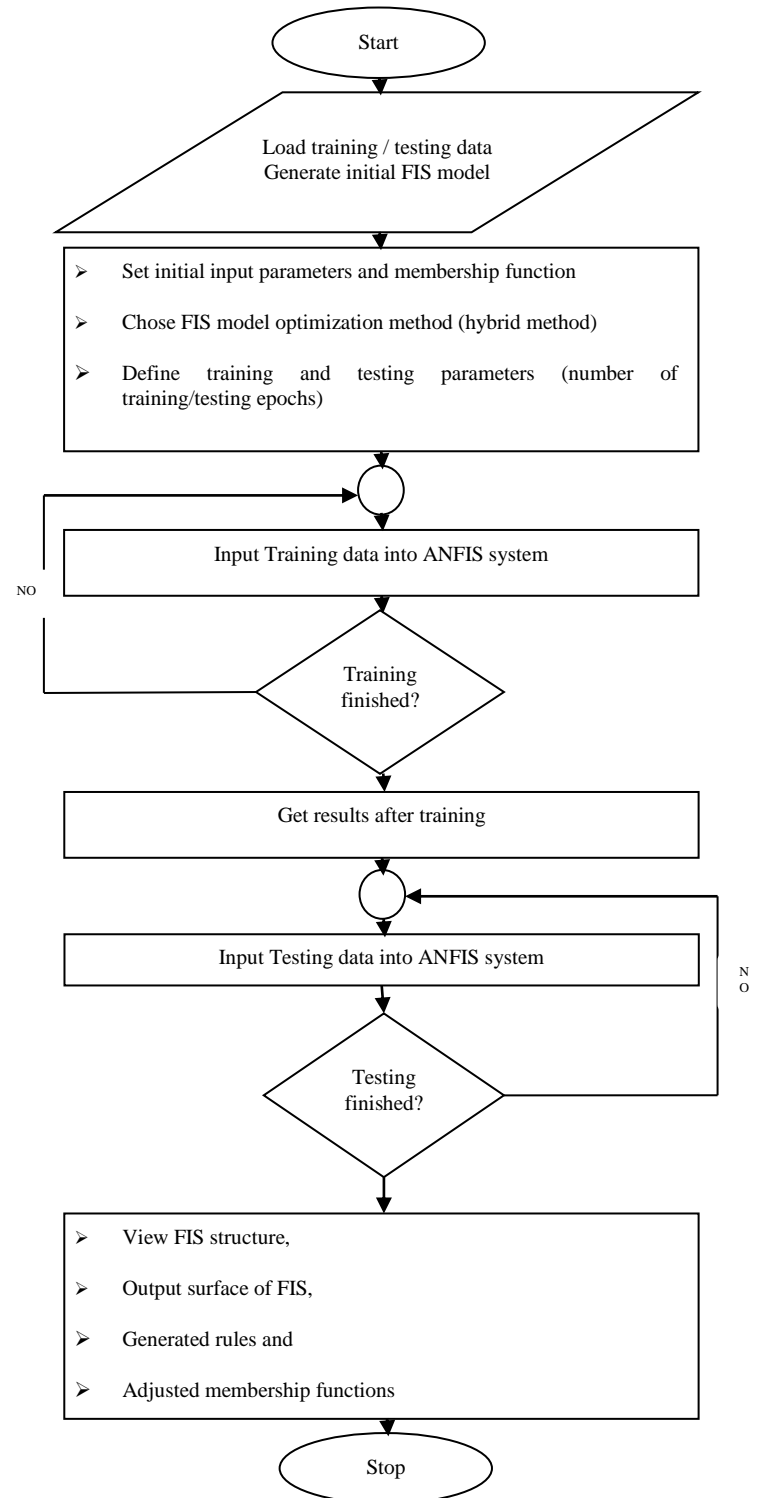


Fig. 4. Flowchart of internal climate prediction of ANFIS system

IV. ANFIS MODELING, TRAINING AND TESTING

ANFIS modeling process starts by obtaining a data set (input-output data) and dividing it into training, testing and checking data sets. Training data constitutes a set of input and output vectors. The data is normalized in order to make it suitable for the training process. This was done by mapping each term to a value between 00, 01 and 10 using the Min, moderate and Max method. This normalized data was utilized as the inputs (control actuators conditions and meteorological data) and outputs (internal climate) to train the ANFIS. In other words, two vectors are formed in order to train the ANFIS (see “Fig. 3”): Input vector = [ventilation, heating, shading, artificial light, CO₂ injection, fogging/cooling, external temperature, external humidity, global radiation and wind speed]. The output vector = [internal temperature, internal humidity, CO₂ concentration and internal radiation]. The ANFIS registers the input data only in the numerical form therefore the information about the control actuators, internal and external climate of the greenhouse must be transformed into numerical code.

The training data set is used to find the initial premise parameters for the trapezoidal membership functions by equally spacing each of the membership functions. A threshold value for the error between the actual and desired output is determined. The consequent parameters are found using the least-squares method.

Then an error for each data pair is found. If this error is larger than the threshold value, update the premise parameters using the gradient decent method as the following ($Q_{next} = Q_{nov} + \eta d$, where Q is a parameter that minimizes the error, η the learning rate, and d is a direction vector). The process is terminated when the error becomes less than the threshold value. Then the checking data set is used to compare the model with actual system. A lower threshold value is used if the model does not represent the system.

“Fig. 5”, shows the uniform falling of the value of testing error E_{Test} with the number of iterations during the testing process for the ANFIS configuration with traingular Mf and with gaussian Mf. The smallest error of testing (E_{Test}) is reached at iteration 145 (traingular Mf) and at iteration 107 for Gaussian Mf. It can be seen in the “Fig. 5” that error converges not to zero but to 12% and 2%. This is caused by the presence of some contradicting examples in the training and testing set.

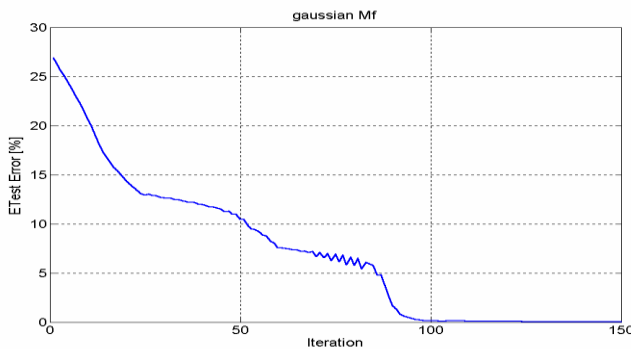


Fig. 5. Decrease of error during the testing process for the ANFIS configuration with Traingular Mf and with Gaussian Mf

Training of the ANFIS can be stopped by two methods. In the first method, ANFIS will be stopped to learn only when the testing error is less than the tolerance limit. This tolerance limit would be defined at the beginning of the training. It is obvious that the performance of the ANFIS that is trained with lower tolerance is greater than ANFIS that is trained with higher tolerance limit.

In this method the learning time will change with the architecture of the ANFIS. The second method to stop the learning is to put constraint on the number of learning iterations. In our study, the ANFIS architecture is stopped to learn after 500 training iterations.

V. DISCUSSION OF RESULTS

This chapter presents the results of experiments and the comparison and analysis of results between the experimental and ANFIS model depending on the greenhouse internal climate parameters. The results and/or the values of internal temperature, internal humidity, CO₂ concentration and internal radiation are graphically represented by means of diagrams depending on the seedlings growth time “Fig. 6”. The values from prediction coincide well with the values from Experiments.

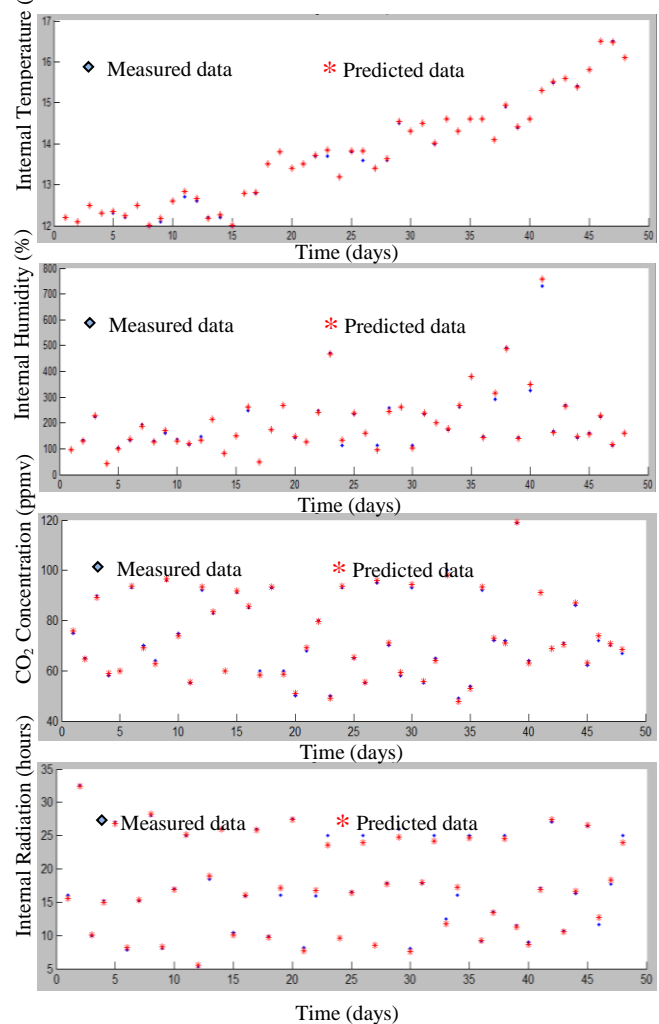


Fig. 6. Comparison of measured and predicted greenhouse internal climate

The predictive capability of using neural network and ANFIS approaches are compared using statistics, which showed that ANFIS predictions for internal temperature (Etest_Tint = 0.723), internal humidity (Etest_Hint = 0.556), CO₂ concentration (Etest_CO₂ = 0.521) and internal radiation (Etest_Rint = 0.385) were for 2% closer to the experimental measurements, compared to 6% using only neural network method.

VI. CONCLUSION

In this paper, an ANFIS is used to successfully estimate the greenhouse climate during seedlings growth process. It can be claimed that the comparison of the results obtained from the ANFIS model and of the experimental results confirms the efficiency and accuracy of the model for predicting the greenhouse climate. By using a back propagation and least square training method, the ANFIS system is trained to an accuracy of 2% error for all four components. The error of the internal climate values predicted by ANFIS with the combination of sigmoidal and gaussian membership function is only 2%, reaching an accuracy as high as 98%. When the traingular membership function is adopted the average error is around 12%, with an accuracy of 92%.

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Face Behavior Recognition Through Support Vector Machines

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Abstract—Communication between computers and humans has grown to be a major field of research. Facial Behavior Recognition through computer algorithms is a motivating and difficult field of research for establishing emotional interactions between humans and computers. Although researchers have suggested numerous methods of emotion recognition within the literature of this field, as yet, these research works have mainly focused on one method for their system output i.e. used one facial database for assessing their works. This may diminish the generalization method and additionally it might shrink the comparability range. A proposed technique for recognizing emotional expressions that are expressed through facial aspects of still images is presented. This technique uses the Support Vector Machines (SVM) as a classifier of emotions. Substantive problems are considered such as diversity in facial databases, the samples included in each database, the number of facial expressions experienced an accurate method of extracting facial features, and the variety of structural models. After many experiments and the results of different models being compared, it is determined that this approach produces high recognition rates.

Keywords—Facial Behavior Recognition; Support Vector Machine; Human Computer Interaction

I. INTRODUCTION

Naturally, Human beings have supremacy in expressing their feelings through different ways such as emotions, body posture, sound, writing, gestures and physiological signals. Most of the major sensory organs of the body are located in the face such as the ears, eyes, nose and mouth. These sensors enable people to hear, see, smell, and taste through biological signals. Besides, they provide output signals which form fundamental portions of human emotions. In general, communication between human beings is classified into verbal and non-verbal communication actions. While human beings' spoken words can be regarded as verbal actions, body movements and physiological reactions can be regarded as communication through non-verbal actions.

Mehrabian indicated that the feeling of the listener with regards to whether he likes or dislikes what he hears is only 7% dependent on the words spoken, and 38% on vocal utterances. Facial expressions influence this feeling up to 55%. This study and the commonly used term “face-to-face” with regards to human interaction indicate that facial expressions play a vital role in human face-to-face interactions [1].

Researchers have categorised facial expressions of feeling into varied categories. Darwin (1872) proposed universal facial expressions of emotion in his evolutionary theory. Floyd Allport (1924), Shalom Asch (1952) and Tomkins (1962, 1963) additionally explored universal emotional facial behavior, although each theorist proposed a unique theoretical basis for his expectation. These theorists proposed that cultural variations are additionally a factor for the variation in facial behavior [2].

Human Computer Interaction (HCI) has developed into an important field of computer study in the last few decades resulting in many forms of communication between computers and human beings. Researchers have worked on and developed numerous solutions to increase the interaction between humans and computers. Emotion detection provides a valuable insight into Human Computer Interaction through computing the direct response of the user. An important reason behind the growth of activity in this research area is the necessity of understanding computers' ability to distinguish between numerous facial emotions.

The main motivations behind this research work are: First, the field of facial behavior recognition through computer algorithms is very demanding for inaugurating emotional communications between humans and computers. Second, despite the fact that researchers proposed several methods of emotion recognition within the literature, nevertheless, their methods primarily have concentrated on using one facial database for evaluating the results of their methods and this might reduce the comparability range and the generalization.

This paper proposes a technique for recognising facial expressions from still pictures taken of people with numerous facial expressions. The main contributions of this system are as follows; first is to use totally two different sets of facial expression that would be extracted via the facial feature point extraction technique, second is to reduce the number of features in the database sets, then, these reduced feature sets are fed into a classification model so that to improve the recognition rate and to determine the performance of the proposed method.

This paper is organised as follows: Section 2 describes previous related works. Section 3 introduces the system structure of the proposed method in which full details in terms of database sets, feature extraction and selection techniques, different structures of a classification method and the

experimental results are demonstrated. Sections 4 presents the main conclusion points and finally, future work steps are suggested.

II. RELATED WORKS

Moon Hwan Kim and his colleagues proposed a way for detecting emotions through frontal facial images in 2005 [3]. For feature extraction, they divided the face into 3 feature regions: the eye region, the mouth region, and the auxiliary region. For every region, features were extracted by comparing geometric and shape data. They used a Fuzzy Color Filter and Histogram Analysis to extract the face region. The facial elements are extracted utilising a Virtual Face Model (VFM) based mostly Histogram analysis. A fuzzy classifier is identified by the Linear Matrix Inequality (LMI) optimization method to classify facial images into five different emotions.

In 2010 Suvam Chatterjee and Hao Shi proposed a brand new method for extracting a novel feature matrix from facial images. They used the Local Binary Pattern (LBP) for extracting facial features, then applied these features to an adaptive Neuro Fuzzy logical thinking System (ANFIS) to determine five facial expressions [4].

Jagdish Lal Raheja and Umesh Kumar introduced a design for human gesture recognition in 2010. The face of a human is detected via employing a technique described by Viola and Jones through the Add Boost Haar classifier. It is then followed by an edge detection method. Since edge detection plays a vital role in finding out the tokens, they therefore, used four accepted algorithms, i.e. Prewitt, Sobel, Prewitt Diagonal, and Sobel Diagonal. The thinning method is applied to scale back the breadth of an edge to one line. After the thinning process, tokens are generated. A Back-propagation Neural Network classifier trained via three completely different gesture images [5].

Priya Metri and his research group studied the multimodal approach for emotion recognition in 2012. They used two completely different models; Facial Expression Recognition and Hand Gesture Recognition, then merging the results of each classifier via employing a third classifier which supplies the ensuing emotion. For extracting features of the face, they followed the favored model of Vagn Walfrid Ekman and Friesen, which identifies all the visually distinguishable facial movements as called the Facial Action Coding (FACS). They used the Clamshift algorithmic program to extract Hand Gesture Unites (HGU). The data from communicative face and hand gestures is classified into labeled emotion classes via a Bayesian classifier that conjointly takes the features of each system then the emotion is recognized [6].

In this paper, the aforementioned works are studied and analysed and in view of that, a new approach is proposed for recognizing facial expressions from still pictures of two different sets of facial expression which would be extracted via the facial feature point extraction technique. Then feature sets are then fed into an SVM classification model. Details of the new approach is explained in the next section.

III. THE SYSTEM STRUTURE

Human face behavior recognition system can be divided up into five main stages: Facial Database, Pre-Processing, Feature Extraction, Feature Selection, and Classification. Figure 1 displays the flowchart of the proposed emotion recognition system structure.

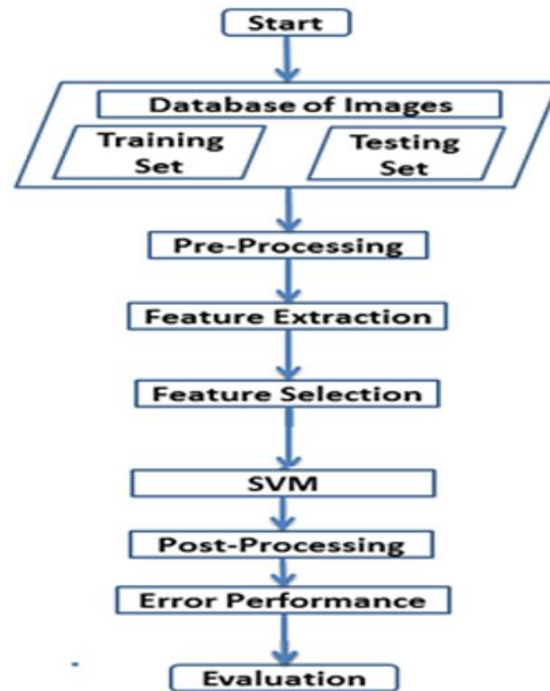


Fig. 1. The Flowchart of the Proposed Emotion Recognition System Structure

Initially, the database samples are divided into up two sets namely; Training set and Testing set. Training set samples composed of 80% of the original database samples and 20% of the samples are specified for testing. The samples are divided randomly. Preprocessing is performed to prepare images for the feature extraction stage. A set of facial feature points is extracted from the images then facial features derived from these points. Different sets of facial features are used for both training and testing classifiers. Facial features are applied onto an SVM. Detailed descriptions of the five stages are shown below:-

A. The Database of Images

Choosing an appropriate database of images is an important task as the specification of images has a significant impact on the results of the emotion recognition system. Different databases are compared based on various criteria such as different subjects of samples, completely different skin colors, variation in size and existence of spectacles, beard, and hair. After conducting a comparison among collections of databases based on the above criteria, the Bosphorus and JAFFE databases were selected as appropriate databases for this research work [7, 8]. The descriptions of database types are explained below:-

1) The Bosphorus Database

This database is planned for research working on 3D and 2D human face processing problems such as expression recognition, facial action unit detection, facial action unit intensity estimation, and facial recognition under poor circumstances, defaceable face modeling, and 3D face restoration. The database contains 105 subjects and 4666 faces all together. Figure 2 shows seven emotions of constant subject of the Bosphorus database. The seven emotions are surprise, sadness, neutral, happy, fear, anger and disgust. The figure shows seven emotions of the same subject. This database is distinctive in three aspects: First, is the well-off repertoire of expressions which includes up to 35 expressions per subject, FACS scoring (includes intensity and asymmetry codes for each AU), and one third of the subjects are certified actors/actresses. Second, is the systematic head which poses 13 yaw and pitch rotations and third is the diversity of face occlusions (beard & moustache, hair, hand, eyeglasses).



Fig. 2. Seven Emotional Expressions of the Bosphorus Database

2) The JAFEE Database

JAFEE Database encloses 213 images of 7 facial expressions (6 basic facial expressions + 1 neutral) posed by 10 Japanese female models. The images are in grayscale mode in this database. Each image has been rated on 6 emotion adjectives by 60 Japanese subjects. These images were taken at the Psychology Department in Kyushu University. Figure 3 shows seven emotions of the same subject.

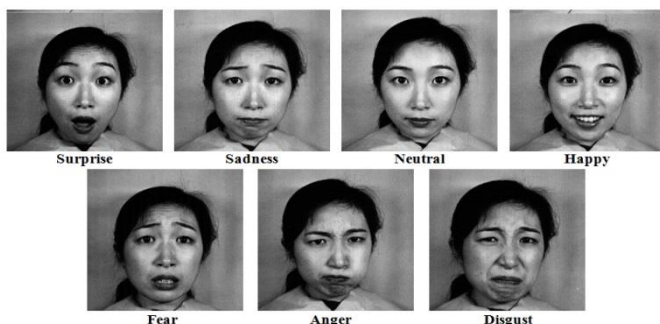


Fig. 3. Seven Emotional Expressions of the JAFEE Database

B. Pre-Processing

The emotion recognition system requires that the input images to be unified in size and brightness especially in our recognition system. The number of pixels of images has an important role in the accuracy of emotion detection. Therefore,

the dimensions of images should be the same for each of the databases. Since all of the images of the JAFEE database have the same dimension (256 X 256) pixels and they are in grayscale mode, so they do not need any filtering or scaling process. Images of the Bosphorus database need to be scaled to an equal size, therefore they are scaled to (384 X 470) pixels.

C. Facial Feature Extraction

In this stage, a review on previous researches is done about the most important characteristics of the human face that have a great impact on the behavior of humans and identifying the regions of the face that are focused by researchers. These regions usually called regions of interest (ROI). Our proposed method for feature extraction is to divide the face into three regions: eye region, mouth region, and auxiliary region. In facial feature extraction stage geometric information is extracted from these regions that can be seen as two separate phases: Facial feature point extraction and Facial feature set detection.

The first step is to find some important points on the face called facial feature points, which are the base for finding the final facial feature set. There are many techniques and tools for extracting these points but not all of them are accurate. Thus, for this purpose, it has been decided in this research work to use the Luxand FaceSDK library as it has many features which are: Easy integration into applications, supports different programming languages (e.g. C++, Visual Basic, C#, Java,...etc.), and supports different platforms (Windows, Linux, Mac OS X), ...etc [9]. This library can detect 66 facial feature points of the face. Not all of the points are used. Only 27 feature points are used in our experiments. Figure 4 illustrates the facial feature points that are found in a sketched face. The red and blue filled circles identify the facial feature points that are used in this paper, while the ellipses are the unused points.

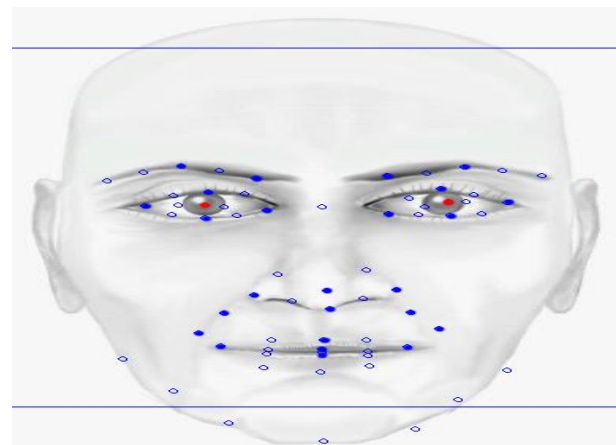


Fig. 4. Facial Feature Points Detected from a Sketched Face (Image Credit: Deepam Pathak)

The second step is to find the final facial feature set which is needed for classification stage. The distance between each pair of these facial feature points could be used as a facial feature. Therefore, the Euclidian Distance Algorithm is used for measuring the distance between each pair of these points. The Euclidean distance between two points (p and q) in a two dimensional space is the length of the line between them.

Identifying facial features from many combinations that can be derived from these points is another important process which needs more investigation. Therefore, the most commonly used features are selected [3, 10, 11]. Figure 5 shows the 15 facial features that are derived from 27 used facial feature points. Each line indicates a facial feature. They are labeled and colorized with different color. Table 1 shows the features extracted from the eye region.

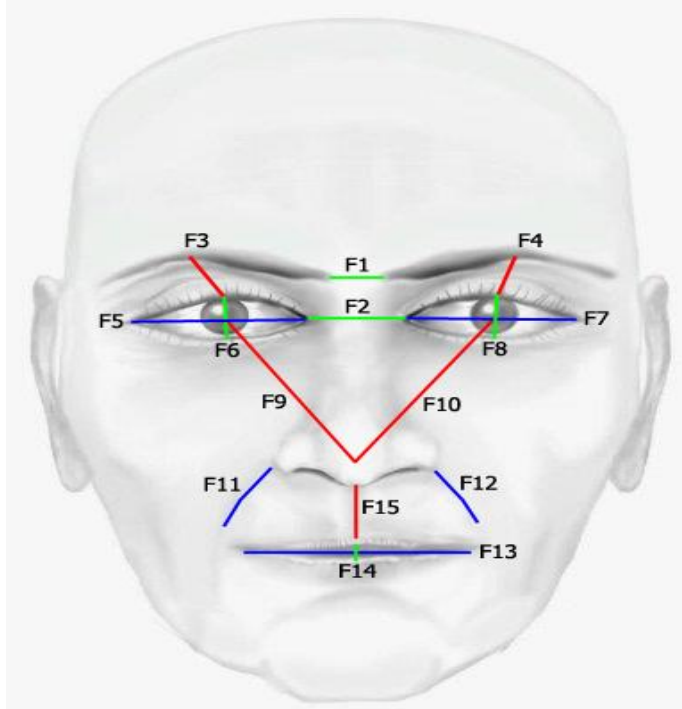


Fig. 5. Facial Features Derived from Facial Feature Points (Image Credit: Deepam Pathak)

TABLE I. FEATURES OF EYE REGION

Features	Description
F3	Distance between upper of right eye and middle of right eyebrow
F4	Distance between upper of left eye and middle of left eyebrow
F5	Right eye width
F6	Right eye height
F7	Left eye width
F8	Left eye height

Table 2 shows the features extracted from the mouth region.

TABLE II. FEATURES OF MOUTH REGION

Features	Description
F13	Mouth width
F14	Mouth inner height
F15	Distance between upper of mouth and bottom of nose

Table 3 shows the features extracted from the auxiliary regions.

TABLE III. FEATURES OF THE AUXILIARY REGIONS

Features	Description
F1	Distance between inner corner of both eyebrows
F2	Distance between inner corner of both eyes
F9	Distance between nose tip and center of right eye
F10	Distance between nose tip and center of left eye
F11	Right nasolabial fold
F12	Left nasolabial fold

D. Feature Selection

In this stage, the most useful features will be selected among the whole set of features. Identifying important features can be done by different techniques like PCA, SVM, etc. Some of these methods have been used to identify and rank the importance of each feature [12, 13, 14]. A combination process is conducted for some of the features, since the face is composed of two similar sides; right side and left sides of the face. Thus, features like width of both eyes might be nearly the same. Basically, Duplicating features might decrease the classification accuracy. A typical solution for this is to combine or taking the average of these features.

E. Classification

Different subsets of the facial feature set are used as inputs to the classifier. The seven facial expressions are encoded in natural numbers (see Table 4). They are used for measuring the performance of learning of both training and testing stages of the system

TABLE IV. CODE VALUE OF EACH EXPRESSION

Expression	Code
Anger	1
Disgust	2
Fear	3
Happy	4
Neutral	5
Sadness	6
Surprise	7

SVM is a classifier method that can be used for classifying emotions [14, 15]. It classifies the input data into class labels by finding the maximum-margin hyperplanes, which is a line, plane or hyperplane. This process maximises the distance between the line and the nearest data points. SVM process can be summarised in three steps: The first step is to find the hyperplanes in the feature space which is able to classify input features. Since emotion detection for seven different expressions is a nonlinear problem due to the high dimension of input features, mapping is performed for each input sample to its representation in the feature space. Maximising the margin and evaluating the decision function both require the computation of the dot product in a high-dimensional space. In this research work, different experimental tests are carried out using different values of parameters of the support vector machines such as the type of Kernel function, the degree of the kernel function, cost, coefficient, and the gamma are presented in real numbers, then in each test, the time taken to build the model in seconds (TTBM), percentage number of correctly classified instances (CCI), percentage number of incorrectly

classified instances (ICI), mean absolute error (MAE), and root mean squared error (RMSE) are computed. This above styling format is used for all other case models. Descriptions of different case models on each database set are explained:-

a) Bosphorus Database

1) Case 1

In case 1, Table 5 shows the input features selected for classification, the output class labels of emotions, and the parameters of the used classifier. These parameters are used for both training and testing the network. The input features are presented as “F1-F15” which means all 15 features are used. The output is formatted as “1-7” which means all 7 facial expressions are used in this experiment.

TABLE V. CASE 1 - CLASSIFIER PARAMETERS

Input Features	Output (Class)	Kernel Type	Degree	Cost	Coef0	Gamma
F1-F15	1-7	Polynomial	3	1	4	1.2

Table 6 shows the results of for Case 1 model classifier.

TABLE VI. CASE 1 - RESULTS OF TRAINING AND TESTING PHASE

Phase	TTBM (Sec)	CCI (%)	ICI (%)	MAE	RMSE
Training	8.73	100	0	0	0
Testing	8.73	78.3784	21.6216	0.0618	0.2485

Table 7 shows the confusion matrix of the training phase of case 1 model. The results show that all the instances of all classes are correctly classified.

TABLE VII. CASE 1 - CONFUSION MATRIX OF TRAINING PHASE

Classified As	Anger	Disgust	Fear	Happy	Neutral	Sadness	Surprise
Anger	66	0	0	0	0	0	0
Disgust	0	58	0	0	0	0	0
Fear	0	0	51	0	0	0	0
Happy	0	0	0	92	0	0	0
Neutral	0	0	0	0	81	0	0
Sadness	0	0	0	0	0	47	0
Surprise	0	0	0	0	0	0	51

Table 8 shows the confusion matrix of the testing phase of case 1 model. It appears that the Neutral class has the most incorrectly classified instances. The lowest classification accuracy of the Neutral class is due to the problem of the Fear class and the Sadness class, since it has the most interference with these two classes.

TABLE VIII. CASE 1 - CONFUSION MATRIX OF TESTING PHASE

Classified As	Anger	Disgust	Fear	Happy	Neutral	Sadness	Surprise
Anger	6	0	2	0	0	2	0
Disgust	1	10	0	1	0	1	0
Fear	2	0	14	0	0	2	1
Happy	0	0	0	25	0	0	0
Neutral	0	0	4	0	15	2	1
Sadness	1	1	0	1	0	8	0
Surprise	0	0	2	0	0	0	9

2) Case 2

In this case, instances of the Fear class are removed from the data set in order to compare results with other models that discarded the Fear class. Table 9 shows the classifier parameters of case 2 model.

TABLE IX. CASE 2 - CLASSIFIER PARAMETERS

Input Features	Output (Class)	Kernel Type	Degree	Cost	Coef0	Gamma
F1-F15	1,2,4,5,6,7	Polynomial	3	1	0.5	2

Table 10 shows the results of case 2 model classifier.

TABLE X. CASE 2 - RESULTS OF TRAINING AND TESTING PHASE

Phase	TTBM (Sec)	CCI (%)	ICI (%)	MAE	RMSE
Training	6.05	100	0	0	0
Testing	6.09	85.567	14.433	0.0481	0.2193

Table 11 shows the confusion matrix of the training phase of case 2 model classifier.

TABLE XI. CASE 2 - CONFUSION MATRIX OF TRAINING PHASE

Classified As	Anger	Disgust	Happy	Neutral	Sadness	Surprise
Anger	65	0	0	0	0	0
Disgust	0	58	0	0	0	0
Happy	0	0	92	0	0	0
Neutral	0	0	0	80	0	0
Sadness	0	0	0	0	46	0
Surprise	0	0	0	0	0	49

Table 12 shows the confusion matrix of the testing phase of case 2 model classifier.

TABLE XII. CASE 2 - CONFUSION MATRIX OF TESTING PHASE

Classified As	Anger	Disgust	Happy	Neutral	Sadness	Surprise
Anger	8	0	0	1	2	0
Disgust	1	10	1	0	1	0
Happy	0	0	25	0	0	0
Neutral	2	0	0	18	2	1
Sadness	1	1	1	0	9	0
Surprise	0	0	0	0	0	13

3) Case 3

For identifying the accuracy of identical models for both databases, the Sadness class together with the Fear class are removed from the data set in this model. Table 13 shows the classifier parameters of case 3 model.

TABLE XIII. CASE 3 - CLASSIFIER PARAMETERS

Input Features	Output (Class)	Kernel Type	Degree	Cost	Coef0	Gamma
F1-F15	1,2,4,5,7	Polynomial	3	1	0.2	2.4

Table 14 shows the results of case 3 model classifier.

TABLE XIV. CASE 3 - RESULTS OF TRAINING AND TESTING PHASE

Phase	TTBM (Sec)	CCI (%)	ICI (%)	MAE	RMSE
Training	3.68	100	0	0	0
Testing	3.12	89.5349	10.4651	0.0419	0.2046

Table 15 shows the confusion matrix of the training phase of case 3 model classifier.

TABLE XV. CASE 3 - CONFUSION MATRIX OF TRAINING PHASE

Classified As	Anger	Disgust	Happy	Neutral	Surprise
Anger	65	0	0	0	0
Disgust	0	58	0	0	0
Happy	0	0	91	0	0
Neutral	0	0	0	80	0
Surprise	0	0	0	0	49

Table 16 shows the confusion matrix of the testing phase of case 3 model classifier.

TABLE XVI. CASE 3 - CONFUSION MATRIX OF TESTING PHASE

Classified As	Anger	Disgust	Happy	Neutral	Surprise
Anger	9	1	0	1	0
Disgust	1	10	2	0	0
Happy	0	0	26	0	0
Neutral	3	0	0	19	1
Surprise	0	0	0	0	13

b) The JAFEE Database

4) Case 4

In this case, Table 17 shows the input features selected for classification, the output class labels of emotions, and the parameters of the used classifier.

TABLE XVII. CASE 4 - CLASSIFIER PARAMETERS

Input Features	Output (Class)	Kernel Type	Degree	Cost	Coef0	Gamma
F1-F15	1-7	Polynomial	3	1	0.7	1.5

Table 18 shows the results of Case 4 classifier model.

TABLE XVIII. CASE 4 - RESULTS OF TRAINING AND TESTING PHASE

Phase	TTBM (Sec)	CCI (%)	ICI (%)	MAE	RMSE
Training	1.56	100	0	0.1587	0.2422
Testing	1.33	65.1163	34.8837	0.1638	0.2769

Table 19 shows the confusion matrix of the training phase of case 4 model. This matrix clearly shows the correctly and incorrectly classified instances. The vertical labels identify the desired facial expressions that are classified as the expressions in the horizontal labels.

TABLE XIX. CASE 4 - CONFUSION MATRIX OF TRAINING PHASE

Classified As	Anger	Disgust	Fear	Happy	Neutral	Sadness	Surprise
Anger	26	0	0	0	0	0	0
Disgust	0	26	0	0	0	0	0
Fear	0	0	28	0	0	0	0
Happy	0	0	0	24	0	0	0
Neutral	0	0	0	0	22	0	0
Sadness	0	0	0	0	0	25	0
Surprise	0	0	0	0	0	0	19

Table 20 shows the confusion matrix of the testing phase of case 4 model. We can see that none of the instances of Sadness class are correctly classified.

TABLE XX. CASE 4 - CONFUSION MATRIX OF TESTING PHASE

Classified As	Anger	Disgust	Fear	Happy	Neutral	Sadness	Surprise
Anger	3	0	1	0	0	0	0
Disgust	0	1	0	1	0	1	0
Fear	0	0	2	0	0	1	1
Happy	0	0	0	7	0	0	0
Neutral	0	0	1	0	5	2	0
Sadness	2	0	2	1	1	0	0
Surprise	0	0	1	0	0	0	10

5) Case 5

In case 5, the Fear class is removed from the data set that is because of the reasons specified in case 2. Table 21 shows the classifier parameters of case 5 model.

TABLE XXI. CASE 5 - CLASSIFIER PARAMETERS

Input Features	Output (Class)	Kernel Type	Degree	Cost	Coef0	Gamma
F1-F15	1,2,4,5,6,7	Polynomial	3	1	0	0.79

Table 22 shows the results of case 5 model classifier.

TABLE XXII. CASE 5 - RESULTS OF TRAINING AND TESTING PHASE

Phase	TTBM (Sec)	CCI (%)	ICI (%)	MAE	RMSE
Training	0.99	93.1034	6.8966	0.1598	0.2398
Testing	0.85	77.7778	22.2222	0.158	0.2637

Table 23 shows the confusion matrix of the training phase of case 5 model classifier.

TABLE XXIII. CASE 5 - CONFUSION MATRIX OF TRAINING PHASE

Classified As	Anger	Disgust	Happy	Neutral	Sadness	Surprise
Anger	18	5	0	1	2	0
Disgust	0	27	0	0	0	0
Happy	0	0	25	0	0	0
Neutral	0	0	0	23	0	0
Sadness	0	2	0	0	23	0
Surprise	0	0	0	0	0	19

Table 24 shows the confusion matrix of the testing phase of case 5 model classifier.

TABLE XXIV. CASE 5 - CONFUSION MATRIX OF TESTING PHASE

Classified As	Anger	Disgust	Happy	Neutral	Sadness	Surprise
Anger	3	1	0	0	0	0
Disgust	0	1	1	0	0	0
Happy	0	0	6	0	0	0
Neutral	0	0	1	5	1	0
Sadness	1	0	1	1	3	0
Surprise	0	0	1	0	0	10

6) Case 6

According to case 5 results, the Sadness class has the lowest recognition accuracy. Therefore, the Sadness class with the Fear class are removed from the data set. Table 25 shows the classifier parameters of case 6 model.

TABLE XXV. CASE 6 - CLASSIFIER PARAMETERS

Input Features	Output (Class)	Kernel Type	Degree	Cost	Coef0	Gamma
F1-F15	1,2,4,5,7	Polynomial	3	1	2	0.3

Table 26 shows the results of case 6 model classifier.

TABLE XXVI. CASE 6 - RESULTS OF TRAINING AND TESTING PHASE

Phase	TTBM (Sec)	CCI (%)	ICI (%)	MAE	RMSE
Training	0.46	91.6667	8.3333	0.1582	0.2348
Testing	0.43	90	10	0.1441	0.2419

Table 27 shows the confusion matrix of the training phase of case 6 model classifier.

TABLE XXVII. CASE 6 - CONFUSION MATRIX OF TRAINING PHASE

Classified As	Anger	Disgust	Happy	Neutral	Surprise
Anger	22	2	0	2	0
Disgust	2	25	0	0	0
Happy	0	0	24	0	1
Neutral	0	0	0	23	0
Surprise	0	0	0	3	16

Table 28 shows the confusion matrix of the testing phase of case 6 model classifier.

TABLE XXVIII. CASE 6 - CONFUSION MATRIX OF TESTING PHASE

Classified As	Anger	Disgust	Happy	Neutral	Surprise
Anger	3	1	0	0	0
Disgust	0	1	1	0	0
Happy	0	0	6	0	0
Neutral	0	0	1	6	0
Surprise	0	0	0	0	11

The following figures demonstrate the difference between various models of the classifier that are used via showing the classification accuracy for both training and testing phases. In each figure, the classification accuracy of the SVM classifier is shown for three models namely; 7 Class Model, 6 Class Model, and 5 Class Model (See Figures 6, 7, 8, and 9).

In table 29 the summarised results of all experiments are shown.

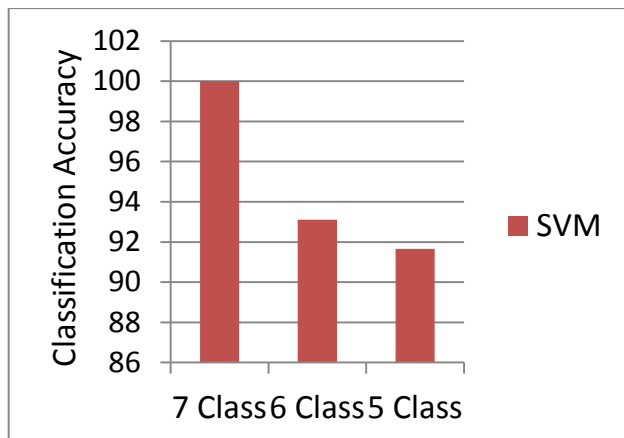


Fig. 6. Classification Accuracy of Training Phase of the JAFEE Database

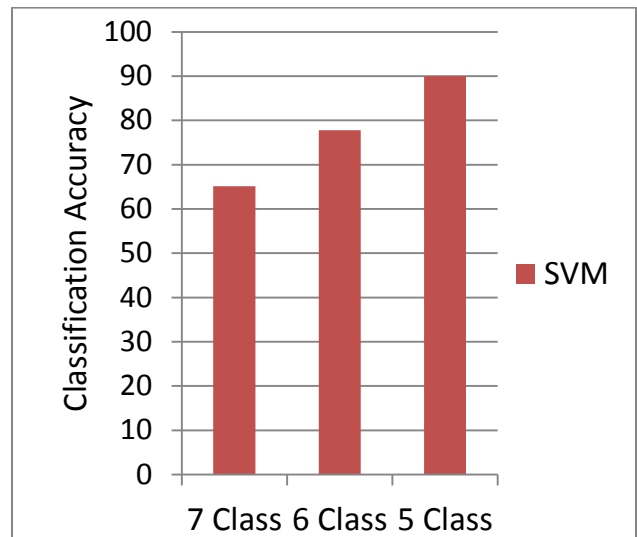


Fig. 7. Classification Accuracy of Testing Phase of the JAFEE Database

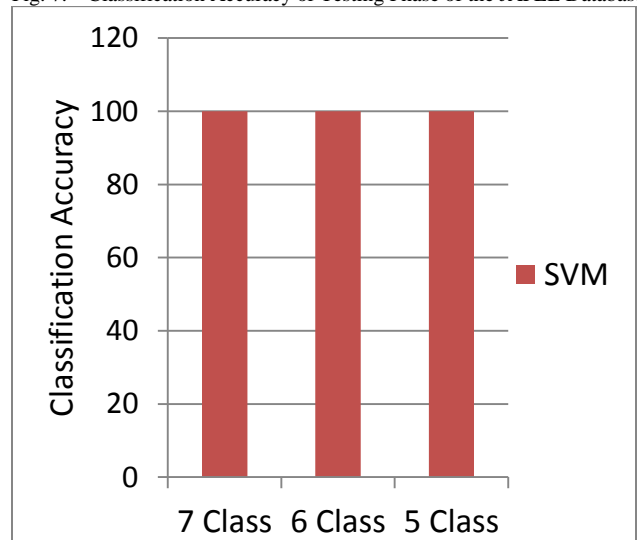


Fig. 8. Classification Accuracy of Training Phase of the Bosphorus Database

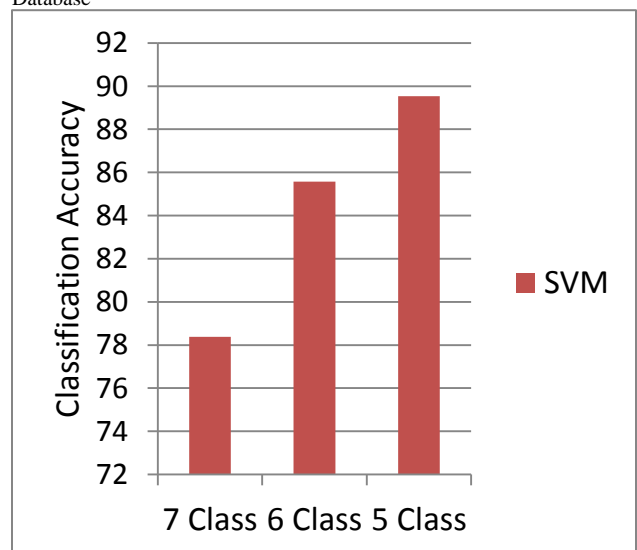


Fig. 9. Classification Accuracy of Testing Phase of the Bosphorus Database

It can be seen that the classification accuracy, or in other words, the recognition rate increases when the number of classes decreased. This statement is true for all testing phases in both databases. It can be realised from the results that almost all of the models give a satisfying outcome, except the 7 class model of the JAFEE database which scores the lowest among all other models (indicated in red colour). The five class model of the JAFEE database using SVM classifiers provides a good recognition rate (indicated in green colour). Table 29 shows percentage of classification accuracy of different experiments.

TABLE XXIX. PERCENTAGE OF CLASSIFICATION ACCURACY OF DIFFERENT EXPERIMENTS

Data Set	Model	SVM	
		Training	Testing
JAFEE	7 Class	100	65.116
	6 Class	93.1034	77.7778
	5 Class	91.6667	90
Bosphorus	7 Class	100	78.3784
	6 Class	100	85.567
	5 Class	100	89.5349

Table 30 shows a comparison between the results of the proposed technique with another approach that was proposed by [3]. Although the used facial databases are not the same in these approaches, the comparison is performed, because in both research works five classes of emotions as an experiment is used. The results prove that the proposed approach provides a much greater recognition rate.

TABLE XXX. PERCENTAGE OF CLASSIFICATION ACCURACY OF OUR PROPOSED TECHNIQUE COMPARED TO A SIMILAR APPROACH

The proposed Approach SVM Classifier	Approach Fuzzy Classifier [3]
90	74

IV. CONCLUSION

From the results of all case models that were considered during this research work, many conclusions can be formed concerning the used data samples, the techniques used for each facial feature extraction and feeling classification. Variation within the subjects of the dataset is very important, since the dataset would be additionally generalized. The very best recognition rate of SVM classifier is 89.5349 % for the 5 classes model of emotions of the Bosphorus database, which is close to the result of the same model of the JAFEE database which is 90%. This demonstrates that this approach for emotion recognition obtains high recognition rates. It is conjointly observed that the fear expression has the greatest interference with other expressions.

V. RECOMMENDATIONS FOR FUTURE WORK

Although the proposed approach in this paper provides promising results for emotion recognition, nonetheless, there are still some other areas that can be considered for studying as a step for more improvement and generalization. Some of the most important ideas for future work are listed as follows:-

- 1) Using some other algorithms for classification such as Learning Vector Quantization, Quadratic Classifier, Iterative Dichotomiser 3 (ID3) and Nearest Neighbour Classifiers,
- 2) It is also advisable to consider Multi-label Classification as another method for classification. This

method assigns multiple target labels to each of the instances. Thus, the process of transforming the data set from a single label into a multi-label problem must be considered and the classifier algorithm has be adjusted so that to directly perform multi-label classification.

- 3) It is worthy to use other facial databases to cover various subjects from several cultures.
- 4) It is a good idea to intermingle various facial databases for experimentations.
- 5) Study a real-time emotion recognition from this research work.

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VoIP Forensic Analyzer

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Abstract—People have been utilizing Voice over Internet Protocol (VoIP) in most of the conventional communication facilities which has been of assistance in the enormous attenuation of operating costs, as well as the promotion of next-generation communication services-based IP. As an intimidating upshot, cyber criminals have correspondingly started interjecting the environment and creating new challenges for the law enforcement system in any Country. This paper presents an idea of a framework for the forensic analysis of the VoIP traffic over the network. This forensic activity includes spotting and scrutinizing the network patterns of VoIP-SIP stream, which is used to initiate a session for the communication, and regenerate the content from VoIP-RTP stream, which is employed to convey the data. Proposed network forensic investigation framework also accentuates on developing an efficient packet restructuring algorithm for tracing the depraved users involved in a conversation. Network forensics is the basis of proposed work, and performs packet level surveillance of VoIP followed by reconstruction of original malicious content or network session between users for their prosecution in the court.

Keywords—Forensics; Packet Reordering; Session Initiation; Real Time Transfer

I. INTRODUCTION

Network forensics is a technique of identifying network anomalies and breaches from the pattern of packets based-network. To understand network usage, a content level analysis of individual traffic flow must be conducted to retrieve the behavior and interest pattern of any intruder from the information stored in the networks as packets. This activity is entitled retrospective network analysis or network forensics [1].

During the misuse of any network activity, network forensic process accesses forensic data through an offline packet level inspection. An identical forensic framework captures network session for tracing the malicious contents involved in the particular network usage. It also grabs down complete identity credentials to prosecute the culprits in the court of law. Currently, a majority of the online telephony is through VoIP mechanism, which is of less operational cost and great flexibility. VoIP applications have grown-up in popularity in the recent years because they facilitate free voice and video chat and connect an establishment between VoIP and Public Switched Telephone Network (PSTN).

A. Challenges of VoIP Forensic Frameworks

In VoIP telephony scenario, a VoIP client uses port randomization for sending actual voice through RTP communication. As a concern, obtainable frameworks for forensic processing fall short in acquiring the imperative forensic details and reinstating the original data to track down the cyber intruders that abuse VoIP software infrastructure to transact their detestable contents. Unlimited use of VoIP telephony consumes most of the network bandwidth in organization and industries, which tend network towards jam in it also the upsurge of VoIP frameworks, challenge barely the respective country's judiciary system, as there is no single assembly point resembling public switch telephone network for centralized control and monitoring.

B. Our Contribution

This paper mainly outlines the idea behind a structural framework developed for forensic process and investigation of VoIP along with GUI design, which produced thriving forensic outcomes in reconstructing the original voice in VoIP conversation. Details of any immoral users and their activities has been traced from these conversation sessions.

This paper also deliberates a specially designed time stamped VoIP packet rescheduling strategy for procuring malevolent packet contents from the respective network pattern. Projected structural framework hosts a fully-fledged high-speed packet-bagging module of its own, which work independently or can associate with third party high-speed packet bagging software and hardware. Following is the order of the paper: Section II bonds with literature review, Section III labels component overview of VoIP, Section IV gives an idea of the working of VoIP, Section V provides workflow of the forensic analyzer, and Section VI outlines the results and discussions, followed by conclusion and references.

II. LITERATURE REVIEW

There are scores of apposite forensic processing tools to analyze, dissect and investigate network packets or network streams. Former such works do not support reconstruction of actual voice data in VoIP connection or rather discussed and analyzed VoIP environments for tracing forensic details. This section includes brief outline of various approaches, which fall into forensic analysis of VoIP streams and stimulated us to develop VoIP forensic processing and investigative strategies

subsequently a comprehensive study. Soundly most of these methods are complex and shareware with restricted functions.

François, J et al. (2010) presented a technique to identify smart devices from a VoIP stream of traffic through a fingerprint process. They were attentive in signaling plane and castoff voice information. This work helped us to study the network pattern of VoIP sessions [2].

Ibrahim, M, et al. (2010) proposed a structural architecture of examining attacks using VoIP. The architecture reinforces information gathering against a possible attack situation. This facilitated us in designing an efficient packet reconstruction algorithm for VoIP sessions [3].

Hofbauer, S,et al. (2012) offered a strategic approach for maintaining communication details and trace VoIP attacks by conserving caller information. This drove us into tracing more forensic information from VoIP sessions [4].

Gao Hongtao(2011) put for a method that forwards VoIP crime analysis on a host computer. His work aided us in designing a proposed forensic method of forensic analyzer for tracing malevolent VoIP usage [5].

Azab, A et al.(2012)open component analysis of skype with respect to call progress. This assisted us to recognize call processing in Skype VoIP environments [6].

Y.Q. Wang (Wang et al., 2011) enlightened an in depth computer forensic skills in dealing with the criminal activities in both wireless and wired communication networks. They emphasize on the credibility, the depth, the extensiveness and the legality of the computer forensic activities [7].

Khidir M. Ali (2012) stated various Practices and Managerial Implications in the field of digital forensics which gave the summary of forensic computing and deliberated key issues to be measured in forensic investigations and digital evidence analysis practices [8].

Edewede Oriwoh et al. (2013) described a technique for Forensics Edge Management framework that autonomously provided security and forensic services within the home Internet of Things [9].

III. COMPONENT OVERVIEW OF VOIP

The foremost signaling ingredient of VoIP infrastructure is Session Initiation Protocol (SIP), which was developed by Internet Engineering Task Force that allows end devices to begin and conclude network communication sessions. The servers and user agents are two significant components of SIP architecture. The User Agent Client (UAC) and User Agent Server (UAS), which respectively generate request and responses, represent a subsystem for user agents. SIP servers are not necessarily separate physical devices but different entities or distinct functions [15]. Four of these server functions are defined as follows.

A. Sip Servers

Proxy servers: They are truly hosts in networks and act as intermediary for clients to initiate request for other clients or hosts in the same network and route SIP generated requests towards UAS, and route SIP generated responses to UAC.

Redirect servers: They act as logical hosts by which client are guided towards a set of alternative Uniform Resource Indicators to accomplish a particular task.

Registrar servers: They accept and process registration based communication messages so that location of end users can be traced.

Location servers: They link address and respective network domain by providing a conceptual database of addresses. Location servers work in parallel with the registration servers to find out alternative set of URIs for a request when generated [15].

B. Sip Messages

Messages of SIP have two defined structures such as requests from client to server and responses from server to clients indicating the status of request. Different types of request messages used by SIP are briefly sketched below.

INVITE: This message is used to invite clients or users to be a part of specific communication session. The content of this message gives description of specific communication session.

ACK: the client uses this message with INVITE messages to indicate confirmation of reception of response in a specific communication session.

OPTIONS: This message is to request for capabilities of the server.

BYE: This message is made sent when a particular user wants to quit a specific communication session.

CANCEL: This message is used to abandon unsettled request

REGISTER: a user or client to start specific communication session uses this message.

RTP: This Real Time Transfer Protocol establishes end-to-end transmission of data such as voice, video over services of network using unicast or multicast strategies. In the case of VoIP, the ports for the RTP communication are decided by the initial handshake done on the SIP.

SDP: The format of initialization parameters for real time transfer is given by this Session Description Protocol and is attached in the SIP message body [15].

IV. WORKING OF VOIP

The functioning of VoIP using SIP and RTP can be worked out into 3 steps. First is the connection establishment using SIP followed by information transference using RTP and connection termination using SIP. To establish a communication as in Fig.1, client1 sends an invite request to the server to be sent to client 2. The server sends an invite request to the client 2; concurrently it sends a trying message to client 1. Once the server gets a ringing response from client 2 it will forward the same to client 1. When client 2 sends the consent it will exchange OK message between server and client1. Now client1 acknowledges the established connection by sending an ACK signal to the server and the server will send that ACK to the client2. Now the connection has been

instituted. During this handshake process the invite message and the OK status message contains the SDP in the SIP body, which will provide information about the ports, and other parameters required for the RTP to perform multimedia streaming. Once the call is concluded the bye request is exchanged between client and server in subsequent steps. The second client will respond with an OK message and the connection will be terminated [3].

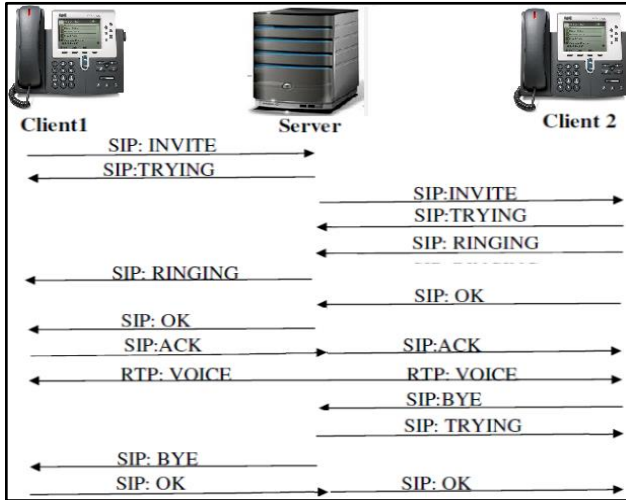


Fig. 1. Working of VoIP handshake

V. ARCHITECTURAL WORK FLOW OF VOIP FORENSIC ANALYSER

The architecture of proposed VoIP forensic analyzer is shown in Fig. 2, which executes offline forensic analysis and recreates the absolute voice data between the VoIP clients. The subjected VoIP stream will be seized and observed when a particular misuse of network environment reported. The bagged network session is then undergone through an assortment of analysis and forensic processing phases by the projected architecture to trace proofs against malevolent communications. The ultimate goal is to supply adequate indications to let the offender to be impeached by respective judiciary system. This chief architectural framework hosts following major modules.

A. PCAP File Crafting Unit

This unit is used to bag the network packets through LAN, Wi-Fi interface of standalone computers or proxy servers in LAN, whichever is pertinent in network environment to create a PCAP file. Network packets are enclosed in a PCAP file with extension “.pcap”. The “PCAP” file is the foremost input for projected architecture where it undergoes succession of modification throughout the forensic processes [14].

B. VoIP Session Recreating Unit

This unit is the spotlight of forensic processing by way of recreating the genuine network stream from VoIP clients. This module rebuilds the existent voice from VoIP conversation. The foremost activities accomplished at various sub stages are described below.

1) SIP Filter

SIP packets are filtered to scrutinize the initial handshake done between the client and the server to mine forensic details about VoIP initialization constraints. The default port of the SIP protocol is 5060. The filtering is done in accordance with this port. Once filtered the SIP packets from the source PCAP file is separated and saved as temporary PCAP file and is fed to the next stage.

2) Call Identifier

In this sub section, the analyzer will try to identify IP addresses of those clients who have made a call with another client in that network. From the section IV, it is seen that when the connection is established between the two clients an ACK signal is send from the client who initiates the call to the server and from there to the other client. Hence, our framework is designed in such a way that it parses the SIP packets for those IP's that send an ACK signal or receive an ACK signal (other than the server). By this strategy, VoIP analyzer collects the IP's and other forensic information of those clients who have participated in a particular VoIP conversation.

3) Port Identifier

There is no default port used for RTP traffic in any VoIP communication. To identify the port that is used for a specific communication, the initial handshake made for that communication is analyzed and separated followed by extraction of the ports.

4) Message Parser

This component traces SDP messages, which consist of INVITE and STATUS messages by filtering and identifying necessary VoIP packets through all available ports identified by port identifier. Parsed sample format of these messages are shown below.

5) INVITE message sample format.

```
INVITE sip:1006@10.100.13.139:5060 SIP/2.0
Via: SIP/2.0/UDP 10.100.12.230:56727;
branch=z9hG4bK-d8754z-ae78c05f8f58042a-1--
-d8754z-;rport
Max-Forwards: 70
Contact: <sip:1004@10.100.12.230:56727;
rinstance=a8c18539cb50ec97>
To: <sip:1006@10.100.13.139:7020>
From: <sip:1004@10.100.13.255:5060>;tag=f11afe5a
Call-ID: M2JhYjBjMGRkYzQzNTA2ZmFm
YWU4MzViN2NiOTVIMTA.
CSeq: 1 INVITE Allow: INVITE, ACK, CANCEL,
OPTIONS, BYE, REGISTER, SUBSCRIBE,
Content-Type: application/sdp
Supported: replaces
User-Agent: 3CXPhone 4.0.13679.0
Content-Length: 407
v=0o=3cxVCE 375041100 14258475
IN IP4 10.100.12.230
c=IN IP4 10.100.12.230t=0 0
```

```
m=audio 40006 RTP/AVP 0 8 3 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15aptime:20 a=sendrecv
m=video 40004 RTP/AVP 34
c=IN IP4 10.100.12.230
a=rtpmap:34 H263/90000
39a=fmtp:34 QCIF=1;CIF=1;SQCIF=1;CIF4=1;
a=sendrecv
```

Thus for each IP's SIP packets are parsed for INVITE and STATUS 200 OK messages. From these SIP packets, SDP parts are separated and parsed for tracing ports.

6) STATUS 200 OK message sample format

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
10.100.12.230:56727;branch=z9hG4bK-d8754z-
8f3c792019497b01-1---d8754z-
received=10.100.12.230;rport=56727
From: <sip:1004@10.100.13.255:5060>;tag=f11afe5a
To: <sip:1006@10.100.13.139:7020>;tag=as043f7e02
Call-ID: M2JhYjBjMGRkYzQzNTA2Zm
CSeq: 2 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE,
REFER, SUBSCRIBE, NOTIFY,INFO
Contact: <sip:1006@10.100.13.139>
Content-Type: application/sdp
Content-Length: 309
v=0
c=IN IP4 10.100.13.139
t=0 0
m=audio 18638 RTP/AVP 3 0 8 101
a=rtpmap:101 telephone-event/8000
a=silenceSupp:off - - -
a=sendrecv
m=video 0 RTP/AVP 34
```

The forensic analyzer tracks the IP addresses and ports caught up in the VoIP conversation from the above parsed messages. Here the message part "m=audio" followed by the number gives the ports assigned by that client for that specific communication. The "From" and "To" fields contain a 4-digit number followed by '@' symbol which provides the IP address of the client who is involved in that particular exchange. [3]

7) RTP Filter

The process of filtering the RTP packets begins as soon as the forensic analyzer traces the ports and IPs used for the VoIP communication. To accomplish this, the forensic analyzer filters all the UDP packets having the selected ports as the source and destination ports since most of the RTP transfer occurs through lightweight UDP packets. RTP filter collects all such UDP packets by using the port number identified port identifier.

8) RTP Rescheduler

Once RTP packets are filtered from a particular communication, the projected packet-reordering algorithm, as

given in Fig.2, reorganizes RTP packets based on the sequence no and separates the data from the packets. The sequence no of the RTP packets ranges from zero to 65536. Once it reaches 65536, it again starts from zero. The marker bit M is set for the beginning of the communication. The time stamps give idea about the time at which the packet is sent. The PT (payload type) specifies the type of the codec used for the payload content. The projected VoIP forensic analyzer will extract the voice information associated with RTP packet payload and writes to PCAP file temporarily. Now packets are reordered according to the packet-reordering algorithm and are given briefly in the following section. The respective RTP packet format with its parameters are given in Fig 3. For a conceptual view.

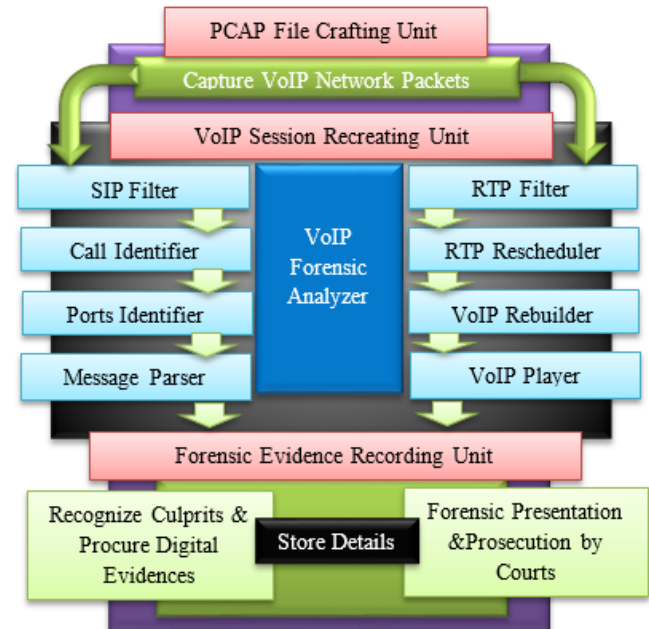


Fig. 2. Architecture of VoIP Forensic Analyzer

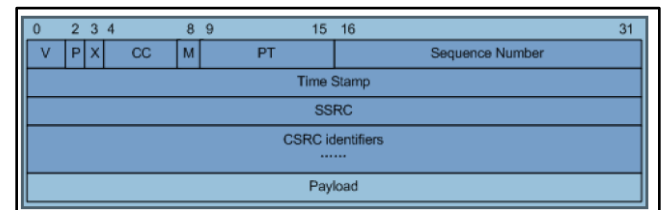


Fig. 3. RTP Packet Format

9) VoIP Stream Rebuilder

This vital component of proposed architecture rearranges the RTP communication session stream from the identified and separated combinations of IPs and Ports of involved VoIP clients. RTP packet payloads will be combined together after extracting it from header section of RTP packet followed by respective session rescheduling to reconstruct actual voice data [13, 14]. The proposed set of TCP/UDP reordering algorithms developed and given in this section would accomplish this task of rescheduling and rebuilding VoIP communication stream. Thus recreated voice stream is warehoused for upcoming forensic processing by forensic

evidence recording unit of the projected VoIP analyzer. The first algorithm will identify the VoIP stream by filtering all SIP packets available in the network session followed by tracing all combinations of IPs and Ports. The second algorithm accomplishes the tracing of retransmitted or duplicate packets from VoIP communication stream. The reordering of VoIP communication stream is carried out by third algorithm. In this set of algorithms, “next” variable directs the subsequent packets. “new_time_seq” indicates regenerated timestamp for each such packets. “packet” denotes present packet for forensic processing. “EOF” designates a packet’s end. “seq_relative” indicates subsequent probable time stamp of the packet. “datalength” provides packet’s length. UDP packet’s offset value is represented by

“next.flag”. We briefly outline the structure of set of algorithms in this section. During the deployment, we consider essential packet parameters and execute packet-processing functions. Thus, this set of algorithm reproduce the actual voice data present in respective VoIP communication session stream [11,14]. The set of three algorithms developed are given below.

10) VoIP Player

VoIP forensic player module automatically identify the codecs used by VoIP communication from the packets and decodes the processed voice data with help of SIP’s codec G.729, which has already encoded using G.711. This player, plays recreated VoIP steam and voice in it perfectly.

Algorithm 1: Identify and filter network SIP packets for Time Stamping

Input: PCAP file

Output: Source and Destination IPs, Ports, Time Stamped PCAP File

Initialize next=1, new_time_seq=0;

While packet!= null **do**

Read the packet from the PCAP file

if packet= EOF, Filter the SIP packets, Affix new_time_seq and save to Time Stamped PCAP file

Extract all combination of Source, Destination IPs and Ports used by RTP from SIP packets

end if

end while

Algorithm 2: Separate duplicate and retransmitted packets

Input: Time stamped PCAP file, Source and Destination IPs, ports

Output: Time Stamped PCAP file of retransmitted packets

Initialize next=1, new_time_seq=0;

While packet!= null **do**

Read the packet from the PCAP file

if packet= EOF, Filter the packets based on IP and Ports **end if**

if next flag then seq=packet.new_time_seq, seq2=packet.new_time_seq-seq **end if**

While next==1 **do** // this is the first packet, current=seq2, next =current + packet.datalength

end while

if seq2>=next, next=seq2+Packet.datalength

else Write that packet to the temporary PCAP file

end while

Algorithm 3: Packet reordering

Input: Time Stamped PCAP file, Source and Destination IPs, Source and Destination ports

Output: Reordered PCAP file,

Initialize next=0, new_time_seq=0,

While packet!= null **do**, Read the packet from the PCAP file

if packet= EOF or null then exit from the loop

if packet.Source ip==source_ip and packet.sourceport=source port

if next.flag is set seq=Packet.time_seq, seq_relative=packet.time_seq-seq **end if**

end if

if next=0, Write reordered file, next=time_seq relative+packet.datalength **end if**

if seq_relative= next do following two steps, Write packet to the reordered file,

next=new_time_seq relative+packet.datalength **end if**

if seq relative >next, Read each packet from the temporary PCAP file

if packet.time_seq-seq=next, write the packet to the reordered file, next=next+packet.datalength

end if , **end if**

end while

C. Forensic Evidence Recording Unit

This component of the projected architecture will excavate forensic facts to confirm and pin down the malevolent VoIP users tangled in underhanded actions. This unit organizes the forensic information along with procuring actual evidences in

form of VoIP voice data between parties and produce a detailed criminal statement against parties of a particular communication session for their subsequent investigation. The criminal statement consists of forensic particulars such as IPs, ports, credentials of VoIP clients or parties and period of malicious act along with regenerated digital proofs. This

criminal statement is submitted to concerned cyber cell of a particular region to grab actual malefactors followed by their prosecution [14].

VI. RESULTS AND DISCUSSIONS

This segment furnishes the description of GUIs developed for forensic investigation of VoIP followed by the performance analysis of packet stream reassembly methodology with prevailing forensic structures. The projected innovative framework is entitled as VoIP Forensic Analyzer (VFA).

A. VOIP Control Panel

This GUI of control panel as shown in Fig 4 contains an option for loading any previous network sessions in the form of PCAP file. After stacking the PCAP file, it demands to specify the specific IP and ports used for the proxy server in VoIP analysis control panel. This data is essential in order to filter the required packets from the selected PACP file and identify the IP's and corresponding ports of the clients that have established a connection with the proxy server.

The ports used by the client and the server for communication are obtained by performing the port extraction procedure that we have explained in the VoIP session recreation unit. In selecting the "Filter Packets" button in the control panel, it starts the filtering process and displays the Client IPs and ports. Now the designed framework finds out the VoIP conversations that have been performed by the selected IPs and ports and will display the forensic results in various GUIs.

Projected framework has many GUI, which traces and displays forensic details in many steps. Here we outline and unveil GUIs that display very critical forensic detail so that the suspected VoIP communication stream can be recognized, the user can be tracked, and legal actions can be carried out with substantial evidences.

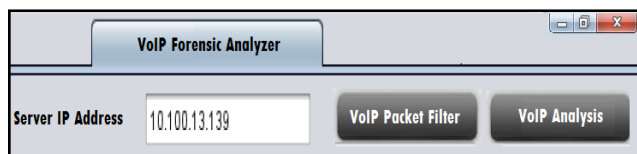


Fig. 4. GUI of VoIP Control Panel

B. Forensic Details from Header Panel

This GUI of header panel as shown in the Fig.5 displays forensic details, which gives an insight into handshake process that has been done for the available VoIP connections. This GUI encompasses the IPs and ports castoff by VoIP clients or parties along with "codec" used voice encoding. Header panel traces these forensic details from VoIP network packets collected and processed with the help of our proposed packet rescheduling and rebuilding algorithm.

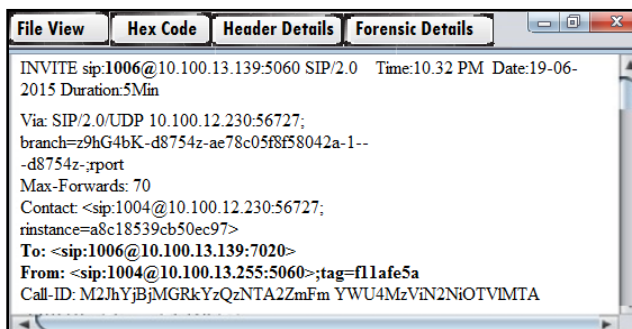


Fig. 5. GUI of Header panel

C. Hex Code Panel

This GUI of hex code panel as shown in Fig.6 provides the hex code of each byte in the data on the left side and its corresponding ASCII code on the right side for analysis.

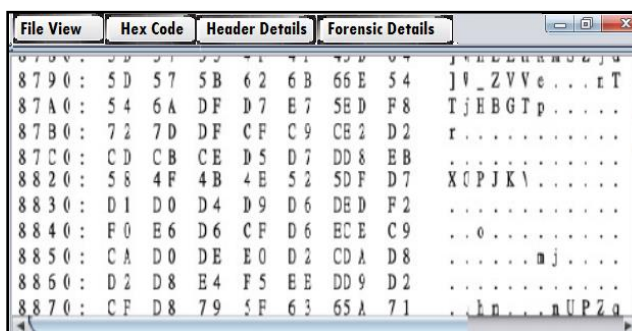


Fig. 6. GUI of Hex Code Panel

This panel decodes the VoIP communication session and display it in hex format that characterize the pattern of loaded PCAP file. This pattern can be further used with other pattern matching vulnerability detection framework or tools.

D. File View Panel

This GUI of file view panel as shown in Fig. 7 provides forensic information about the files reconstructed during the process of analysis. It also helps the investigator to know how many files (calls) are reconstructed to get an idea about the duration of each call with specific IP and port numbers

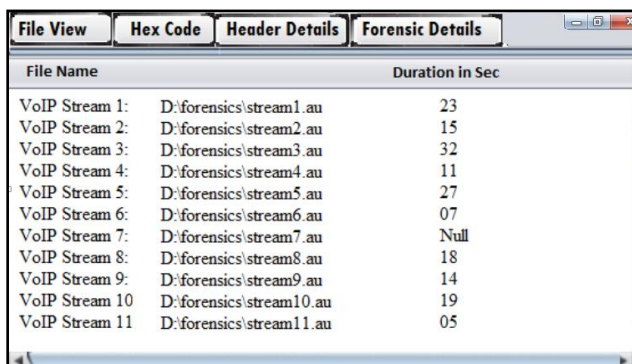


Fig. 7. GUI of File view panel

E. Forensic panel

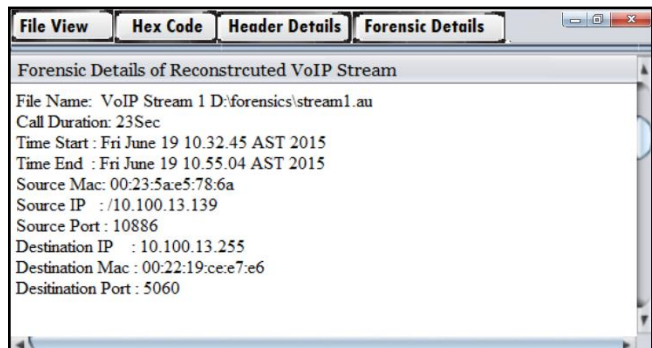


Fig. 8. GUI of Forensic panel

This GUI of Forensic panel as shown in Fig.8 provides an in depth forensic details of each of the VoIP call that framework has reconstructed. It contains the details about file name of the reconstructed file. Call duration in seconds. Starting time of the call, End time of the call, IP address, port numbers and Mac Ids of VoIP clients in a particular communication stream.

F. VoIP Forensic player

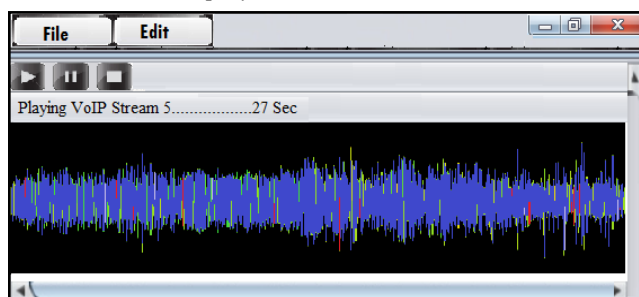


Fig. 9. GUI of VoIP Forensic player

This VoIP forensic player as shown in Fig.9 plays the actual voice communication reconstructed from the suspected VoIP network stream. Once particular VoIP stream is reconstructed, VoIP forensic player decodes the processed voice data with help of SIP's codec G.729. These voice data were already encoded using G.711. This is one of the critical milestone that we have achieved in our research work which provides substantial evidences to prosecute malicious users with their conversation.

G. Performance Analysis of Projected Forensic Analyzer

The significant outcome that proves efficiency and trustworthiness of RTP rescheduling and reconstruction technique of proposed framework are sketched in this section. Performance of projected framework is systematically compared and verified against Wire shark and Ethercap, which are general-purpose free access packet analysis tools. Even if these tools collect and reorder packet, they fail in replaying the VoIP voice data after reconstruction. Here we compare rescheduling and reconstruction time taken by projected framework with respect to above-mentioned tools using PCAP file with malicious VoIP streams up to two lakh packets as shown in Fig.10. The result produces better yield in

delivering commendable faster and exact reconstruction of VoIP streams. This is mainly due to managing retransmitted and duplicate packets by time stamping strategy in VoIP sessions, which fastens reconstructing and replaying VoIP sessions in various cases.

Projected architecture exhibits commendable quickness in regeneration actual VoIP communication stream from PCAP file and withstand large no of packets and its subsequent processing shows its reliability in tracing digital corruptions using VoIP platforms. We also tested the performance of proposed framework with and without using novel time stamping technique as shown in Fig. 11 with respect to the size of PCAP file against reconstruction time. When time stamping technique is devised, the session reconstruction time is drastically boosted by more than 60% of the reconstruction time taken by the technique without employing time stamping. Above saturation point (SP) as shown in graph, the performance of proposed framework provides poor reconstruction time in handling large PCAP file sizes without employing time stamping technique. Thus our novel time stamping technique critically is uplifting the performance of proposed reconstruction technique.

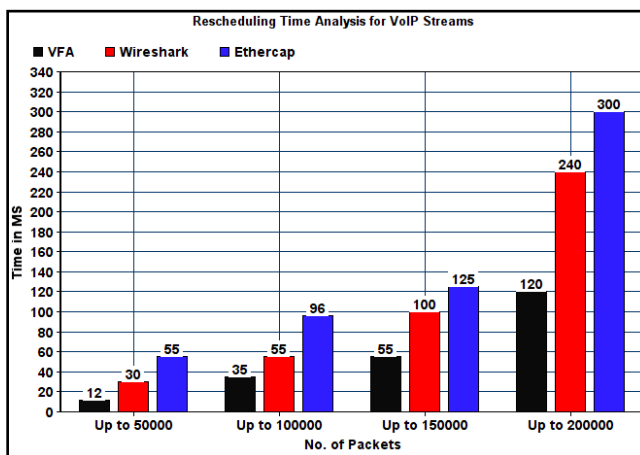


Fig. 10. Time analysis for VoIP reordering

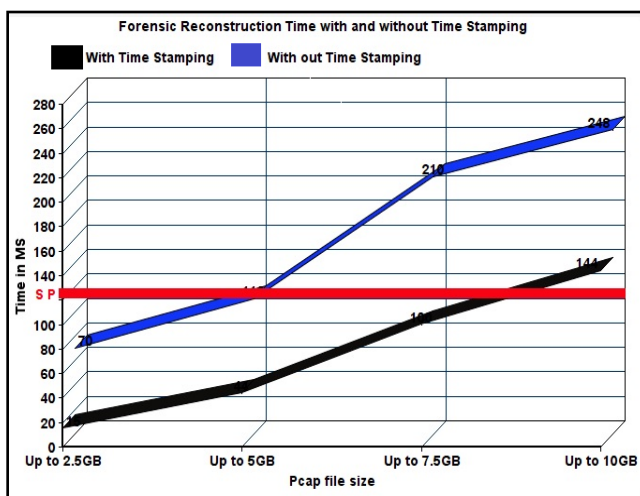


Fig. 11. Time analysis with and without time stamping technique

H. Brief Implementation details

This framework is developed with Java using JPCAP library [17] and Winpcap which are typical softwares used for data link layer processing. It permits bagging and dissecting network layer packets. Minimum configurations of 4 GB RAM and dual core level processor are preferred for processing PCAP files with 100000 packets. Since it uses Java and related subsidiaries, proposed work is thus platform independent.

VII. CONCLUSION

Network administrators and investigators use the projected architectural framework for retrospective forensic analysis of VoIP communication infrastructure. We have also outlined the novel technique for packet reconstruction for subsequent forensic analysis. This VoIP forensic analyzer effectively hints credentials of malicious VoIP users and craft environment for their subsequent prosecution. This software tool is easily deployable with Windows and Linux environments. While the framework outlined above is capable of detecting, identifying and reconstruction in a VoIP network, there remains a significant legal hurdle due to the cross-border nature of internet systems, especially international law needs to clear provision for the prosecution of malicious users worldwide. The proposed idea can be easily extended for forensic analysis of other internet protocols to make it support multiple protocols.

ACKNOWLEDGEMENTS

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Comparing the Usability of M-Business and M-Government Software in Saudi Arabia

A Revised Nielsen's Heuristics Method

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Abstract—This study presents a usability assessment of mobile presence in the Kingdom of Saudi Arabia (KSA), with a particular focus on the variance between M-business and M-government presence. In fact, a general hypothesis was developed that M-business software is more usable than M-government software, with eleven sub-hypotheses derived from Nielsen's heuristics method. To examine the hypotheses, a true representative sample of thirty-six (n=36) mobile software applications in Saudi Arabia were identified from prior research, representing two main categories: M-business and M-government. Within each category, eighteen (n=18) mobile software applications were carefully chosen for further evaluation, representing a wide variety of sectors. A questionnaire was devised based on Nielsen's heuristics method; this was tailored to fit the context at hand (mobile computing) to establish a usability checklist (consisting of eleven constructs). A group of thirty-six (n=36) participants were recruited to complete the usability assessment of examining each software application against the usability checklist, by rating each item using a Likert scale. The results herein reveal that mobile interactions in KSA were, in general, of an acceptable design quality with respect to usability aspects. The average percentage score for all heuristics met by the evaluated mobile software applications was 68.6%, this reflected how well the usability practices in mobile presence were implemented. The scores for all usability components exceeded 60%, with five components being below the average score (of 68.6%) and six components being above it. The variance between M-business and M-government software usability was significant, particularly in favor of M-business. In fact, the general hypothesis was accepted as well as seven other sub-hypotheses, as only four sub-hypotheses were rejected.

Keywords—Usability; interaction; heuristics; interface; mobile; Saudi Arabia

I. INTRODUCTION

The usability of software is an important aspect of systems development, however it is particularly challenging to study the usability of mobile systems as several limitations are posed by Mobile Computing (M-computing) environments [1]. In the current literature, scholarly research considers the proposition of design patterns, software quality and testing models for different mobile software applications [2]. Other streams of research have focused on investigating the feasibility of usability testing environments, frameworks, contextual effects and other aspects of mobile interaction. In addition, several studies investigated the usability of experimental mobile

software applications developed to solve context-specific problems. Few studies examined the status of mobile presence in the Kingdom of Saudi Arabia (KSA), with respect to the usability of mobile interfaces; thus, most of the research efforts need to examine the usability of mobile software in KSA using heuristics evaluations. This paper not only reveals a gap in the current literature, but it also attempts to overcome this somewhat. Consequently, this paper presents an empirical study of the usability of mobile software applications in KSA. This paper represents an extended version of an earlier conference paper [3]. In addition, it differs from the conference paper by examining the difference between M-business and M-government software usability in KSA.

The paper aims to assess the usability of a mobile presence, while also ranking each evaluated mobile software application. Specifically, it identifies several aspects of mobile software usability, based on the Nielsen's heuristics method, in order to enrich the understanding of mobile interaction design. Another aim of this study is to demonstrate the empirical application of Nielsen's heuristics method, in the context of mobile computing. It also sheds more light on the variance in usability between M-business and M-government software in KSA. In order to achieve the aims, eleven dimensions (n=11) were adopted from instruments proposed by prior research, all of this helped to design a questionnaire for assessing mobile software usability. The eleven usability components were based on the Nielsen's heuristics instrument that was published in 1994 [4]; however, it was tailored to fit the context of touch-based mobile interaction in accordance with the arguments presented by [5]. In order to compare the usability practice between M-business and M-government, the sample of mobile software applications was divided equally between the two conditions, each of which was represented by eighteen applications (n=18).

The remainder of this paper is organized as follows: the literature review is provided in Section 2, and more detail of the empirical study is presented in Section 3. The research results and discussions are presented in Sections 4 and 5 respectively and, finally, a conclusion is provided in Section 6.

II. LITERATURE REVIEW

Usability assessment has been a central topic of several empirical studies in the field of human computer interaction (HCI) [6]. In particular, testing of the usability of mobile

software is an emerging area of research, as several challenges may affect these software applications, including: bandwidth, screen size and the nature of the mobile environments [7]. For example, Zhang and Adipat (2005) proposed a framework for the design and implementation of mobile interface usability investigations, with regard to the selection of tools, methods, measurements and data collection approaches [7]. Another study by Kallio and Kaikkonen (2005) investigated mobile usability testing in two different environments: the laboratory and field testing; the findings indicated that laboratory-based usability testing was sufficient, apart from when there were contextual elements that needed investigation in terms of the actual use environment [1]. In addition, Ryan and Gonsalves (2005) empirically examined the effect that contextual elements had on mobile interaction design, they implemented four different conditions: mobile or personal computer interaction design, against web-based or traditional interfaces, all of which were tested in a with-participants experimental design [8]. The usability results for mobile interaction outweighed those for personal computer interaction, due to the unique characteristics of mobile computing, such as location-based services. Yet, some limitations were encountered, such as limited input capabilities of the actual mobile devices [8]. Other studies considered the holistic view of mobile usability; to illustrate, Coursaris and Kim (2006) proposed a framework for usability evaluation, this framework offered a holistic view of mobile usability dimensions which could form the foundation to guide further research in usability [9]. Another study, by Tsai et al. (2007), developed a mobile software application for caloric self-monitoring in a real-time manner; they conducted a usability study to examine the feasibility, compliance and satisfaction of the software application – their findings revealed positive feasibility and usability results [10]. Furthermore, Balagtas-Fernandez and Hussmann (2009) proposed a four-phase framework for mobile software usability evaluation, which aimed at simplifying the technical usability analysis [11], using the following four phases: preparing the system for analysis, collecting usability data, extracting information and analyzing usability practices.

Within the usability research endeavor, heuristics evaluation has been subject to several improvement attempts, resulting in the proposition of different variations of context-specific usability heuristics. For instance, representing the contextual effect of mobile environments, a study by Po et al. (2004) proposed two variations of heuristics evaluation, namely heuristic walkthrough and contextual walkthrough [12]. The former incorporates scenarios of use into the usability assessment by heuristics evaluation, while the latter involves the application of the former in the field [12]. This study demonstrated that usability assessment could be improved by incorporating contextual details, particularly in favor of the heuristics walkthrough method [12]. In addition, Korhonen and Koivisto (2006) proposed context-specific usability heuristics for mobile games, incorporating game usability, mobility and game play dimensions. The findings indicate that it is easy to discover playability flaws, but their counterparts within game play were identified as harder to uncover [13].

In KSA, several usability studies utilized the heuristics evaluation method to assess different information systems. It is

noteworthy that KSA is considered to be one of the biggest information and communications technology (ICT) markets in the region. According to the Communications and Information Technology Commission's (CITC) annual report, it is estimated that ICT spending in KSA reached SAR 94 billion in 2012, with an annual growth of 16% [14]. Considering the E-government web presence in KSA, Eidaroos et al. (2009) adapted the heuristics evaluation method to produce a usability checklist for E-government websites, so as to discover several usability flaws in the current practice of E-government interaction design [15]. Furthermore, Al-Khalifa (2010) exploited the heuristics evaluation method in order to assess the usability of E-government web presence in KSA; it was found that the heuristics evaluation method was useful as an initial phase to uncover usability issues [16]. Specifically, a sample of fourteen (n=14) governmental websites were selected from different sectors and considered for further evaluation. Using six usability components the sampled websites were evaluated, it was demonstrated that they had good usability implementation as all components achieved a score above 50%. More recently, Alotaibi (2013) conducted an investigation of the usability of university web presence in KSA, using heuristic evaluation to demonstrate its empirical application in KSA [17]. The study evaluated twelve (n=12) university websites from different categories, thirty (n=30) evaluators found that the usability practices in Saudi universities were demonstrated to be of an acceptable quality with regard to seven usability components.

III. EMPIRICAL STUDY

Heuristics evaluation is considered to be among the most appropriate usability evaluation method, with regard to initial assessments of an interface, because it is low cost, quick to set up and achieves fast outcomes [16]. Heuristics evaluation was first proposed in 1990 by Nielsen and Molich [18], and has since been refined by many authors. To illustrate, Inostroza et al. [5] adapted the Nielsen heuristics approach to fit the context of mobile computing. Specifically, they adopted Nielsen's approach, which consisted of ten (n=10) dimensions, and made some modifications to the heuristics to enable use within the context of touch-based mobile devices. In addition, a newly added dimension, concerned with physical interaction and ergonomics, has been incorporated to describe alignment between the shape, position and dimensions of the interface elements with the natural posture/position of the hand.

This research proposes an assessment method that is based on Nielsen's heuristics [4], but it has been extended and tailored to fit within the context of touch-based mobile interaction [5]. The proposed method incorporated eleven usability components (UC), each of which consisted of several heuristics. In order to guide software application evaluation, a questionnaire was devised which represented all usability components and heuristics. Each heuristic was evaluated by rating its status using a Likert scale ranging from 1 to 5 (1 – not met and 5 – fully met), in terms of whether it was met by the evaluated mobile software application. Table 1 shows the usability components and the number of heuristics identified.

The sampling of mobile software applications for further assessment involves selecting a few software applications that

truly represent the mobile software population within an area, in this case within KSA. Sampling should reflect a balanced diversity of mobile software applications; thus, the sampled software applications should represent different sectors, activities and organization sizes. In 2014, Alotaibi surveyed all mobile software applications that were developed by organizations in KSA, he identified a sample of thirty-six (n=36) mobile software applications – this should be regarded as a true representative sample of the population. Based on that argument [19], the thirty-six mobile software applications were considered for further analysis as a sample of the population; this sample can be viewed in Appendix A.

Scholarly research has considered evaluation of the adoption of emerging technologies whether in the Saudi private or public sectors. For example, a study by Alsenaidy and Ahmad (2012), on one hand, assessed the status of M-government in KSA by reviewing several mobile services offered by government organizations. [20]. On the other hand, a recent study by [21] investigated the adoption of mobile technology in the Saudi private sector, particularly the banking industry, and provided solid evidence for the maturity of M-business practices. Other researchers reviewed different aspects of E-service adoption in the Saudi private and public sectors, such as E-shopping [22], E-government [23-25] and online retailing [26]. Another stream of research considered the adoption of mobile technologies in education, which can be offered by both government owned and private universities [27-30]. In fact, few of these studies considered examining the difference between M-business and M-government practices. Therefore, the current study separated M-business and M-government software in order to facilitate a comparison between the two categories.

In particular, the sampled mobile software applications were selected to represent two main categories: M-business and M-government, with eighteen software applications in each category. Based on the findings of prior research that M-business was more progressive than M-government in KSA [19], a general hypothesis was developed and put forward with eleven sub-hypotheses (derived from Nielsen's heuristics method [4]) to examine the different aspects of mobile software usability; to illustrate:

H1: The usability of M-business software will outweigh that for M-government software, particularly in terms of the following usability components:

- H1(a) Visibility of system status.*
- H1(b) Match between system and the real world.*
- H1(c) User control and freedom.*
- H1(d) Error prevention.*
- H1(e) Minimize the user's memory load.*
- H1(f) Consistency and standards.*
- H1(g) Customization and shortcuts.*

TABLE I. USABILITY COMPONENTS AND THE NUMBER HEURISTICS (ADAPTED FROM [4, 5])

Code	Usability Component	Number of Heuristics
UC1	Visibility of system status	9
UC2	Match between system and the real world	6
UC3	User control and freedom	4
UC4	Error prevention	5
UC5	Minimize the user's memory load	3
UC6	Consistency and standards	3
UC7	Customization and shortcuts	3
UC8	Aesthetic and minimalist design	2
UC9	Help users to handle errors	3
UC10	Help and documentation	2
UC11	Physical interaction and ergonomics	3

- H1(h) Aesthetic and minimalist design.*
- H1(i) Help users to handle errors.*
- H1(j) Help and documentation.*
- H1(k) Physical interaction and ergonomics.*

Using the questionnaire, the sampled mobile software applications were evaluated by thirty-six (n=36) participants, all of which were regular mobile users. A balanced recruitment procedure was followed in order to control the effect of gender, knowledge and education of the participants. Firstly, the sample participants were split against gender with 50% being female and 50% being male. Within the eighteen participants for each gender group, three further sub-groups were created to represent educational background; consequently, there were six bachelor students, six master students and six practitioners. This resulted in six sub-groups (each representative of 16.67% of the sample size) whereby the delegates were equally balanced between gender, knowledge and education.

A within-participants design [31] was adopted for this research; all participants were instructed to interact with and then evaluate all of the mobile software applications. The order of the mobile software application evaluation was counter balanced among the participants in order to alleviate any participant learning effects. An experimental procedure was deployed to facilitate the evaluation. Fifteen-minute lectures and short (thirty-minute) training sessions were also introduced to describe the basics of the heuristic method and to demonstrate the way in which the method could be used to evaluate mobile software applications.

IV. RESULTS

Fig. 1 shows the percentage values of the usability assessment for the mobile software applications in accordance with the overall usability score and the eleven usability components (UC). At a glance, it is clear that more than 60%

of the usability heuristics for mobile interaction design were met by the mobile presence in KSA. Although not all of the mobile usability heuristics were met, the figures clearly indicate that the usability design of the mobile software applications were of an acceptable quality. In particular, the score for visibility of the system status (UC1) was 70%, with all heuristics being rated with average scores, to illustrate there were no extreme values in the evaluation. As for UC2 (match between the system and real world), it can be seen that 72.2% of the heuristics were met, with two heuristics being rated consistently above average (displaying information in a logical and natural order and following real-world conventions). Furthermore, the two components of UC3 (user control and freedom) and UC4 (error prevention) were determined to be lower than the average score of 68.6%, scoring 68.0% and 63.4%, respectively. These results could be attributed to two main reasons: the lack of undo and redo options and the fact that the users were not warned when the error was likely to occur. In terms of the other usability components, UC5 (minimize the user's memory load) had the same value as UC1, with a 70% score. Further analysis of the raw data revealed that all heuristics in these components were rated with average scores (no extreme scores). In fact, further analysis of the results showed that UC6 (consistency and standards) achieved the highest score with 73.5%. This could be attributed to the way in which the standards were set within the mobile interaction design, whereby the concepts were consistently expressed, and established conventions were also followed. In addition, UC7 (customization and shortcuts) achieved the lowest score at 60.3%, due to a lack of simple configuration options and other features, such as: setting shortcuts and customizing, and grouping the interface elements. As for the usability component UC8 (aesthetic and minimalist design), it scored higher than the average score for all usability components, scoring 69.4%. This could be attributed to the richness and appropriateness of the displayed contents. Moreover, the two components of UC9 (help users to handle errors) and UC10 (help and documentation) were evaluated to be lower than the average, scoring 63.8% and 62.4%, respectively. This could be attributed to two main reasons: firstly, the occurrence of errors was not precisely indicated, and secondly no solutions for the errors were suggested in the error messages. Some mobile software applications provide help and documentation features that are not primarily easy to find; whereas some other software applications lack these features. Finally, the usability component UC11 (physical interaction and ergonomics) scored higher than the average score for all usability components, scoring 71.4%. This could be attributed to the appropriateness of the buttons' shapes, positions and functions. In summary, the assessment of mobile software applications in KSA were found to be above average (with an average score of 68.6%), which reflects how well the usability practices in mobile presence were implemented. Furthermore, Fig. 1 clearly documents that five of the usability components scored lower than the average score of all of the heuristics (total score= 68.6%), while five components scored above average.

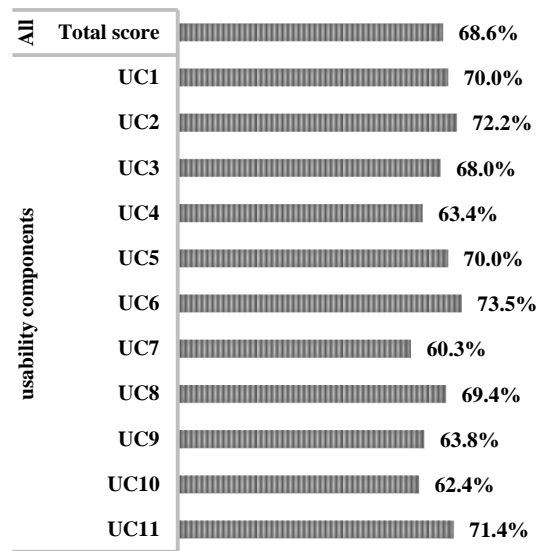


Fig. 1. Percentage values of total score for the eleven usability components (UC) for mobile software applications

Fig. 2 shows the percentage values of usability scores for the M-government and M-business software in accordance with the overall usability score and the eleven usability components (UC). In general, it can be seen from the figure that the total usability score for M-business was approximately 7% higher than that for M-government. Similarly, the M-business scores were found to be greater than for M-government in all aspects of individual usability components, except UC10 (help and documentation) and UC6 (consistency and standards), with a difference percentage ranging from 0.4-4.2%. In particular, the score in UC1 (visibility of the system status) for M-business was 4.5% higher than for M-government, while the score in UC2 (match between the system and real world) for M-business was 3.6% greater than that for M-government. Likewise, the score in UC3 (user control and freedom) for M-business was slightly higher than that for M-government. However, the score in UC4 (error prevention) showed a different picture, where the difference between M-business and M-government was found to be minimal. Similarly, the difference in UC5 (minimize the user's memory load) was found to be slight. The score in UC6 (consistency and standards) for M-business was 3.6% higher than for M-government. Furthermore, the score in UC7 (customization and shortcuts) for M-business was 4.2% higher than that for M-government. Moreover, the difference between M-business and M-government was found to be minimal with regard to UC8 (aesthetic and minimalist design) and UC10 (help and documentation). Finally, the score in UC11 (physical interaction and ergonomics) for M-business was 2.9% greater than that for M-government. In summary, the M-business usability scores were shown to be consistently higher than those for M-government, except for three components where minor differences were found.

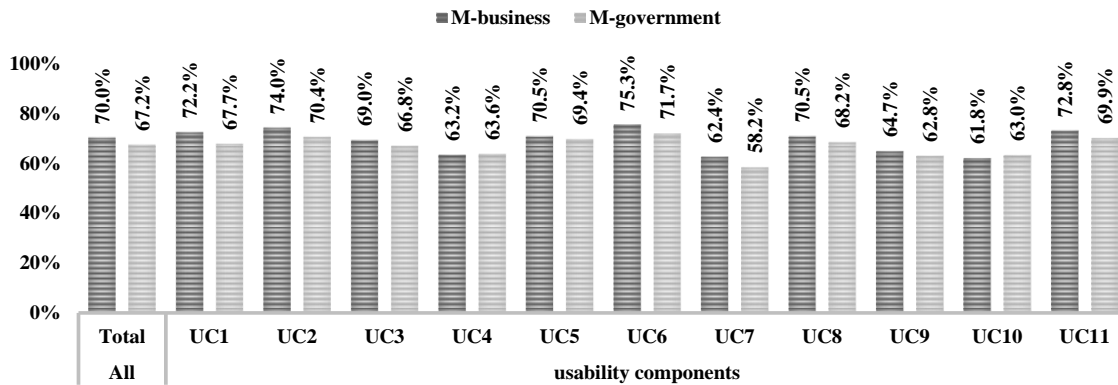


Fig. 2. Percentage values of usability scores for M-government and M-business software in accordance with the overall usability score and the eleven usability components

V. DISCUSSION

The variation in usability scores between M-business and M-government software was statistically examined using t-test. Since the experimental design was within-participants [31], in which all participants were exposed to all conditions, the appropriate test for examining the difference between groups was the paired t-test [31]. The degree of freedom was found to be six hundred and forty-seven ($df=647$) which represents one less than the number of pairs. With the critical significance levels (α : alpha) of $\alpha = 0.05$ and $\alpha = 0.01$, the critical values (cv) of the paired t-test were $cv = 2.021$ and $cv = 2.704$, respectively. Overall, the difference in usability between M-business and M-government software was statistically significant ($t_{647} = 3.323$, $cv = 2.021$, $p < 0.01$). Therefore, it can be said that the general hypothesis (H1) was accepted at 99% confidence level.

Table 2 reviews the paired t-test result in accordance with the eleven usability components. At a glance, it can be seen from the table that seven tests out of eleven achieved significant results at different confidence levels. In fact, the difference in the visibility of the system status (UC1) between M-business and M-government software was found to be statistically significant at 99% confidence level. The same picture was shown in the match between the system and the real world (UC2). In addition, the difference in user control and freedom (UC3) between M-business and M-government software achieved statistical significance at 95% confidence level. However, the variance in error prevention (UC4) between M-business and M-government software did not reach any significance level. Similarly, neither did the fifth usability component (minimize the user's memory load). In contrast, the variance in consistency and standards (UC6) between the two software categories was found to be statistically significant at 99% confidence level. The same picture was shown in customization and shortcuts (UC7). Moreover, the difference in the aesthetic and minimalist design (UC8) between the two conditions of software achieved a statistical significance at 95% confidence level. On the contrary, the variance in helping users to handle errors between the two categories did not reach

any statistical significance. Similarly, the tenth usability component (help and documentation) revealed insignificant results. Finally, the variance in physical interaction and ergonomics (UC11) between the two conditions reached a significant result at a 99% confidence level. In summary, all aspects of usability assessment achieved a statistical significance, except for UC4, UC5, UC9 and UC10. This asserts the rejection of sub-hypotheses $H1(d)$, $H1(e)$, $H1(i)$ and $H1(j)$, as well as acceptance of the sub-hypotheses $H1(a)$, $H1(b)$, $H1(c)$, $H1(f)$, $H1(g)$, $H1(h)$ and $H1(k)$. In summary, the sub-hypotheses were partially accepted, as most of the associated tests (seven out of eleven) showed statistically significant results.

The implications of this study can be seen as being of great importance for the theory and practice of M-computing in KSA. As for the implications for academia, it is important for researchers in the M-computing field to consider the gap between M-business and M-government usability practices. The study provided an insight into the status of mobile interaction design in KSA, with a particular focus on the difference between M-business and M-government software usability. Further research in this field is yet important to highlight the critical success factors of mobile interaction design. Another effect of this study is the validation of Nielsen's heuristics method in the context of the M-computing field. During the course of the experiment, it was noticed that the usability checklist was developed for Windows-based software, but was fit in the M-computing context. However, the usability checklist needs further improvement to match the mobile HCI requirements. It is rather important to develop a usability checklist for touch-based mobile software, particularly with regard to physical interaction and ergonomics. In terms of practical implications, managers in ICT departments should double efforts to enhance mobile software, with a particular focus on the usability of the user interface. Experience gained from this study suggested that mobile software developers and interface designers should provide the necessary features to help users to handle errors. Alternatively, they should design their interface and develop the software to prevent the occurrence of errors in the first place.

TABLE II. REVIEW OF PAIRED T-TEST RESULTS IN ACCORDANCE WITH THE 11 USABILITY COMPONENTS ALONG WITH THE CONFIDENCE LEVEL (CL)

Sub-hypothesis / Usability component	t_{647} value	P value	Significant t?		CL
			yes	no	
H1(a)/ Visibility of system status	4.53	p<0.01	√		99%
H1(b)/ Match between system and the real world	3.62	p<0.01	√		99%
H1(c)/ User control and freedom	2.29	p<0.05	√		95%
H1(d)/ Error prevention	-0.31	p>0.05		√	---
H1(e)/ Minimize the user's memory load	1.26	p>0.05		√	---
H1(f)/ Consistency & standards	3.63	p<0.01	√		99%
H1(g)/ Customization & shortcuts	3.81	p<0.01	√		99%
H1(h)/ Aesthetic & minimalist design	2.07	p<0.05	√		95%
H1(i)/ Help users to handle errors	1.59	p>0.05		√	---
H1(j)/ Help & documentation	-1.02	p>0.05		√	---
H1(k)/ Physical interaction and ergonomics	2.83	p<0.01	√		99%

This can be achieved by avoiding misplacement of the control button and implementing customization and shortcut features. It is also recommended to adhere to interface design standards in the mobile computing field and follow mobile usability guidelines.

Although, this study was useful for academics and professionals in the M-computing field, it encountered several limitations. First, the study examined mobile software usability in a domestic context, representing only one country. The idea could be extended beyond the boundary of KSA to cover the usability of mobile software in neighboring countries, or rather overseas. It is important to link the status of mobile software usability in KSA with usability practices at the regional and international level. For example, a set of mobile software could be selected from regional or international key players in the M-computing field and considered for further evaluation, in order to compare the status of Saudi mobile interaction design against such benchmarks. Second, it is important to consider that the M-computing field is rapidly evolving and its software applications regularly change. Therefore, replication of this study on a regular basis would improve the understanding of software usability and its improvement and evolution. Finally, this study adopted a revised version of Nielsen's heuristics instrument [4], which was proposed originally to examine the usability of Windows-based software systems. This instrument was of a generic nature while assessment of the usability of M-computing software required the development of a usability model for M-computing research. It was rather essential to rely on a well-known instrument to investigate and establish an initial understanding of the issue at hand, as recommended by Al-Khalifa (2010): the use of a heuristics evaluation is useful as an initial phase to uncover usability issues [16]. However, developing a context-specific model for M-computing software usability merits further investigation.

VI. CONCLUSION

This paper has provided an empirical study of the usability of mobile software applications in KSA, using the heuristics evaluation method. Based on prior research, eleven dimensions

(n=11) were considered and a questionnaire was designed specifically for this study. The questionnaire items and constructs were derived from Nielsen's heuristics instrument, they were then adapted to fit the context of mobile environments. A sample of thirty-six (n=36) mobile software applications were carefully chosen based on prior research, representing two main categories: M-business and M-government. A general hypothesis was developed that the usability of M-business software would outweigh that for M-government, with eleven sub-hypotheses reflecting the usability components of Nielsen's method. Using the questionnaire, the sampled mobile software applications were evaluated by thirty-six (n=36) evaluators, using a within-participants experimental design, and whereby each participant evaluated all software applications. The order of the mobile software applications evaluation was counter balanced among the evaluators in order to alleviate any possible bias. The results indicated that the usability designs of mobile software applications in KSA were an acceptable quality. In fact, an average percentage score of 68.6% for all mobile software applications was evaluated and determined by the heuristics; thus, indicating the extent to which usability practices in mobile presence were implemented. The scores for all usability components exceeded 60%, with five components being below the average score and six components being above it. The usability of M-business software was shown to be greater than for M-government and, therefore, the general hypothesis was accepted. On the other hand, four sub-tests related to the usability components (UC4, UC5, UC9 and UC10) failed to achieve any statistical significance, and therefore the four corresponding sub-hypotheses were rejected.

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APPDENIX A

TABLE III. PERCENTAGE VALUES OF THE OVERALL USABILITY SCORE AND THE RANK FOR EACH EVALUATED MOBILE APPLICATION

Sequence	Category	Mobile Application	Usability Score	Application Rank
1	M-government	Ministry of Commerce and Industry (MCI)	72%	22
2		Ministry of Foreign Affairs (MFA)	76%	14
3		Ministry of Health (MOH)	79%	10
4		Ministry of Education (MOE)	80%	8
5		Ministry of Labor (MOL)	67%	32
6		King Fahd University of Petroleum & Minerals (KFUPM)	78%	12
7		Taibah University	66%	33
8		Saudi Food & Drug Authority (Saudi FDA)	67%	31
9		Saudi Arabian General investment Authority (SAGIA)	71%	23
10		King Fahad Medical City (KFMC)	73%	21
11		Saudi Post Corporation	68%	29
12		Saline Water Conversion Corporation (SWWCC)	73%	20
13		National Information center (NIC)	79%	9
14		Riyadh Chamber of Commerce & Industry	74%	16
15		Saudi Arabian Airlines (Saudia)	62%	34
16		Saudi e-Government (Yesser)	70%	24
17		General Directorate of Civil Defense	68%	28
18		Saudi Press Agency (SPA)	79%	11
19	M-business	Al-Rajhi Bank	88%	2
20		Albilad Bank	84%	3
21		Tadawul	70%	26
22		Saudi Electricity Company (SEC)	74%	17
23		Saudi Automotive Services Co. (SASCO)	67%	30
24		E-Mall	59%	35
25		eXtra Stores	90%	1
26		Othaim Markets	69%	27
27		Jarir Bookstore	58%	36
28		Panda	81%	5
29		Saudi Company for Hardware (SACO)	74%	19
30		Mobily for Individuals	83%	4
31		Saudi Telecom Company (STC)	81%	6
32		Al-Jazirah newspaper	70%	25
33		Al-Riyadh newspaper	77%	13
34		Aleqtesadiah newspaper	81%	7
35		Al Tayyar Travel	76%	15
36		Solidarity Saudi Takaful	74%	18

Detection of Malware and Malicious Executables Using E-Birch Algorithm

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Abstract—Malware detection is one of the challenges to the modern computing world. Web mining is the subset of data mining used to provide solutions for complex problems. Web intelligence is the new hope for the field of computer science to bring solution for the malware detection. Web mining is the method of web intelligence to make web as an intelligent tool to combat malware and phishing websites. Generally, malware is injected through websites into the user system and modifies the executable file and paralyze the whole activity of the system. Antivirus application utilizes the data mining technique to find the malware in the web. There is a need of heuristic approach to solve the malware problem. Dynamic analysis methods yield better result than the static methods. Data mining is the best option for the dynamic analysis of malware or malicious program. The purpose of the research is to apply the enhanced Birch algorithm to find the malware and modified executables of Windows and Android operating system.

Keywords—Birch; Malware; Executables; Android and Windows

I. INTRODUCTION

Web Intelligence is the new field of computer science. There is a need of protective environment for users in the web. Users are isolated in web and web criminals are grab their attention towards them. Hacking websites, software and Internet accounts, phishing, malwares are the issues remains unsolved in internet. Arrival of web Mining (WM) gave a new definition for internet security; even famous software firms started using WM to find malware and malicious websites in Internet. Web Intelligence is the shield for internet users to protect them from the criminals exists in the web. The following applications are very useful for the internet communities [1][2]. Malware study the system and send the details to the Malware provider and that details will be useful for them to hack the system. Figure 1 shows the application of web mining and finding malicious website is one prime application of it.

A. Find Malicious website

WM methods are more intelligent to find the websites offering malicious software and deposits harmful files in the visitor system. The nature of ordinary websites differs from the malicious websites and WM easily identifies the difference and inform users about the site.

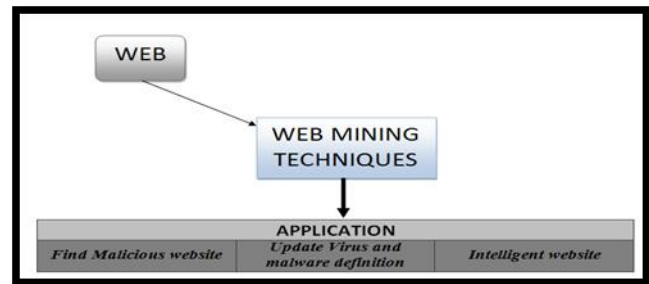


Fig. 1. Application of Web Mining

B. Update Virus and malware definition

It is easy to develop a database to store the virus and malware information. WM identifies newly arrived viruses in internet and stores automatically into the database. User can utilize the updated information to avoid storing harmful information into the system.

C. Intelligent website

Machine learning methods are used to build robots and can be used to design a website to become more intelligent to find their clones exist in internet. It has the ability to eradicate the malicious websites targeting their visitors in web.

Signature and Behavior based approaches are widely used in malware detection. The signature based approach is not effective because of the nature of detecting the malware but the behavior based approach is more effective in the process of detecting the malware and malicious executables. The purpose of research is to build an effective algorithm to detect the malware and malicious executables of operating system. The following sections will provide the data collection, design of algorithm and results and discussion.

Section II discuss the relevant research and its result and section III gives detail about the data collection for the study and Section IV discuss the experimentation and the result of the proposed research and finally section V gives the conclusion and future scope of the research.

II. REVIEW OF LITERATURE

Komashinskiy.D and Kotenko[1] have proposed a research on malware detection based on positionally

dependent features. The research considers the specificities of object's file format of potential malware containers. It also describes the realization and investigation of the common methodology for design of data mining based malware detectors using positionally dependent static information. Muazzam Siddiqui et.al.[2], proposed a work presents a novel idea of extracting variable length instruction sequences that can identify worms from clean programs using data mining techniques. The research deployed tree based classifiers including decision tree, bagging and random forest. Mohammad M. Masud et. al.[3], proposed a research on cloud based malware detection for evolving data streams. The research proposed a multi partition ensemble classifier in which a collection of classifiers trained with fold partitioning of the data, yielding an ensemble of classifiers. The work also proposed a feature extraction and selection technique for data streams that do not have any fixed feature set. J.k.kolter et.al[4], have done a research in classifying malicious executables in the wild using machine learning methods. The work has gathered malicious executables and encoded each as a training example using n-grams of byte code as features. They have evaluated the methods classified executables based on the function of their payload. The work could be used as the basis for an operational system for detecting previously undiscovered malicious executables. J.Dai et.al.[5], have presented a novel approach to detect unknown virus using dynamic instruction sequences mining techniques. They have built a program monitor which is able to capture runtime instruction sequences of an arbitrary program. The monitor utilizes the derived classification model to make an intelligent guess based on the information extracted from instruction sequences to decide whether the tested program is benign or malicious.

III. DATA COLLECTION

Identification of malware is a difficult task, many commercial websites failed to detect it. Some websites are offering malware executables for the purpose of research[6][7][8]. 250 malwares were downloaded from www.vxnetlux.org, www.29a.net and www.vxheaven.org. 25 infected executables of windows 7 and 10 infected executables of Android 4.4.2 were collected and included in the research to investigate the way of injecting malwares to infect the executables of the operating system. Numerical values were assigned to collected malwares and executables to reduce the complex nature and improve the performance of Birch algorithm[9][10][11].

IV. BIRCH ALGORITHM

Balanced iterative reducing and clustering using hierarchical (BIRCH) is an efficient clustering algorithm for large data set. It is a type of hierarchical clustering uses a clustering feature (CF) tree to reduce the memory size of the algorithm. The CF – tree uses 3 parameters (N, L S,SS): N – Number of objects in the cluster ; LS – Linear sum of data points; SS – Square sum of data points; The following formula use to evaluate LS and SS

$$LS = \sum_{p_i \in N} \bar{p}_i$$

$$SS = \sum_{p_i \in N} |\bar{p}_i|^2$$

CF tree is a multi-level compression of data that has the inherent clustering structure of the data. The limitation of main memory minimizes the time complexity of the data[12][13][14]. Birch algorithm is sensitive to the order of data, lacks in performance to cluster the non – numeric data. Generally it works well in spherical shape clusters but does not perform well for other shapes [15][16][17].

V. RESULTS AND DISCUSSION

Birch algorithm performance is limited for certain circumstances and it is not possible to produce better results. The CF tree parameters have been modified to two parameters and the value of LS is neglected for the study of malware and malicious executables[19][20][21][22]. The CF tree structured as an image to reduce the size and the total time of the algorithm[23][24][25].

The figure 2 shows the comparison of Birch and Enhanced Birch performance on 250 malware detection[26][27][28]. Initially, the performance of both algorithms was same but the Enhanced Birch shown better result in the later part of the malware[29][30][31].

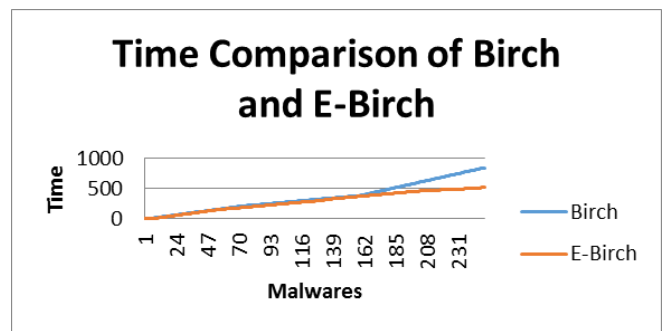


Fig. 2. Detection of Malware

The figure 3 shows the performance of the 25 executables of windows operating system. Both algorithms generate the optimum result with the mild difference in the execution time.

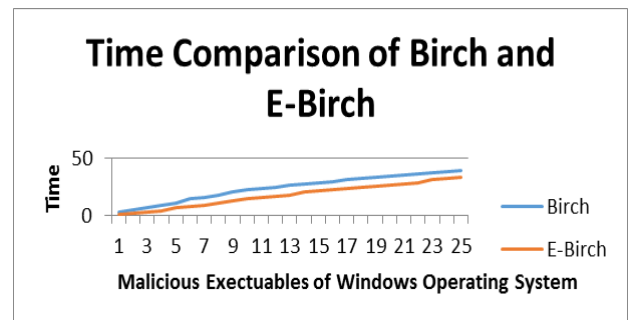


Fig. 3. Detection of Windows Malicious Executables

The figure 4 shows the performance of algorithm on 10 executables of Android operating system. The results are optimum but enhanced birch algorithm execution time is better than the Birch algorithm

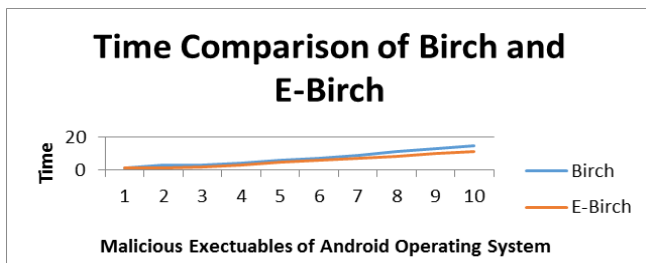


Fig. 4. Detection of Android Malicious Executables

TABLE I. EXECUTION TIME OF ALGORITHMS ON MALWARE AND MALICIOUS EXECUTABLES

Algorithm	Time to detect Malware (seconds)	Time to detect Malicious Executables of Windows operating system (seconds)	Time to detect Malicious Executables of Android operating system (seconds)
Birch	841	39	15
E-Birch	519	33	11

VI. CONCLUSION

The study proved that enhancement of BIRCH generates the accurate result with less memory and computation time. Dynamic analysis of malware detection will also generate the optimum results as the algorithm has learnt the behavior of malware and malicious executables. Malwares of Windows and Android were deployed in BIRCH AND E-BIRCH and found that E-BIRCH overall performance is better than the BIRCH. The future scope of the study is to design a hybrid application of signature and behavior based malware detection by implementing E-BIRCH.

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Formalization of Learning Patterns Through SNKA

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Abstract—The Learning patterns found among the learners community is steadily progressing towards the digitalized world. The learning patterns arise from acquiring and sharing knowledge. More impact is found on the usage of knowledge sharing tools such as facebook, linkedin, weblogs, etc that are dominating the traditional means of learning. Since the knowledge patterns acquired through web unstructured data is insecure, it leads to poor decision making or decision making without a root cause. These acquired patterns are also shared to others which indirectly affect the trust patterns between users. In this paper, In order to streamline the knowledge acquisition patterns and their sharing means a new framework is defined as Social Networking based Knowledge Acquisition (SNKA) to formalize the observed data and the Dynamic Itemset Count (DIC) algorithm is tried for predicting the users about the usage of web content before and after the knowledge is acquired. Finally the rough idea in building a tool is also suggested.

Keywords—Data Acquiring methods; Learning Patterns; Knowledge Management; Data Mining Tools

I. INTRODUCTION

The student's role in utilizing the web enabled systems has dominated their learning patterns in a day today life. Its growth rate is inevitable. In the process of gaining an updated knowledge in a faster mode, the social networking platform is used. This platform provides an easy structure to share information to anybody, at anyplace and at anytime. Even though, there are various educational tools and Environments, students prefer to learn through Digital web pages, web spaces and blogs.[7] The tremendous growth of information in blogs and other unauthorized web spaces leads to acquiring wrong information or unbiased information. These unstructured data becomes a real challenge for all data analyzers. the framework for acquiring specific valid data from various sources becomes the most important problem for all learners.

Most of the generic online users look into the social networking for finding friends of like-minded or domain-oriented or casual-time pass nature; to share their ideas, experiences and knowledge to others in turn they start learning or acquire knowledge indirectly. Still there is an issue of absoluteness and correctness in the acquired data.

This paper discusses about the social networking phases and their utilization among online learners. The facts behind the sharing of one's own ideas or forwarding the others post should be verified for their originality and trueness of the post. The Second part depicts the knowledge management frame work which is newly defined as Social Networking based Knowledge Acquisition (SNKA)[51] to formalize the observed data and the third part briefs about the supportive

data mining algorithms which indirectly supports the last section describing the SNKA implementation through Dynamic Item set Counting algorithm.

II. SOCIAL NETWORKING BASED LEARNING PATTERNS

The Primary platform for the information and communication sharing is having a higher growth because the social networking tools. Hence, Social networking is defined as a virtual platform for people to interconnect and share among themselves. The users group of social networking is broadly categorized as known and unknown. The known group here depicts the known friends, friends of another friend and relatives in relationship whereas unknown group depicts the strangers who are either popularly known by all and people who we meet while walking through our life.[1][2] Social Networking is revolving around the world with different faces. Social Networking usage creates knowledge, shares information, explores data and finally depicts as a decision maker for many crucial factors. At a learner's juncture, this paper helps us to visualize the knowledge acquired through social networking and their analyzing efficiency towards effective decision making.

The reasons for making social networking more popular are: i) most attractive and faster communication source as it bridges a distance of less cost. Ii) All the posts quickly get response either through a Like or a comment. Iii) reduce tension at various levels of problems where one to one or one to many communication is regretted.

Apart from the above, the usage of social networking as a platform leads to gaining advantage with the factors like: i) Faster publishing ii) Quicker Impact and response, iii) Trust Building through originality and validity of the content posted among users, iv) Faster Publicity and Popularity and v) Quicker Reusability options.

A questionnaire was created as part of the study and circulated among various learners irrespective of age factor. The questionnaire was created as a online blog and also circulated manually. Table 1 describes the resultant of the questionnaire. The online filled questionnaire was overwhelming from the age group of 9 - 45 whereas none were there between 0 - 8 and 46 - 73. The traditional and manual filled forms had less impact and responses. Table2 depicts the impact of the students over their learning patterns dependent more on the digital web pages rather than the traditional learning patterns.

In order to prove the domination of social networking based learning six hypotheses were generated and studied.

They are:

- i) To Analyze the users background information for understanding their browsing needs and abilities.
- ii) To Check their sharing ethics and principals
- iii) To Verify their priority in downloading patterns
- iv) To Study the trust and behavior analysis
- v) To Find the importance of designing a KM Framework for social networking web pages
- vi) To Formalize the social networking web pages through SNKA and other data mining Algorithms

TABLE I. ONLINE QUESTIONNAIRE RESPONSE

Questionnaire Analysis (Age Range)	
(Normative Study Method - Data Collection)	
Children upto 9 years of age	0
Adolescents 10-19 years	3
Adults 20-45 years	208
Middle age 46-60 years	21
Old > 60	2
Total	234

TABLE II. IMPACT ON VARIOUS LEARNING PLATFORMS

Learning Pattern Observations	Impact
Social Networking as Learning Environment	66.5
General Browsing Based Learning	24.2
Traditional Learning Environment	9.3

The above impact of the study made to think about the decision making skills of the students who simply make their learning patterns as Social Networking based. A set of interview questions were forwarded to many Researchers, HR Managers and Administrators where knowledge is the base of Human Earnings. It was circulated to analyze their view about their employee's frequent usage and knowledge updating factors leading to decision making through social networking.

A few selected questions and their answers are represented in the table 1 and table2 respectively.

The following analysis shows that a serious cause over the learning patterns is indirectly supported by our environment. It has to be formalized for making a faster and efficient decision making efforts.

It also enables a strong bond between the sharing community irrespective of their groupings. The validity or originality behind the acquired and shared content makes a strong trust factor among users in the group which later gets multiplied within other users also.[5]

TABLE III. Q1-SECONDARY DATA COLLECTION - INTERVIEW QUESTION

Interview Question : 1 Companies Expect Employees	Result
Frequent Update - Any means	35%
Allow SN usage	45%
Not Allow SN usage	20%

TABLE IV. Q2-SECONDARY DATA COLLECTION - INTERVIEW QUESTION

Interview Question : 2 Usage of Social Networking	Result
Supports Decision Making Somewhere	55%
Does not Support Decision Making	25%
Not Allowed	20%

III. KNOWLEDGE MANAGEMENT

Knowledge is flourished everywhere in the form of information. The invention of knowledge from information is the process of knowledge management practices. Knowledge management is the process established to capture and use knowledge for the purpose of improving organization performance (Marakas, 1999) or individual's self interest. Knowledge management is emerging as the new discipline that provides the mechanisms for systematically managing the knowledge digitally. In order to have more promotions in the competitive world various knowledge acquisition tools are derived for faster knowledge generation and managing themselves. Knowledge management involves acquisition, generation, codification, storage, transfer, retrieval (Alavi and Leidner, 2001).

Knowledge is the asset of markets, products, technologies and organizations that a business owns or needs to own and which enable its business process to generate profits, and values. These knowledge management processes include:

- i) Developing knowledge,
- ii) Preserving knowledge,
- iii) Using knowledge, and
- iv) Sharing knowledge.

The key elements of knowledge management are collaboration, content management and information sharing (Duffy, 2001). Collaboration refers to colleagues exchanging ideas and generating new knowledge. Common terms used to describe collaboration include knowledge creation, generation, production, development, use and organizational learning (Duffy, 2001). Content management refers to the management of an organization's internal and external knowledge using information skills and information technology tools. Terms associated with content management include information classification, codification, storage and access, organization and coordination (Alavi and Leidner, 2001)

The incorporation of SNKA along with data mining algorithms will closely formalize the socio web pages where people can trust the posts published and could either be used for further purposes or as a decision making system. This SNKA was accomplished based on the generic knowledge acquisition cycle.

DEFINING SNKA CYCLE

The Social Networking based Knowledge Acquisition (SNKA)[51] Cycle depicts the process of absoluteness in knowledge building or knowledge gaining procedures. It provides a faster learning and structured acquisition patterns. The conversion process of unstructured data into structured data solves many deficiencies in learning patterns.

It's a chain of process where all social networking sites work under the same umbrella with a difference in typical human behavior. The Figure1 above depicts the SNKA Cycle which involves the process as:

Sent / Received: Information is sent or received through various sources of social networking tools.

Classify: Received information is to be classified for smoother usages.

Trueness: The validity of the received information is to be tested through search engines or through trusted experts of the domain. The testing leads in finding the trueness such that a network of trustiness begins.

Re-usability: The verification process of the received information defines the status of reusability nature.

Store: Information is stored in the form of knowledge for futuristic usage after satisfying the verification and validation process

Share: After the successful completion of entire cycle, the acquired knowledge could be shared to others and hence this process surely leads to quality in decision making process at learner's view.

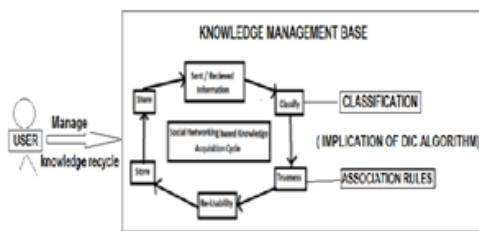


Fig. 1. Implementation of SNKA with Data Mining Algorithms

IV. DATA MINING ALGORITHMS

The Primary data collected through the online mode was taken for analysis. Firstly the statistical tools were incorporated and in order to cross verify and finalize certain decisions pre - defined data mining algorithm tools like classification and clustering algorithms were tested in the obtained data.

Before selecting Dynamic Itemset Counting Algorithm, Apriori algorithm was analyzed. Since, DIC performed faster processing for samples data, We have undertaken taken to release the process without storing the counting of items sets that occurs frequently. [9][10]

The Generic DIC algorithm [Figure 2] was taken up as a case study that overrides as an alternative of Apriori algorithm in finding the association and frequency in web posts followed by their usage. the count in no. of revisit frequencies are used to determine the trustiness and the originality factors. Keeping these key factors as an input a rank indicator will be shown in future to users while and before they start sharing the posts they receive. The overall integration of classification and clustering relatively helps us in finding the correlation among the data sets. A FOAF was also taken for study (Friend Of A Friend).

ALGORITHM (Taken For Study)

1) Mark the empty itemset with a solid square. Mark all the 1-itemsets with dashed circles. Leave all other itemsets unmarked.

2) While any dashed itemsets remain:

a) Read M transactions (if we reach the end of the transaction file, continue from the beginning). For each transaction, increment the respective counters for the itemsets that appear in the transaction and are marked with dashes.

b) If a dashed circle's count exceeds $minsupp$, turn it into a dashed square. If any immediate superset of it has all of its subsets as solid or dashed squares, add a new counter for it and make it a dashed circle.

c) Once a dashed itemset has been counted through all the transactions, make it solid and stop counting it.

INTEGRATION OF DIC ALGORITHM WITH SNKA

Many Data mining methods were analyzed. The classification method was tried to classify the user profile classification based on their profile characterization. Their supportive algorithms are decision tree, Naïve Bayesian, Neural network and C&RT. The clustering methods were also mainly used for studying information retrieval patterns. There were two ways to solve social networking data analysis. I) Graph Based requires study on nodes generated on the web structure and ii) test is based on clustering where new clusters were created based on the behavior of numerical data. Figure1 depicts the implication of SNKA with data mining algorithms. The user can manage his knowledge by following the SNKA framework to maintain his knowledge base stronger. This may impact more on their decisions making stronger.

```

While (DS!=0) or (DC!=0) do begin
read M transactions from database into T
forall transactions t ∈ T do begin
/increment the respective counters of the itemsets marked with dash
for each itemset c in DS or DC do begin
if ( c ∈ t ) then
c.counter++;
for each itemset c in DC
if ( c.counter ≥ threshold ) then
move c from DC to DS ;
if ( any immediate superset sc of c has all of its subsets in SS or DS ) then
add a new itemset sc in DC ;
end
for each itemset c in DS
if ( c has been counted through all transactions ) then
move it into SS ;
for each itemset c in DC
if ( c has been counted through all transactions ) then
move it into SC ;
end
end
    
```

Fig. 2. Dynamic Itemset Counting (DIC)

[Figure3] depicts the process of creating a tool that rides on a browser especially on social networking blogs to provide a rank indicator for the posts that comes as inbox and takes care even after it goes out through the reference links

STEP1: USER LOGINS

STEP2: USER CHECKS FOR NEW UPDATES IN POSTS

STEP3: IT MAY EITHER BE NEW POSTS OR RETURN COMMENTS

STEP4: THE NEW MODEL CHECKS FOR THE RANKING

BASED ON: I) No. of Keywords/ Posts Searched, II) No. of Times Blog viewed, III) No. of Times further the Posted Content was downloaded or Referenced. These are calculated to identify the Frequent Item sets dynamically as the web posts are in the format of unstructured data.

STEP5: RANK INDICATORS PREDICTS THE USER AN INDICATION ABOUT THE POST THEY HAVE RECEIVED.

STEP 6: SHARABILITY OF THE POST IS FINALLY DEFINED.

Using this ranking the user may finally decide whether to forward the post blindly or not. This process also avoids the redundancy in data storage in web platforms.

It is also further may be linked with search engines for better ranking of web pages. Many other tools are provided by many developers like Graph Characterization Toolkit, Tweet hood, Meerkat, Netdriller, Hits/ISAC, D-Dupe and X-RIME are with certain limitations and different in implications.

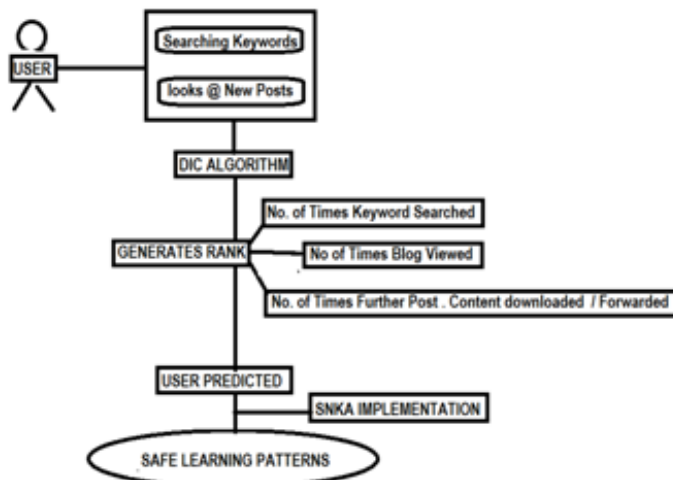


Fig. 3. A Tool supporting SNKA

V. LIMITATIONS OF THE STUDY

This Study focused on the observations of the Indian digital learners alone. Only a few social networks are available for the study and hence, we have taken more focus on facebook, my blogs and linkedin. These social networking sites were mostly preferable among the users hence taken up for user analysis.

VI. CONCLUSION

The Study would help the digital learners to acquire valid knowledge patterns through SNKA and Data mining tools. This study helps to find the originality, trustfulness and the

sharing patterns of any content through knowledge management.. This paper helps to create a SNKA for the unstructured data. The sample datasets of unstructured social networking web pages were analyzed through empirical data collected by qualitative mode. As a further step, we would develop a tool to retrieve a fine tuned knowledge acquisition patterns as a trial and error method.

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Comprehensive Study and Comparison of Information Retrieval Indexing Techniques

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Abstract—This research is aimed at comparing techniques of indexing that exist in the current information retrieval processes. The techniques being inverted files, suffix trees, and signature files will be critically described and discussed. The differences that occur in their use will be discussed. The performance and stability of each indexing technique will be critically studied and compared with the rest of the techniques. The paper also aims at showing by the end the role that indexing plays in the process of retrieving information. It is a comparison of the three indexing techniques that will be introduced in this paper. However, the details arising from the detailed comparison will also enhance more understanding of the indexing techniques.

Keywords—Information Retrieval; Indexing Techniques; Inverted Files; Suffix Trees; Signature Files

I. INTRODUCTION

Information retrieval refers to the process of obtaining relevant information from an existing database that consists of different data that has been collected together. The current state of information retrieval depicts the existence of two search indexing. The first one is metadata and the second one is full text. Metadata is formally outlined as data about other sets of data [1]. It is more precisely described as information regarding other information that is structured. Metadata tool of information retrieval does not take into consideration the complexity of the search question. It can give relevant results to a simple query like the name of an author of a certain book. It can also provide relevant objects to other queries that are complex, like geographical codes. It is usually mostly utilized in education institutions in libraries other resources with large databases [1]. Catalogs of Libraries represent a remote metadata. Reviews of books, art collections as well as summaries also take the form of remote metadata.

On the other hand, full-text tool of retrieving information refers to the use of techniques that search documents stored from single computer. It can also refer to the use of techniques that search of a document that exists in a collection of documents in a collection of full-text database. A full-text search usually performs an examination of all the words that exists in the documents stored in the attempt of matching criteria of searching [1]. Over the years, it has been commonly used in online searches from databases of bibliography. Most application programs, as well as websites, provide capabilities of the full-text searches. Most search engines of the web usually employ techniques of full-text search. However, there

are others that partially index the web pages. The only condition is that the web pages must undergo examination by their indexing systems.

Baeza-Yates and Ribeiro-Neto [2] explain that indexing with full text usually depends on the number of documents. Small numbers of documents can prompt direct scanning of the contents. A strategy known as serial scanning is applied to each query. Serial scanning is the protocol that is usually followed by most tools in the searching process. An example of such tools is Grep, which uses the strategy of serial scanning. Potential largeness of documents or increase of the quantity of queries to search prompts the division of full-text searching process into two stages. The first is indexing and the second one is searching. The first stage of indexing focuses on the scanning of all the existing documents. The stage also sees the building of a search term list. This list of search terms that is usually built at the indexing stage is referred to as an index. However, there are people refer it to as a concordance. The second stage of known as search only references the index in the performance of specific queries. The stage does not reference the text of the documents that are original.

However, this research will focus on a study of the indexing stage and the various techniques that are applied in the process.

There are several indexing techniques in information retrieval. However, this research is going to focus on three indexing techniques namely inverted files, suffix trees, and signature files. The three are the most commonly used techniques in the current world of information retrieval. The process of retrieving information usually begins with a query from a user into the system. A query is a statement that is formal indicating the need for particular information. An example of a query is the search of information in online search engines. Information retrieval queries stated by users do not usually offer a specific object or solution to the problem. Rather, it gives a collection of related objects that match the problem stated in the query. However, the objects have different levels of relevance to the query. Depending on the technique that is used, the relevance of available information is determined with respect to the entered queries. The results given in a form of objects are based on their relevance to the queries. The techniques have proven to the most reliable and usually generate desirable results. However, the indexing techniques differ in many ways. They usually differ in the way

they perform the relevancy tests. They also differ in their simplicity of application. The indexing that is performed by the techniques does not take a similar route. The research in the paper will outline the processes of indexing that the various techniques undertake.

Accuracy is a key factor in retrieving information [3]. Users expect accurate answers from the objects offered in respect to their queries. The accuracy expectation usually cuts across all the information retrieval methods as well as indexing techniques. Users expect to have accurate information no matter the technique that is used in the indexing process. However, it will be shown in the paper that the techniques differ in their accuracy. This research will compare the accuracy levels of the three mentioned techniques.

Despite the high preference of inverted files, suffix trees as well as signature files, they all have limitations. There are various challenges that are associated with each technique of indexing. The level of challenges associated with the application of each technique will be measured in the paper. A detailed comparison of the challenges will offer an understanding about which technique is more limited as compared to the others. Further, the benefits associated with the use of each technique will be outlined. Each indexing technique has benefits that are associated with its use. These benefits and advantages will be critically evaluated and compared. This comparison will offer information about which technique among the three has the most accrued benefits upon its application in the process of retrieving information. Finally, each indexing technique has an objective. The objectives of the various techniques differ across the techniques. This paper will also undertake a study of the main objectives of the techniques which they focus their performance.

II. INDEXING TECHNIQUES

A basic definition of indexing was given in 1988 by Salton [4] as the facilitation of information retrieval accuracy by collecting, parsing and storing data. The accuracy facilitation is performed by use of various methods and techniques. As earlier stated, users need accuracy in the information retrieval process. The indexing process usually has an incorporation mechanism that allows use of concepts from various disciplines. It has been stated that there exists many information retrieval techniques with the common ones being inverted files, suffix trees as well as signature files. This section will discuss each technique into details as well as the way they work.

A. Inverted Files

Inverted files are defined as central components of an indexing algorithm in a search engine. The engine that searches information has a goal of query speed optimization. This means finding documents where a certain word occurs. Then the next step is developing a forward index. The index that is developed plays a role of storing the lists of words in every document. The document is then inverted, leading to a developed inverted index. Sequential iteration is usually required in order to query the forward index. In 2006, Belew [5] suggested that the iteration requires to be performed in each word and document in order to allow the verification of a

document that matches the query. Technically, the resource in terms of time and memory that is required in the performance of such a query lacks an aspect of being realistic. However, the structure of the inverted files that is developed lists the documents per every word. This is done in place of listing the vice versa, where the words would be listed per every document. To perform a clear illustration of the inverted file concept, we assume an existing set of documents. Further, we assume that every document in the set is assigned a list that comprises of keywords. These keywords can also be referred to as attributes. We also assume that there are optional weights of relevance for every keyword. With the assumptions, the sorted list of keywords will be the inverted file. Each attribute will have a link to the documents that contains the specific keyword. Fig. 1 shows how the concept of inverted files works [5].

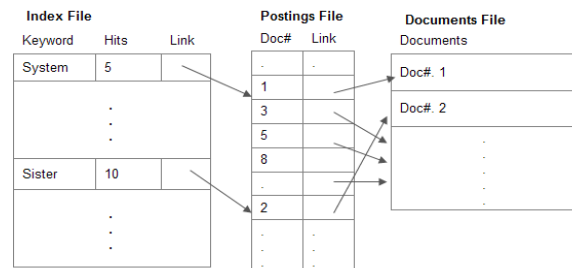


Fig. 1. Inverted files, the index file contains all words in the document and their index, the posting file contains a link to each word with the corresponding frequency, the document file contains the documents

According to Belew [5], the concept of inverted files is mostly used in library systems that are commercial. It is also used in libraries that belong to various education institutions. The reason for the popularity is that inverted files have enhanced efficiency in searching. Basically, the efficiency associated with inverted files is usually necessary when dealing with files that comprise large texts. This is the case for such institutions, which justifies their preference of inverted files.

1) Structures Used in the Inverted Files: There are several structures that are usually used in the implementation of inverted files. The most commonly used structures are sorted arrays, B-trees as well as tries. These structures will be discussed in this section. This will help in giving more information about the concept of inverted files. It will give more understanding of the relationship between inverted files and the various structures that are used in the files' implementation process.

a) Sorted Arrays: The implementation of an inverted file through this structure enables the file to support storage of keywords' lists in a sorted array. This includes several documents that are associated with each attribute. Further, it also includes a link to the documents that usually contain the attributes. Primary storage based systems use a binary search that is standard and are the most commonly used in searching a sorted array. On the other hand, systems that are based on secondary storage usually adapt the sorted array in conformity to their secondary storage's characteristics. Fig. 2. shows the structure sorted arrays as outlined by Barto, et al [1].

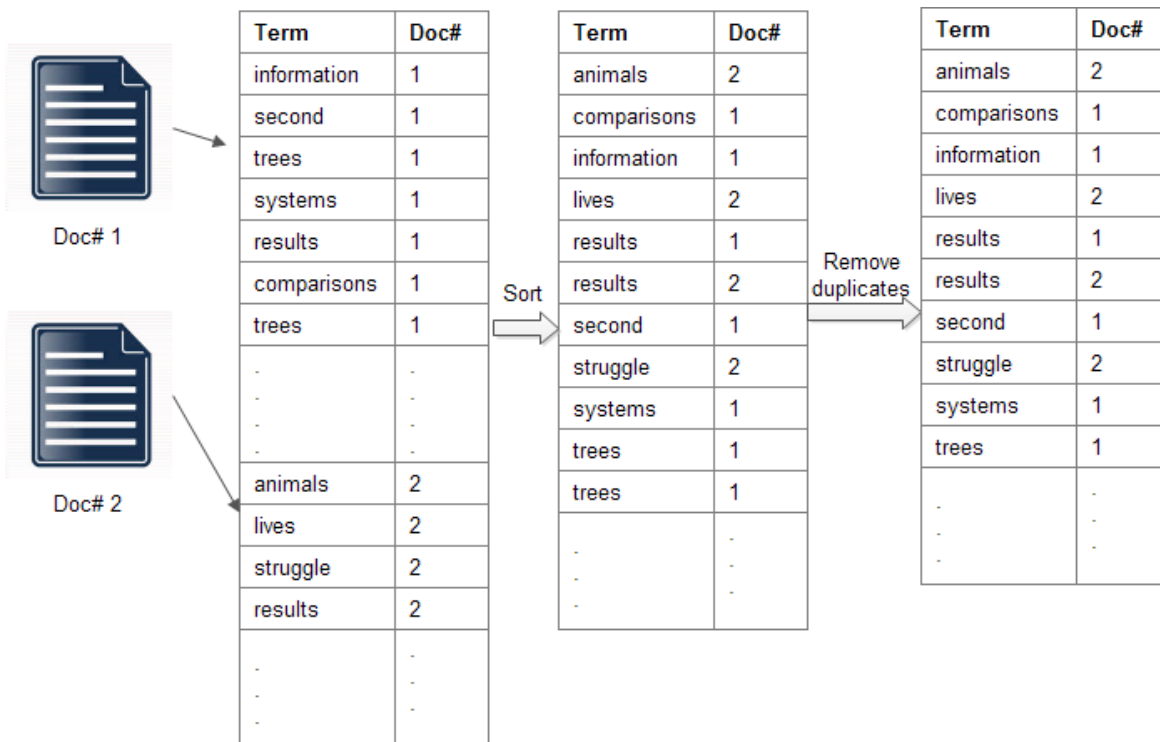


Fig. 2. Sorted arrays, The terms are sorted in lexical ascending order, then the duplicate words are removed

The figure contains two documents, Doc#1 and Doc#2. The words in each document are extracted and inserted in table. The words from both documents are sorted and possible duplication in single document is removed. For example the word trees are repeated in document 1, so it has been removed. The word results are found in both documents so it can't be removed.

The sorted arrays structure has an easy implementation process. It has a reasonable speed that enhances its performance. However, the structure is limited in that it requires frequent update of the index. The frequent updating sometimes is expensive.

b) B-trees: The most common type of the B-tree structure is prefix B tree. It utilizes word prefixes as the primary keys in an index of B-tree. This makes it well structured for the storage of indices that are textual. Every node that is internal usually carries a number of keys that are variable. The shortest word distinguishing the keys stored in the next level is usually named as the key. It is not necessary for the key to be a prefix in the index that is an actual term. The last level in the structure is known as leaf level, as shown in Fig 3. It carries the mandate of storing the attributes with the data associated with them. The order of every node of the prefix B-tree varies because there is dependence on attributes by the internal node keys as well as their lengths [6].

Fig. 3 shows simple Prefix B tree, the first level contains two keys, B and T. The two keys represent separators of the following leaves,

- Words beginning with letter less than or equal B such as Ar and Am,

- Words between B and T such as Co Fi and Ja,
- Words after T such as Un and Wa.

The second level represents other keys for the leaves beneath them, and so on. The last level contains the words of the documents with pointers to the corresponding document.

The B-Tree requires continuous update for maintenance of balance in the tree. The structure has a limitation in that it is not capable of handling many words within the same prefix. The B-tree method is broken down in cases of multiple words. The prefixes that are common usually call for division to avoid space wastage. B-trees usually occupy more space as compared to sorted arrays. However, updates are easier to implement and are faster in comparison with the sorted arrays [6].

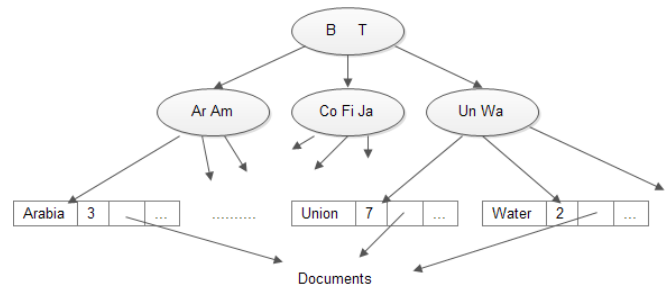


Fig. 3. Binary Tree indicating three levels with keys rpresenting each node

c) Tries: The structure's name was generated from the word retrieval. This structure is widely used to implement inverted files. Digital decompositions of the attributes are highly used by the structure in the representation of the same

keywords [1]. In structures of tries, keys associated to specific nodes are not stored by the nodes. There is similarity of the prefixes of a string in all the descendants of a particular node. There is no necessity of associating values with every node. Values are instead associated with leaves alongside several inner nodes that have correspondence to keys of interest [7]. Fig. 4 shows an example of the Trie structure.

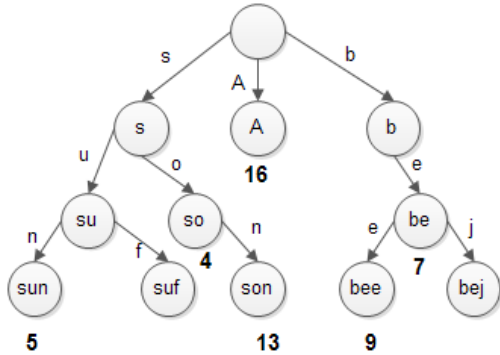


Fig. 4. An example of Trie structure showing no association of values with every node. Values are instead associated with leaves alongside several inner nodes

The example above shows listing of some keys in the nodes while the values are indicated below the nodes. There is an arbitrary integer value associated with every English word that is complete. The example reveals a Trie as a finite automaton that is deterministic and tree shaped. There is generation of finite language by automaton tries. Further, compression of each trie into a state automaton that is deterministic acyclic is implemented as clearly shown in the above example. Tries are also fast in terms of the time used in implementing inverted files. It is also easy to implement as the example is easy to understand.

B. Suffix Trees

A suffix tree is a Trie that is compressed and contains suffixes of the given texts as the keys that belong to them as well as their values as the positions present in the text. The idea of compressing tries makes suffix trees be referred to as tries. Consequently, the sub-trees are referred to as sub-tries. The concept of suffix trees was developed in the year 1973 by Weiner [8]. The first online construction of suffix was to be developed by Ukkonen [9].

The running time associated with the algorithm was ranked as one of the fastest at that time. However, the algorithms were all linear-time for a size alphabet that was constant [10]. Generally, they had a running time of $O(n \log n)$. In 1997, Farach [11] designed an algorithm of suffix tree construction that had optimism for all alphabets. It was the first algorithm of linear-time for strings that were drawn from integers of the alphabet, in a range that was polynomial. This was the foundation of new algorithms that have been later developed in the construction of both suffix trees as well as suffix arrays.

Assuming a suffix tree for the string S and length n , the definition must meet several requirements [12]. Firstly, there must be exactly n leaves that are numbered from 1 to n in the tree. Every node that is internal must also have at least two

children with the exceptional of the root. The labeling of the edge is done with non empty S substrings [13]. Any two edges that start out of a node should have strings labels that start with a different character. This condition means that it is not possible for a suffix to be a prefix that is proper for another. The digit that comes last in the data is a , and it appears two times in the data. Lastly, suffix $S[i..n]$ is spelt out by the string obtained after the concatenation of string labels. These labels are the one present in the path of root to leaf.

Let us assume a string $s = \text{peeper}$. The non empty suffixes of the string will be peeper , eeper , eper , per , er and lastly r . Developing a suffix tree for the string peeper will comprise a compressed trie containing elements peeper , eeper , eper , per , er and r . The alphabet of the string is e, p , and r . This means that the radix of the trie that is compressed is 3. Fig. 5 based on [10] indicates a Trie for suffixes of the word "peeper".

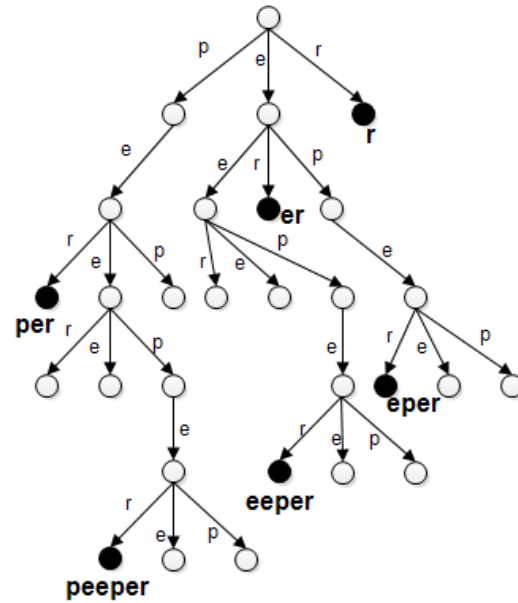


Fig. 5. Trie representation of the word "peeper", the compressed version the Trie is done with eliminating leaf white nodes

The use of suffix trees is applied when solving multiple string problems occurring in free text search as well as text editing. Suffix trees are also used in computational biology as well as other areas of application. However, there are several primary main applications of suffix trees. Firstly, they perform a search of a string in $O(m)$ complexity. In such an application, M represents the substring's length. However, there is a mandatory requirement that there be time $O(n)$ sufficient for building the string's suffix tree. Secondly, suffix trees are used to find the repeated string that is longest. It is also applied in the process of finding the common substring that is longest. Lastly, the longest palindrome in a string is found through application of suffix trees [14].

The above mentioned applications are useful as they expand the use of suffix trees. They enable them to be used in real life processes. For example, they are widely used in bioinformatics applications. They are also widely used in the search of DNA patterns as well as sequences of proteins.

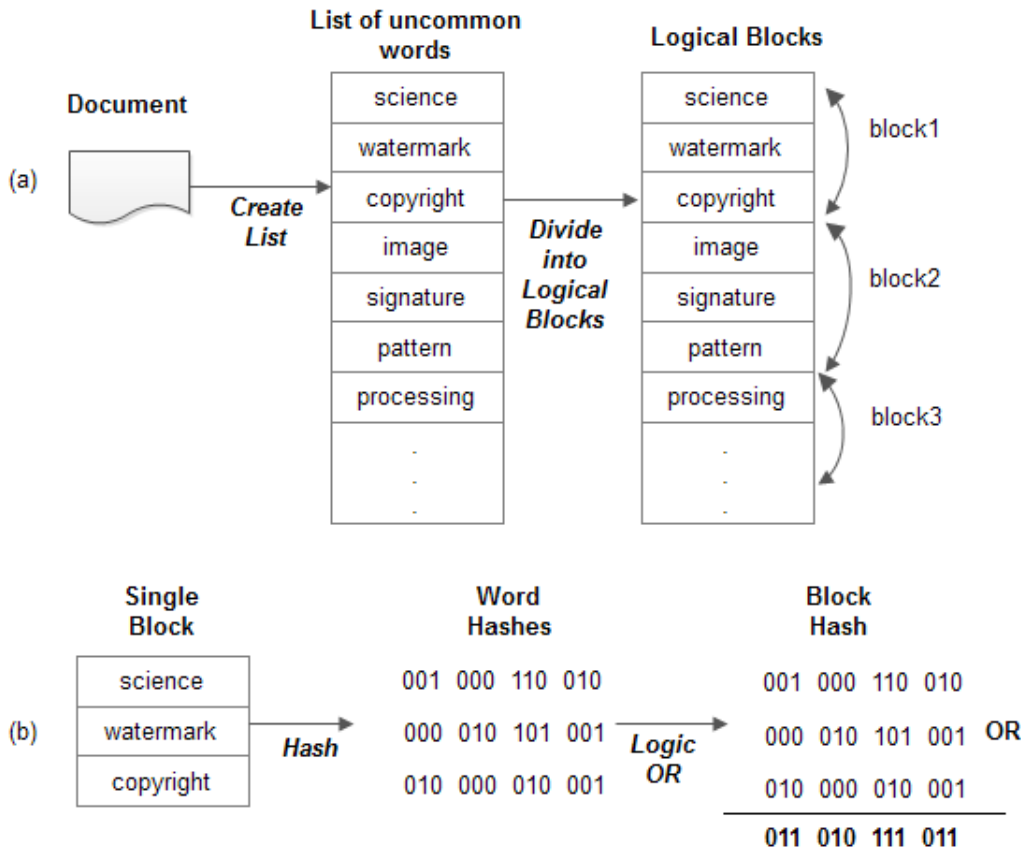


Fig. 6. The signature file is created by hashing every uncommon word to a given number of bits with fixed width

However, the sequences must be viewed as characters that are long stringed. According to Callan et al [15] the greatest advantage that makes suffix trees popular is their ability to make long searches with minimal mismatches. This makes them candidates to be used in data compression when they enable the finding of data that is repeated. Lastly, most search engines also use suffix trees in the process of clustering data.

C. Signature Files

A signature file is a technique of indexing that usually creates a filter that is “dirty”. An example of such a filter is the Bloom filter that keeps all the existing documents matching to the query entered by a user and also hopes to keep the ones that do not match the criteria. This is done through creation of a signature for every file which is typically a version of a hash code [16]. Therefore, a signature is an abstraction of a record which has been mapped. Signature files are generated through two main methods: Word signatures as well as Superimposed coding. The word signature approach involves hashing of identifiers which are basically words of a record to a bit pattern. The patterns or word signatures later form the record signature through concatenation. On the other hand, Superimposed generation of signed signatures involves hashing every uncommon word to a given number of bit positions, say S with a width that is fixed, say F , Fig. 6. Superimposing, through bitwise OR is performed on the resulting signatures for the generation of the record signature [17].

Fig. 6 shows a document is processed by creating list of uncommon words. A stop list of common words must be created to remove such words from further processing. Common words have no effect on defining the document character. The word list is divided into logical blocks. Each logical block, as shown in Figure (b), is hashed by hashing each single word. The block signature is obtained by logical ORing the word hashes. The main idea of Bloom Filter [18] k -th order Bloom filter has k independent hash functions $H_1(x)$, $H_2(x)$, ..., $H_k(x)$, that maps a word to a hash value in the range 0 to $N-1$, where N is the length of the hash bits. Formally,

$$H_i(X_j)=y, \quad 1 \leq i \leq k; \quad 1 \leq j \leq D; 0 \leq y \leq N-1; \quad (1)$$

Where, X_j is the j th word in the uncommon list, D is the number of uncommon words in each document. The following procedure is applied

1) A has table of N bits size is created and all of its bits are set to zero.

2) For each word in the word's list, its k hash values are calculated, and accordingly the corresponding bits are set to 1. Thus for example if $H_i(X_j)=68$ for some (I,j) , then the sixty-eighth bit of the hash table is set to 1, if the bit is already 1, then no change will be done.

When searching for specific keyword, the keyword's k hash values are calculated. If all the corresponding values in the

hash table are all set to 1, then a matching is found, otherwise no match.

A signature approach that is naive would involve a uniform and a random hashing. Same S bit positions are hashed by any given n -gram [10]. A possibility exists that two n -grams that are different will most probably hash too the same position of bit. The occurrence of hashing same bit position by different n -grams is referred to as collision. The possibility of collision is contributed by the fact that the F chosen in most cases is usually less than the number of unique n -grams in total.

III. COMPARISON OF INDEXING TECHNIQUES

This section will focus on the comparison of the three discussed techniques of indexing in information retrieval. It is clear from the previous discussion that the techniques differ in many aspects. This is despite the fact that they work towards yielding same results. All the techniques are aimed at indexing and undertaking successful information retrieval. However, the approaches that are used by then various techniques are different. The techniques also vary in their performance as well as their stability. The techniques also vary in terms of their limitations as well as advantages. This section will critically focus on the performance and stability of each technique and compare them with the rest. It will also compare the limitations of each technique and make a detailed comparison. This will enhance the understanding and knowledge about the three indexing techniques.

A. Performance Comparison

The performance comparison between inverted files, suffix trees as well as signature files can take several dimensions. However, the main parameter used in determining the performance of the techniques is the processing time of the various techniques [19]. This is the time that is taken for a system using a particular technique to give response to a query raised by a user. The comparison of the performance between the various indexing techniques in this section will be based on the response time. This will help to effectively determine the performance of each indexing technique.

1) *Inverted files*: The structure was developed with a primary goal of optimizing the speed of the query. The structure's performance is based on an iteration of a developed inverted index. Querying the forward index is the main reason as to why the iteration is necessary. However, as discussed, it would be technically unrealistic to take the time required for the iteration. Inverted files have several approaches of performance that enhance the response time that is required.

Firstly, developed inverted files usually list the documents on basis of "per every word". Secondly, inverted files have a special performance approach that is known as skipping. This involves introduction of synchronization points which are additional locations that usually offer a platform for the commencement of decoding to the inverted list [19]. The index in the inverted files contains both the difference in document number as well as the difference in bit address. This results to the capability of inverted files to be stored as sequence that is compressed. It is the compression capability that enhances the performance of inverted files and highly saves on the processing time [2].

The results of an experiment on 100 documents from the internet with applying skipping on the inverted files retrieval technique is shown in Fig. 7. The experiment executed some queries on the documents, the figure shows the CPU time required to process those queries.

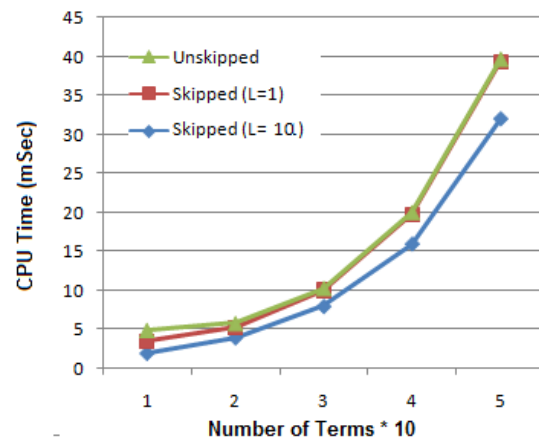


Fig. 7. The effect of skipping on the inverted files indexing performance

In the figure, L represents the skipped index

2) *Suffix Trees*: A suffix tree has already been expressed in the previous section as a compression of sub tries. Therefore, the performance of suffix trees is based on basis of compression. It is clear that suffix trees support compression and indeed perform when compressed. The processing and running times associated with the algorithm are one of the fastest. Its performance is ranked as one of the most efficient taking processing time as a parameter. The running time of suffix trees is generally given as $O(n \log n)$.

There are several reasons for the efficient performance in terms of time. The first reason is the support of insertion. This is put as a condition in any dynamic suffix tree. The second reason is the ability to perform deletion. Lastly, suffix trees carry special capability to perform modification of strings. These are the unique traits with suffix trees that makes the technique's performance to stand out among all other indexing techniques. There is no other indexing techniques whose performance involves insertion, deletion as well as strings modification in the manner that suffix trees perform.

3) *Signature Files*: The performance of the signature files is largely based on unique signature development for every file. This development of signatures as explained in the previous section is done through word signatures as well as superimposed coding. An evaluation of the procession time as a parameter to determine the efficiency in performance reveals several things. Firstly, there is a possibility of slowness as compared to other techniques due to the concatenation due to the word signatures. Secondly, the time taken to respond to queries raised by users can increase when using signature files technique due to the sequential nature of the files [20]. This is mostly the case for the files that use superimposed coding other than word signatures.

There are special features associated with the technique that make it fast and pass the efficient performance test. The technique like others supports compression in order to enhance its performance. This enables the technique to be well placed to improve the time for processing objects in the indexing process. The technique also utilizes partitioning. This is usually the most unique feature that is associated with signature files as a technique of indexing. There are not many techniques that are known for both vertical as well as horizontal partitioning.

B. Stability Comparison

The aspect of stability is explained in the ability of various techniques to handle the files that contain information which the users are looking to retrieve. Stability in the field of information retrieval is simply the variance that is associated with the results of the queries of various queries. This means the relationship between the objects provided by a certain technique with the query that is entered by a user. The relevance of the results in respect to the queries forms the basis of stability discussion. This section will compare the results that are given upon the use of various techniques. This will shed more light about the stability of suffix trees, signature files as well as inverted files. Generally, variance is usually measured by undertaking a balance between the risks and rewards. The risks are the threats to a technique in performing and giving the desired results [3]. On the other hand, the rewards are the desired objects that can be obtained by using a certain technique of indexing in the process of retrieving information. However, this is usually challenging as there must be a clear way of determining the rewards as well as the risks. The study of stability best illustrated in a risk/reward curve as shown in Fig. 8. Algorithm A dominates algorithm B. The figure shows two algorithms that appear identical in terms of mean average precision (MAP) gain may have very different risk profiles.

1) *Inverted Files*: Inverted files have a considerably desirable trade off between the risks and the rewards. They are seen as one of the most stable indexing techniques. The index construction in an inverted file was explained by Kanaan et al [20] as shown in Fig. 9.

The diagram shows the possibility of deriving an inverted file upon completion of the Trie structure. The structure as indicated enables access to the file in main memory. This is the basis of the strength and stability of inverted files. The reason is that every entry has a reference position of the posting file which is usually held in storage that is secondary. This brings out an aspect of back up and easy tracing of entries. This has few risks associated with it and results in a stable indexing process.

2) *Suffix trees*: A suffix tree is built with a high threshold of stability. The construction of suffix trees is performed with the principle that every string is supposed to be padded a marker symbol that is out of alphabet and unique. This serves the purpose of ensuring that any suffix in the construction does not become a substring of the other. Since the building of the suffix trees involves leaves, every suffix has a representation by a leaf that is unique. This means that the reward risk assessment is passed by suffix trees. The risk that

is associated with most techniques is false results. However, the suffix trees eradicate that by ensuring that every suffix is served by only one leaf. Therefore, stability is maximized in suffix trees.

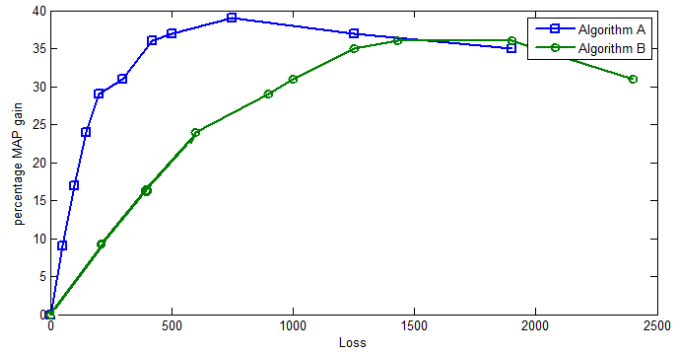


Fig. 8. Risk/reward curve showing query expansion. The curve shows two retrieval algorithms compared in performance. Algorithm A performance is better than Algorithm B.

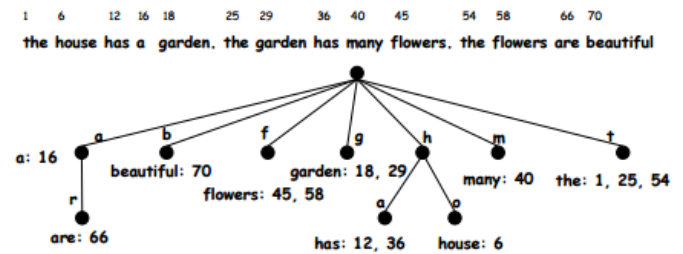


Fig. 9. The index construction in an inverted file. The figure based on [20]

C. Limitations Comparison

Despite the many strong points that the discussed indexing techniques have, they usually have various limitations that make it hard for them to perform optimally. However, the indexing techniques are not limited in the same way. They usually have different limitations in their performance. This section will focus on the limitations of every indexing technique. The limitations will then be compared to offer a synopsis of the limitations and the ability of each technique to overcome the challenges.

Inverted files have their share of disadvantages that usually pose a challenge to their efficiency in offering optimum indexing. This in turns affects their application and usage by most users across the world. Firstly, the technique has a limitation in that there is difficulty in the update of insertions especially of new records [1]. This usually requires moving of proportions of files that are large. Secondly, random access of any system by use of the inverted file technique is usually slow. In some cases, files are usually considered to be organized in a sequential manner even when there is no order to a certain key. This creates the possibility of false objects because sometimes acquisition date can be regarded to as the key value.

On the other hand, Robertson and Sparck [21] suggest that suffix trees are limited in that they usually require a lot of space due to the nature of their construction. The internal pointers in the tree usually require more space for storage. This

is in comparison to most of the other techniques that consume considerably less space. Suffix trees also have a challenge in that it is necessary for them to be built in an order of reverse. This means that characters have to be added from the input's end. Lastly, the nature of the tree works against it because the string's length can be a single variable. This could occur within the same class that the segments of the leaf belong. This works against it because the side to side co existence of suffix trees could be impossible because there would be similarity of the class of leaf segments.

Lastly, signature files indexing technique has a share of practical problems in its performance. There are many methods that are used by signature files to enhance operation. The variety in the methods of operation also expands the limitations of the signature files. Firstly, signature files' performance is known to deteriorate as the files grow [22]. This simply means that signature files have a limitation of performance in files that are large. Secondly, the technique has a disadvantage because in case the keywords number in every document is large, then a huge hash table must be made. It might also lead to usual queries touching a proportion of the database that is large. Lastly, the signature files technique is limited in handling queries that are not conjunctive [23]. This difficulty in dealing with non- conjunctive queries limits the performance of signature files. This mostly happens when signature files utilize the method of Gustafson's in the process of indexing.

IV. CONCLUSION

The research in this paper has clearly achieved a critical analysis of indexing techniques. It has offered information about the construction of various techniques such as inverted files, Suffix trees and Signature files. In addition the paper introduced detailed structures that make up these techniques. The research has also given more understanding of the building of the structures and the way that they work. The paper has detailed few benefits that are associated with the use of every technique. The speed, as well as the space that is required for the various techniques to optimally operate, has been outlined. This has provided the basis of the comparison that has been done between the various techniques.

The comparison done in this paper has taken the dimension of performance, stability as well as limitations. The performance of inverted files, suffix trees, as well as signature files, is compared in the paper by using the processing time as the parameter. The paper has done a comprehensive comparison of the time taken for every technique to respond to various queries that are raised by users. On the other hand, the stability of every technique of indexing has been discussed and compared to other techniques. The parameter used for the stability discussion is the measure of rewards and risks associated with every technique. Lastly, the paper has undertaken a comparison of limitations and challenges of every technique. This comparison has helped in knowing the challenges a user would get by using a certain indexing technique.

TABLE I. COMPARISON OF INVERTED FILES, SUFFIX TREES AND SIGNATURE FILES INDEXING TECHNIQUES

Indexing Technique	Capabilities	Limitations
Inverted Files	<ul style="list-style-type: none">• Compression• Skipping• Iteration	<ul style="list-style-type: none">• Difficult update of insertions• Slow random access
Suffix Trees	<ul style="list-style-type: none">• Compression• Insertion• Deletion• Modification	<ul style="list-style-type: none">• Requires a lot of space• Require reverse building• String's length can be a single variable
Signature Files	<ul style="list-style-type: none">• Compression• Vertical Partitioning• Horizontal Partitioning	<ul style="list-style-type: none">• Limited performance for large files• Must build huge hash table in case of large keywords in each document• Difficulty in dealing with non-conjunctive queries.

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Writing Kurdish Alphabets in Java Programming Language

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Abstract—Nowadays, Kurdish programmers usually suffer when they need to write Kurdish letter while they program in java. More to say, all the versions of Java Development Kits have not supported Kurdish letters. Therefore, the aim of this study is to develop Java Kurdish Language Package (JKLP) for solving writing Kurdish alphabetic in Java programming language. So that Kurdish programmer and/or students they can convert the English-alphabetic to Kurdish-alphabetic. Furthermore, adding Kurdish language to standard Java Development Kit (JDK). Additionally, in this paper we present the JKLP standard documentation for users. Our object-oriented solution composed of a package consisting two classes which have been implemented in the Java programming language.

Keywords—Java; Arabic Scripts; Java language support; Java issues; Kurdish Language

I. INTRODUCTION

Kurdish (Kurdish: Kurdî, كوردی, Kurdî, Кӱрдî) language belongs to the Indo-European family of languages. Kurdish dialects are members of the northwestern subdivision of the Indo-Iranic language, Iranian branch of this largest family of language in the world. The Kurdish language is an independent language, having its own historical development, continuity, grammatical system and rich living vocabularies. The Kurdish language was derived from the ancient "Median" language or "Proto-Kurdish". Ca. 30 million people in the high land of Middle East, Kurdistan, speak different dialect of Kurdish.

In fact, Kurdish language has four major dialects (groups) and sub-dialects namely northern Kurdish dialects (Kurmanjî and Badînanî), Central Kurdish dialects Sorani (Sulaimani and Mukrayani). Moreover, the other two major branches of Kurdish language are Luri ("Mamasani, Kurmanshani and Kalhuri) and Goranî (Hawrami, Zazayee and Shabak). Meanwhile, these are further divided into scores of dialects and sub-dialects as well [6]

The statistics shows that the number of Kurdish speakers is Between 25 and 35 million Kurds inhabit a mountainous region straddling the borders of Turkey, Iraq, Syria, Iran and Armenia. They make up the fourth-largest ethnic group in the Middle East, but they have never obtained a permanent nation state [2].

Kurdish and Arabic letters are similar but they are not exactly the same. The Kurdish alphabet contains the Arabic alphabet and six additional letters. In fact Kurdish and Arabic alphabets are consisting of 36 and 28 letters respectively. In the following some of differences between these two alphabets are given [3]

- Arabic alphabet lacks six letters namely "Woo" , "Ooh" "Peh", "Tcheh", "Jeh" and "Gaf".
- There exist some letters in both alphabets which have Arabic and Kurdish versions. For example Arabic "Kaf" and "Yeh" are different from Kurdish ones.
- The orders for some letters are not the same. For example in Arabic alphabet "Heh" is prior to "Waw" whereas in Kurdish, they appear in opposite order and "Waw" is prior to "Heh" [3]

II. LITRETURE REVIEW (BACKGROUND)

This section of the study is focused on the literature available to the team in the form of journals, books and the reliable websites. The research will be carried out by the researchers. The topics which are covered in this section are: Definition of languages, Prosperities of Kurdish languages, Java programming languages and the others that will be discussed in the following.

A. Description of Language

It can be said that language is the human capacity for obtaining and using a complex system of communication. The term of LINGUISTICS is the scientific study of language. There are no studies and researches to know accurately about how many languages there are in the world. Furthermore, it could not be known about the number depends on a partly arbitrary difference between languages and dialects. However, approximations differ between around six thousand and seven thousand languages in number [5]. In any country, people have two kinds of languages: mother tongue is the primary language which has spoken between peoples, and secondary language is the language that some people can speak with the other people who are in different country. In Kurdistan Region of Iraq

(KRI), there are two kinds of language: Kurdi and Arabic. The Kurdi language is the mother tongue, but the Arabic language is the secondary language. Moreover, in any country have some people who can speak more than two languages. When used as a universal conception "Language" might be referred to the cognitive ability to learn, and utilize systems of complex communication.

B. Properities of Kurdish Language

In this section, there are some of the characteristics of Kurdish scripts are introduced and compared to Arabic/Persian and English languages. Firstly, Kurdish scripts are written in a right to left writing like Arabic/Persian languages. On the other hand, Firstly, Kurdish scripts are written in a right to left writing unlike English language [10].

Kurdish scripts could be written in different forms as identified in table 1. It could be written in the start, middle, end and isolated respectively. In Table 1 the list of all Kurdish characters and their Unicode's are presented.

TABLE I. EXISTING KURDISH ALPHABETS [9]

Isolated	Start	Middle	End	IPA	Isolated	Start	Middle	End	IPA
ا	ئا	ـا	ا	[a:]	م	م	م	م	[m]
ب	با	ـب	ب	[b]	ن	ن	ن	ن	[n]
ت	تا	ـت	ت	[t]	و	و	و	و	[o]
د	ـ	ـ	د	[d]	پ	پ	پ	پ	[p]
ه	هه	ـه	ه	[æ]	ق	ق	ق	ق	[q]
ع	ع	ع	ع	[ʕ]	ر	ـ	ر	ر	[r]
ئ	ئا	ئا	ئ	[ɛ]	ر	ـ	ر	ر	[r]
ف	فا	ـف	ف	[f]	س	س	س	س	[s]
گ	گا	ـگ	گ	[g]	ش	ش	ش	ش	[ʃ]
ـ	ه	ه	ـ	[h]	ت	ت	ت	ت	[t]
ح	حا	ـح	ح	[ħ]	و	ـ	و	و	[u]
ـ	ـ	ـ	ـ	[I]	و	و	و	و	ـ
ی	یا	ـی	ی	[i:]	ـ	[u:]	ـ	ـ	ـ
ج	جا	ـج	ج	[dʒ]	ف	ف	ف	ف	[v]
ز	ـ	ـ	ز	[ʒ]	و	ـ	و	و	[w]
ک	کا	ـک	ک	[k]	خ	خ	خ	خ	[x]
ل	لا	ـل	ل	[l]	ه	هه	ه	ه	[Ha]
ـ	ـ	ـ	ـ	[t]	ـ	ـ	ـ	ـ	ـ

C. Java Programming Language

Computer language indicates a sequence of instructions which will make a computer follows and run a program based on these instructions. Instructions are composed of a sequence of ON "1" and OFF "0", such as 101100011001, that a computer follows as it runs them through the processor, turning switches ON and OFF [4]. Program languages are able to help computer to understand what people would like to do. It may not be able to classify computer programming languages since there is no single standard for categorizing computer languages [8]. In reality, dozens of classified is by paradigms that provides the programmer's view of code execution.

The most common Object-Oriented Programming (OOP) languages include Java, Visual Basic, C#, Python, C++ and the other OOP languages. Java language is a computer programming language, and originally developed by James Gosling at Sun Microsystems. In general, Java became available in 1995. It is based on C++ language and shows substantial similarity to it. Java language is intended to let developers (write once, run anywhere). It is currently one of the most popular computer languages in use, and is widely used form application software to Web applications [11]. Nowadays, it can be seen many application developers attempt to use Java language to build their applications because Java can provide lots of features, namely, Portable, Secure, Simplicity, Object Oriented, Robust, Distributed, Performance, Dynamic and the other features.

D. Related Between Java Language and Kurdish Students

There are many Universities in KRG that provide computer science degree for students. University students should know about one type of OOP languages after graduation. The computer science degree in all Universities must teach one type of OOP which is Java language. Therefore, university students should know about Java language. Because of these reasons we have decided to choose Java language to accomplish this study.

This research will be more helpful for university students since they can write Java instructions by their mother tongues. On the other hand, lots of students in KRG are not able to speak and write English language, so this study will assist students to accomplish their assignments. Furthermore, software developers will obtain benefits from this study since they can use Kurdish alphabetic in their applications and software.

E. How to Create Package in Java Language

Packages in Java are a namespace which organize a group of related classes, interfaces and sub packages. They will be able to provide access protection and namespace management [12]. Theoretically you can consider of packages as being similar to various folders on your computer. Developers might keep images and video in one folder, HTML pages in another and scripts files in another. Due to the software written in Java could be collected of thousands of individual classes with related things in the same directory, which is really making sense to organize things [12]. There are several advantages that can be taken from using packages: reusability, simple to locate the files, name-space crash and collisions.

While building a package, developers should select a name "it must follow the Java code names rule" for the package and include a package statement along with that name at the top of each source file which holds the classes, interfaces and notation types that developers want to add in the package. The following code showed that how the package can be created in Java code:

```
package kurdPackage;
class kurdClass{
}
```

As it can be seen in the above Java code, the package statement must be the first line in the source file. It can be used

only one package statement in each source file, and that package name applied to all types in the file [12].

F. Previous Works

Our team investigated into existing solutions which are similar to the solutions that have been discovered in this study, but no solution has been reported on the solution to the Kurdish language in the Java codes. As a result, it can be said that this study will be the first which worked on the find the way to use the Kurdish language in all Java codes. However, there are several solutions that have used for the other languages, namely, Sindhi and Arabic and the other languages. In this section it will look at two solutions that are similar to our solution.

The first solution was defined by Ismaili et al in 2011, which they have built the Graphical User Interface (GUI) for the Sindhi language [7]. This solution helps Sindhi people to use of computing for several basic tasks, for instance editing, composition, formatting and printing documents in Sindhi by using this solution [7]. The second solution was made in 2013 by Alshahad, this solution was designed a new Java package to learning Arabic languages for non-Arabic speakers [1].

III. WORK UNDERTAKEN

This study composed of two main phases, which are system design and system implementation.

A. System Design

Our proposed package (Java Kurdish Language Package) JKLP consists of two main classes. The first class named as KurdishKeyboard which converts the written-English alphabets to Kurdish-alphabets. The second class named as KurdishKeyboardTest to compile and run the KurdishKeyboard class. Figure one illustrates the detailed structure of the JKLP package.

Class Diagram:

As it can be seen our solution is composed of two main class which are KurdishKeyboard and KurdishKeyboardTest.

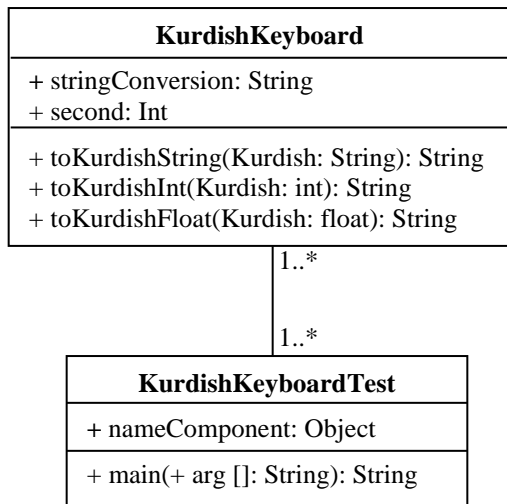


Fig. 1. Class Diagram

The KurdishKeyboard consists of three main methods which are:

1) ToKurdishString:

This method converts the English-alphabets to Kurdish-alphabets. In order to learn and use the corresponding Kurdish letter with English letter, users must stick with the following table. As the number of Kurdish alphabet is more than English alphabet, we were obliged to use some Capital English letter to be equal with the numbers of Kurdish-alphabets. For more information please see the following table.

TABLE II. KURDISH- ENGLISH TABLE CORRESPONDING LETTERS

English Letter	Kurdish Letter	English Letter	Kurdish Letter
a	ا	n	ن
b	ب	o	و
c	ج	p	پ
D	د	q	ق
E	ة	r	ر
F	ف	s	س
g	ط	t	ت
h	ه	u	ئ
i	ي	v	ظ
j	ز	w	و
k	ك	x	خ
l	ل	y	ی
m	م	z	ز
P	ا		ع
X	ص	L	ل
C	ض	R	ر
G	غ	S	ش
H	ح	Y	ی

2) ToKurdishInteger:

The main aim of this method is to Convert English Integer numbers to Kurdish Integer numbers. It is worth mentioning that this method converts all integer number regardless of its size such as Byte, Short and Long data types. Please see table three in order to find the corresponding English- Kurdish integer numbers.

TABLE III. ENGLISH- KURDISH CORRESPONDING NUMBERS

English Number	Kurdish Number
0	٠
1	١
2	٢
3	٣
4	٤
5	٥
6	٦
7	٧
8	٨
9	٩

3) ToKurdishFloat:

ToKurdishFloat method Converts English Floating type numbers to Kurdish Floating type numbers. Please see table four in order to find the corresponding English- Kurdish floating point numbers.

4) *ToKurdishDouble*:

As we have two floating point types we were obliged to provided another methods to convert Double English Number to Double Kurdish numbers. Please see table four in order to find the corresponding English- Kurdish double numbers.

TABLE IV. ENGLISH- KURDISH CORRESPONDING NUMBERS

English Number	Kurdish Number
0,0	٠,٠
1,1	١,١
2,2	٢,٢
3,3	٣,٣
4,4	٤,٤
.....

The second class which called TheKurdishKeyboardTest contains one Main mehods which is:

Main methods:

In java programming main methods is considered as default method to compile and run java program. In our solution we have provided this method to run and test the solution. Beside above classes our package has included the JUnit testing classes for validation process. All the operation has been tested using unit testing in accordance with current standards.

More importantly, as we all know documentation has been considered as crucial stage of developing packages and applications. Based on that, the documentation has been provided for users.

B. System Implementation

As we have explained earlier, our solution consists of two main classes. The KurdishKeyboard is consists of four main methods. In the below flowchart explains the complex operations about the solution.

It is evident that texts usually contain alphabetic, numbers including integers and floating point number types, and other symbols. The program will accept any input text from interface or any other sources and then comparison stage will start. The program search into the text and analyze to find out which method should be called. Obviously, the loop will continue until the end of the text.

When the program find alphabetic, the toKurdishString method will be called to convert the English character to Kurdish character. The program will go through to the memory to find the correspondent memory number for Kurdish character.

However, when the Numbers are found in the text, the program needs to check whether the number is integer or floating point. As it shown in the flowchart, the toKurdishInt method will be called when integers found.

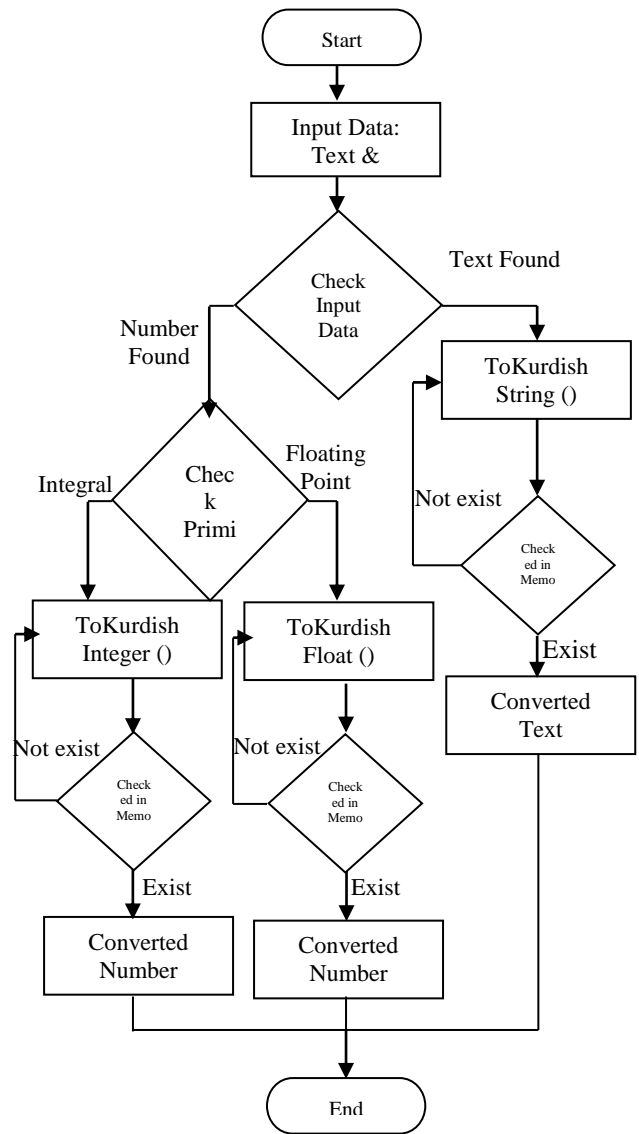


Fig. 2. Flow Chart

```

public String toKurdishString(String kurdish) {
    strConv=kurdish;
    // Initializing String builder to store
    //the Letter number in the memory
    StringBuilder sb = new StringBuilder();
    for(int i=0;i<strConv.length();i++)
    {
        second=i+1;

        if(strConv.charAt(i)=='a')
        {
            sb.appendCodePoint(1575);
        }
    }
}
    
```

Fig. 3. toKurdishString Implementation Code

```
public String toKurdishInt(int toKurdish)
{
    StringBuilder sb = new StringBuilder();
    String castInt=toKurdish+"";
    for(int i=0;i<castInt.length();i++)
    {
        if(castInt.charAt(i)=='0')
        {
            sb.appendCodePoint(1632);
        }
    }
}
```

Fig. 4. toKurdishInt Implementation Code

On the other hand, the toKurdishFloat will be called when floating points are detected.

```
public String toKurdishFloat(float toKurdish)
{
    StringBuilder sb = new StringBuilder();
    String castInt=toKurdish+"";
    for(int i=0;i<castInt.length();i++)
    {
        if(castInt.charAt(i)=='0')
        {
            sb.appendCodePoint(1632);
        }
    }
}
```

Fig. 5. toKurdishFloat Implementation Code

Similarly, the toKurdishDouble method will be called when Double floating point number types are found.

```
public String toKurdishDouble(double toKurdish)
{
    StringBuilder sb = new StringBuilder();
    String castInt=toKurdish+"";
    for(int i=0;i<castInt.length();i++)
    {
        if(castInt.charAt(i)=='0')
        {
            sb.appendCodePoint(1632);
        }
    }
}
```

Fig. 6. toKurdishDouble Implementation Code

IV. RESULT AND DISCUSSION

As we have mentioned earlier, we have developed an interface to showcase our solution with the aim of providing better understanding for java users. The issue of writing Kurdish Letters has been long there and programmers have been suffered considerably. However, this issue is no longer available since programmer can use JKLP package to write Kurdish letters whenever they need it.

Using the JKLP package is handy by only invoking the methods you need to obtain the Kurdish letters as you can see from the following example:

```
public class KurdishKeyboardTest{
    KurdishKeyboard ku = new KurdishKeyboard();
    public KurdishKeyboardTest()
    {
        lab = new JLabel(ku.toKurdishFloat(88.999f));
        JFrame f= new JFrame(ku.toKurdishString("Malawa"));
        f.setSize(400,600);
        f.setVisible(true);
        JButton b = new JButton(ku.toKurdishString("Hawkari"));
        JButton bl= new JButton(ku.toKurdishString("Gorini Text"));
        JLabel lab = new JLabel(ku.toKurdishString("Nusin"));
        JTextField txt = new JTextField(ku.toKurdishString("Program"));
    }
}
```

Fig. 7. Using JKLP Package and its Methods

As it can be seen, the use of our JKLP package is so easy you just need to call the necessary methods. We have provided four methods which are toKurdishString, toKurdishInt, toKurdishFloat and toKurdishDouble, the below figure shows the output of the main program.



Fig. 8. Main Screen

The toKurdishString converts English-alphabetic to Kurdish-alphabetic. An example this conversion can be seen in the following figure.

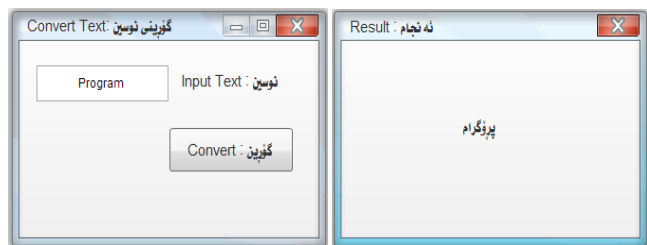


Fig. 9. toKurdishString Conversion Screen

As it can be notices the program converts all the English-Written letters to Kurdish-Written letters in accordance with defined table of corresponding letters which have used as the dictionary in the solution. Similarly, the program can converts number to required Kurdish system. Below figures shows how the systems converts English Integer numbers to Kurdish Integer numbers.

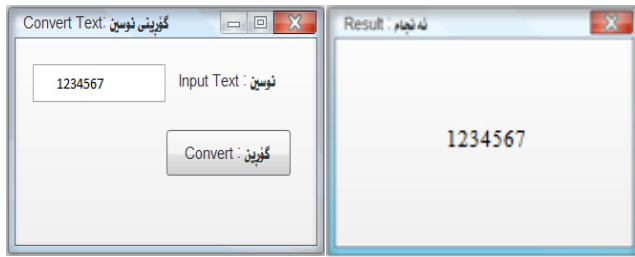


Fig. 10. toKurdishInteger Conversion Screen

Likewise the system automatically converts floating point type number to Kurdish floating point numbers. Below figures demonstrates how the system converts floating points number including float as well as double data types.

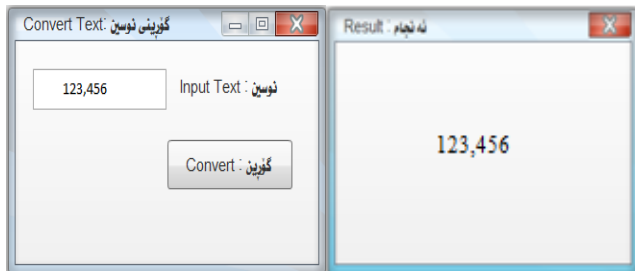


Fig. 11. toKurdishFloat Conversion Screen

As we have mentioned previously, the documentation of JKLP package has been provided for better understanding. The bellow figure shows the overall look of the documentations result. However, users can find the full documentations file in the attached study.

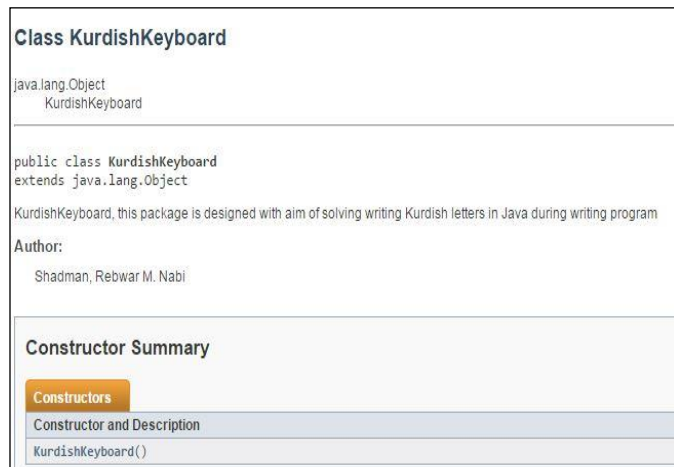


Fig. 12. Summary

V. CONCLUSION

We are glad to announce for Java users especially for Kurdish Java programmers that from now they do not need to be panic when they want to use Kurdish Letters in java programming. Our solution has solved one of major issues which have been around for decades. We developed a java packaged called Java Kurdish Language Package (JKLP) for solving writing Kurdish scripts in Java programming language. JKLP package converts all English letters to Kurdish Letters including texts, symbols numbers which can be integer and/or floating point numbers. The use of our package is thoroughly handy in which you just need to import the package into your existing project and initialize an objects to call the methods. In order to provide better understanding of how to use JKLP package, you can download or look at the documentation which has been provided in the appendix of this study. Finally, we have test the authenticity and the validation of the results of this study by using Junit testing in accordance with currents to standards to give more value to the study.

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Modeling of Compensation in Long-Running Transactions

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Abstract—nowadays, the most controversial issue is transaction in database systems or web services. Specifically, in the area of service-oriented computing, where business transactions always need long periods of time to finish. In the case of a failure rollback, which is the traditional method, it will not be enough and not suitable for handling errors during long running transactions. As a substitute, the most appropriate approach is compensation which is used as an error recovery mechanism. Therefore, transactions that need a long time to complete are programmed as a composition of a set of compensable transactions. This study attempts to design several compensation policies in the long running web transaction especially when the transaction has parallel threads. Meanwhile, one thread in sequence steps of the transaction may fail. This paper also describes and models many different ways to compensate to the thread. Moreover, this study proposes a system to implement creating long running transactions as well as simulating failures by using compensation policies.

Keywords—*transaction; compensation; long-running transaction and interruption*

I. INTRODUCTION

Over the past decades, business transactions have become incredibly important and compound. It is usually referred to as coordinations and communications between multiple partners. In this case, faults are possible at any stage of the transactions. In standard ACID transaction, (with properties of Atomicity, Consistency, Isolation, and Durability), solving and handling faults are done by using a rollback mechanism to provide all or nothing atomicity for transactions [1]. However, rollback is not always satisfactory, especially for transactions needing long life of the time, as widely known as long running transactions (LRT). LRTs usually involve more than one agent. It can be seen that handling faults or errors are difficult and critical, particularly when several partners are involved. Check-pointed is impossible in the LRTs due to their nature, e.g. an email that was sent cannot be unsent. Therefore, LRTs need a comprehensive and separate mechanism to solve such problems. In order to terminate such problems the system need to be designed as a compensation mechanism for those actions that cannot be undone.

Compensation as described in [1] is taken from recovery faults in a business transaction as an action. Consider bookshop as an example, a user buys some books from an online book shop. The system charges the customer's account for the payment of the selected books. Meanwhile, the store of the bookshop knows that one or more books are not on hand at that time. So, to compensate the customer the system can refund the amount already debited and also notified the customer about the situation. Based on this scenario, the importance of the compensation is more reasonable than traditional database rollback. It can be argued that compensations are imperative in terms of handling faults in the long-running transaction. Compensations in LRT are set up for every committed activity.

Recently, many policies have been produced and proposed which used to define approaches of modelling LRTs such as Sagas [3], StAC [4, 5], CSP [6], π calculus [7, 8], Join calculus [9]. There are also some compensation policies as shown in figure 1. Firstly, no interruption and centralised compensation; Compensations only occur, if some transactions abort at the end of the process after all branches were executed [17, 18, 19, 20, 21]. Secondly, no interruption and distributed compensation; Parallel flows compensate as needed without others to complete [17, 18, 19, 20, 21]. Thirdly, coordinated interruption; Parallel branches are free to stop when abortion happens, but compensation is in a centralised way [17, 18, 19, 20, 21]. Finally, distributed interruption which flows are interrupted, if needed, and later on their compensation procedures can be activated independently from the rest of the flows [17, 18, 19, 20, 21]. In the case of compensation, all executed steps have to be compensated in reverse order. The system evaluates all possible compensation forms. It implies we have to compensate the executed steps more than once, each time in a dissimilar manner. For instance, if we have four executed steps, then we have to pay off the system twenty four times for example.

In this study, we model some different policies of compensation in the LRTs. It is obvious that the transaction has many parallel as well as sequence steps and occasionally some of them may fail.

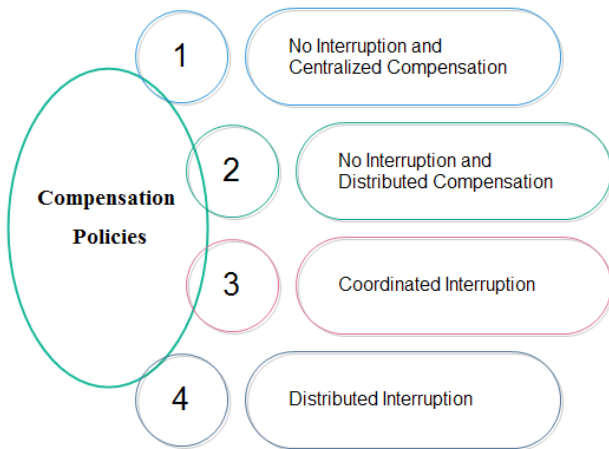


Fig. 1. Compensation Policies

Therefore, defining and applying all possible ways of compensating found to be crucial for the falsified cases. In addition, developing a system to create such transactions and implementing it. Through using compensation policies failures will be simulated.

II. LITRETURE REVIEW

Nowadays, integration between applications and processes is needed in web services on the enterprise level. Web services need a mechanism of transaction that run long-running transactions to address loosely coupled threads, instead of having traditional ACID transaction. BPML [10] by BPMI, XLANG [11] and BizTalk [12] by Microsoft, WSFL [13] by IBM, BPEL4WS [14] by OASIS, these proposals all use long running transactions to describe the activities.

In the sense of web services, interaction and coordination among various services that each one might refer to different companies involve in business transactions. Dealing with faults that occur at any level of such situations in long running transactions is essential but it is difficult. Traditional database mechanism in a long running transaction, such as rollback, is not suitable to rectify faults. For example, cancellation of hotel booking, or hiring taxi, in such cases rollback is not an option as they may need further instructions to handle faults. Usually, in the real world of the long running transaction undoing the transaction is difficult.

Ideally, the idea of compensation was introduced to recover from faults in long-running transactions. As described in [1] compensation is a mechanism to handle the error or change in the plan. When one branch goes wrong in a long running transaction, programmable compensations are to be set to compensate the parts that already completed of the transaction. The transaction concept was defined in [15] where compensation is suited with transactions that correct errors or faults of committed transactions. The later idea of the saga in [16] was defined; to describe long running transaction compensations. Transactions in the saga are divided to a chain of sub-transactions and each of them has its own compensation. Compensation of the committed sub-transaction executes when failures occur of a sub-transaction in the sequence.

There are other approaches that have been modelled in the long-running transaction and have used compensation mechanism. The π -calculus in [22] which is inspired by BizTalk and it contains asynchronous polyadic as an extension π -calculus [23] with the transaction notion. However, the compensation is defined statically in each transaction.

The extension of Join calculus [24] is the cJoin calculus [25] with the primitives for representing transactions. The compensation method is defined statically in this calculus.

StAC language is another model in [26], which is inspired by BPBeans. The language includes the theory of the compensation pair; it is similar to the conception of sagas that is defined by Gargia-Molina and Salem [27]. In StAC, a long-running transaction is a structure of one or more sub-transaction. On the same hand, compensating CSP [28], proposed by cCSP, and Sagas calculi [29] are also based on flow composition, particularly, the embrace a centralized coordination mechanism. However, they have different compensation policies. A calculus named $web\pi$ that described by Laneve and Zavattaro [30]. It is also an extension of π -calculus with a timed transaction structure. $Web\pi^\infty$ is the untimed version of $web\pi$ that was proposed by Mazzara and Lanese [31]. Although, both calculi and $Web\pi^\infty$ are different in some syntax, with following different rules. Namely, nested transactions are surfaced. Therefore, nested failure do not supply by these calculi because the abortion of the sub - transaction cannot be affected by the failure of a transaction.

An extension of SOCK [33] expanded by GuiDi et al. [32], which inspired by WSDL and BPEL. The clear primitives for dynamic handler installation included in this calculus, for example, error and compensation handlers and automatic failure announcement. The correctness properties, namely expected behaviour of a scope cited by them for their calculus, the right termination upon an error, communication and the warranty correct behaviour of fault activation. Our approach is dissimilar in the sense that activation of compensations is obvious to the user. Therefore, addressing the syntax is explicitly and uncomplicated. Related to $web\pi^\infty$, the only identification of the transaction is assumable by them, do not help to ensure this specialty. Another difference is that we utilize a kind system to assure soundness and activation of transaction compensations.

Gray [36] defines compensating transactions as expanding on the same idea, later saga [39], as an added layer on top of ACID transactions. This is debatable: (i) ACID transactions are impossible in the long live transaction, as in the long period of time systems, locking resources are impractical in a highly concurrent world, and (ii) nesting transactions are not supported in the ACID transaction, so working with compensation as counter transaction, transactions can be composed and nested into a saga. According to contemporary literature there are compensable transactions as shown in figure 2.

A. Atomicity

All or nothing; That is, all the changes to the state of the transaction are done, none of them happen. For example, sending money from one account to another, the atomicity

guarantees that, if a debit is made successfully from the first one, the responding later is made to the other account.

B. Consistency

When a transaction either begins and when it finishes its execution, data is in a consistent state. For example, transferring some amount from one account to another, the consistency secures that the transfer amount in both accounts at the same of each transaction.

C. Isolation

The intermediate step of the transaction is hidden to other transactions. As a result, transactions that run simultaneously seems to be serialized. If we consider the example above, the isolation feature assures that the transferred money in one or the others can be seen by another transaction, but it is impossible in both.

D. Durability

Changing to data continues and cannot be undone, once the transaction successfully finishes even in the case of failure. Again, the same example, the durability property warrants that the accounts have been changed, it will not be converted.

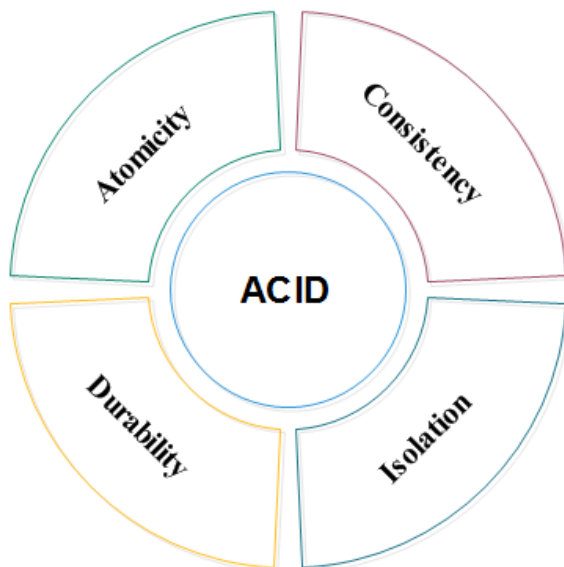


Fig. 2. ACID

III. SYSTEM DESIGN

This chapter will describe and justify the design of the system and how it satisfies the requirements. The diagram below demonstrates the intended design for the system.

As can be seen in the figure 3, the process starts by indicating the numbers of the steps that involve in the transaction digram. Afterwards, each step has a compensation step as well as having a name. Then, users can create a connection between the steps and the diagram according to the connections that have been created by them. Also, there is a command to check the connection, whether there is any incorrect connection such as loop connection and the connection between the step and itself.

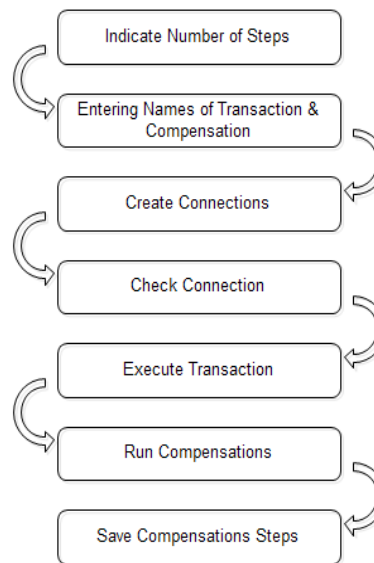


Fig. 3. System Design

After that, by that time a digram is ready to start. Once, the transaction command is pressed, the transaction starts as a forward execution. This means from the beginning to the end (Digram may contain parallel and sequence steps.).

This study demonstrates two policies regarding forward execution. They are distributed interruption and no interruption and distributed compensation. The former stops the whole execution immediately after catching the fault [17, 18, 19, 20, 21]. In contrast, the latter parallel step may continue until the completion [17, 18, 19, 20, 21]. It is unlikely for the transaction digram expecting an error in each step of the whole transaction. Assume that the transaction digram catches the fault and now the system should compensate for these steps that already were executed. This absolutely happens in reverse order. The compensation, which is our main goal, likely to be run many times and each time compensates the executed step in a different order. More importantly, in order to remember which order they compensated, the system has to save the steps that are compensated each time.

IV. IMPLEMENTATION

This section in this study intends to explain the implementation of the compensation of modelling in long running web services transaction. C# programming language has been used to build a system and design a graphical user interface (GUI). The GUI allows users to choose the number of the transaction steps and preparing all the works that are needed for the system. The designed system consists of many different parts. Figure 4 shows the GUI system.

As shown in figure 4, the system contains different parts. In the beginning, the number of the transaction steps should be chosen. Then the names of the steps have to be written as well as the names of the compensations that are programmed for each step. This is used for creating dynamic diagrams which users can indicate and create a connection on demand.

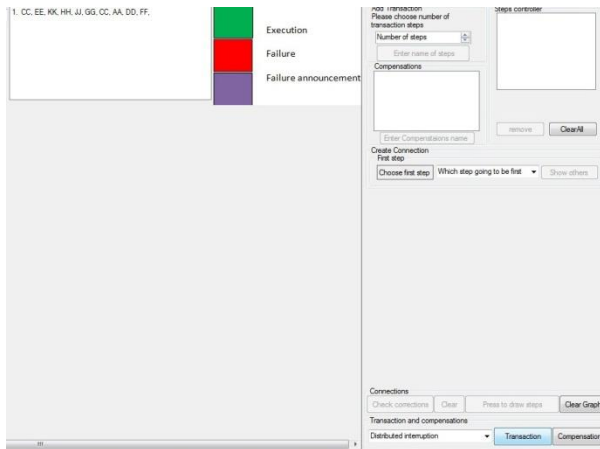


Fig. 4. Compensation of Modelling in web services LRT

Moreover, there is an option that offers two saved diagrams. Hence, users have two options, one they can create diagrams as they want, later is to choose one among these diagrams that are already saved before. Continuously, the second peace is that the users' concern with creating connections between steps. In this section system should prevent users to create incorrect connections such as loop connections or any inconvenient relations.

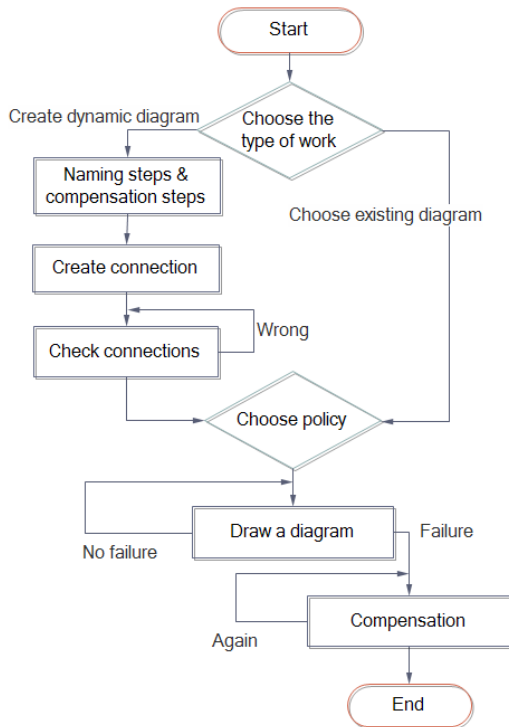


Fig. 5. Compensation of Modelling in web services LRT

V. CASE STUDY

1) Distributed Interruption

As usual the transaction starts from the beginning of the diagram. In the figure 6, there are three parallel branches and two sub-transaction. The system runs the first step and then should randomly choose one of the three branches and then continues to the end. According to the distributed interruption

policy, when the transaction gets an error, all other steps in the diagram are forced to stop the forward execution by the system. Meanwhile, the system should also set up a checkpoint in the step after the last executed steps.

For instance, according to figure 6 the system starts the transaction from A, and the form one among B, C and D randomly. The system continues until it faces a failure in the F step. Then the system immediately asks to stop the forward execution, and is ready to compensate the executed steps. Again, from the diagram, the green color declaration to the successful execution, red color indicates failure occurred and the purple colors are checkpoints as knowledge centre to the previous step that the system stopped because of failure.

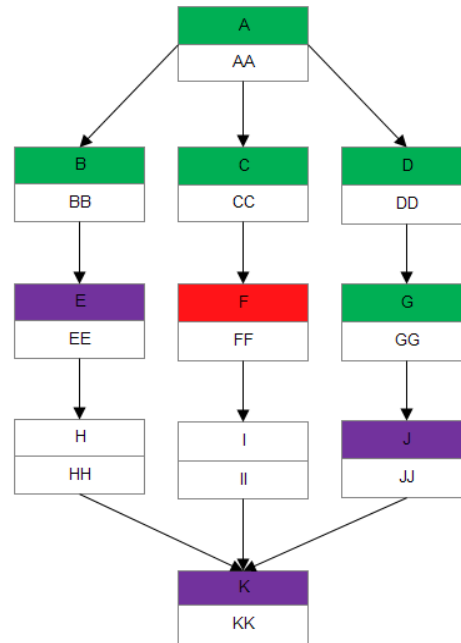


Fig. 6. Distributed Interruption

Simultaneously, in the case of failure, all steps that have been executed are likely to be compensated. Unlike the forward execution, it starts from the reverse order, from the end to the beginning. This study attempts many different simulations. This means the system compensates all executed steps in the different ways. For example, the first step that should start its compensated is the red color, then there are three options GG, CC and BB that should be compensated. Noticeably, we have a lot of probabilities to compensate. To exemplify, the set of compensation should be FF, GG, CC, BB, DD and AA, or FF, BB, GG, CC, DD and AA, or FF, CC, GG, BB, DD and AA, or FF, GG, DD, BB, CC and AA, and so on.

2) No interruption and Distributed Compensation

The transaction starts from the beginning to the end. Similar to the distributed interruption, parallel branches should randomly start the forward execution, but the only and very important point is that the difference in the failure case. This means when the transaction failed, the parallel branches still continue to the step that cannot be continued any longer (the step that set up the checkpoint in.). If consider figure 7 with a bit change from figure 6.

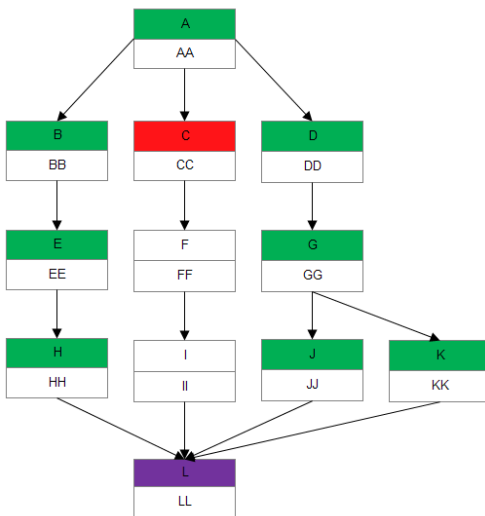


Fig. 7. No interruption and Distributed Compensation

From the figure 7, the transaction starts from step A, then B, C or D to the end. Once the transaction noticed that the failure occurred, unlike the distributed interruption, parallel branches can continue and have not been informed about the issue until reaching the step that cannot be executed (that is the purple color). As shown in the above diagram, green colors indicate the executed steps, red color means failure and the purple one is a point to inform the previous step that the system went wrong and cannot be run any more.

In the term of compensation, similar to the distributed interruption it starts from the back to the beginning. The executed steps should start their compensation as soon as the system failed. There are a lot of compensation simulations such as CC, JJ, HH, DD, EE, KK, GG, DD, BB, and AA, or CC, KK, HH, JJ, GG, EE, DD, BB, and AA, or CC, HH, KK, EE, JJ, GG, FF, DD, and AA, or CC, JJ, KK, GG, KK, HH, GG, EE, DD, BB and AA, or so on.

VI. EVALUATION AND ANALYSIS

The original goal of this project was to build a system to model compensation in the long running transaction. This means when we have a diagram including the parallel and sequence steps, once one of the sequential steps fail the system should compensate to the step and to the other steps as well. It is worth mentioning that the main aim is achieved.

The system is capable of creating dynamic diagrams and users can also choose the organised diagrams. Moreover, the system allows the users to indicate the number of steps and make them being able to create connections between involved steps. Another feature is that the system can check connection. Afterwards, the diagram can be drawn easily. Once it has been done, the diagram can be transacted according to the distributed interruption and no interruption and distributed compensation. The former, the execution of the steps may stop immediately when a failure occurs. Later, the parallel step may continue to the end. It is clear that it is the forward execution. At the same time,, in the case of failure the system has to compensate the failed step and the others that are already executed. In contrast, the compensation starts from the end to

the beginning. As the system programmed, the compensation mechanism should provide different ways to compensate these steps.

VII. CONCLUSION AND FUTURE WORK

In conclusion, the complex business logic, many partners interactions today transactions require a strong framework able to handle efficiently with failures. Furthermore, because of the communication involved with various agents transactions became more and more important and longer, delivering ACID transactions unsuitable. The mechanism of compensation was introduced to transactions, enabling them to manage the new difficulties. Later, the sense of compensable transactions developed and were integrated with more complicated models concerning, amongst other aspects, parallelism, exception handling, transaction composition, and communication amongst activities. Many approaches and models have emerged, providing different solutions to the design issues involved.

In this study, the modelling of compensation in long running transaction system has been developed. The system allows users to choose between diagrams that saved or to create dynamic diagrams. Moreover, if users decide to work on the dynamic one, there are a variety of features such as giving a name to each transaction step as well as for each programmable compensation. Furthermore, some other features are applied, for exemple, creating a connection between transaction steps. Then the system can approve these connections that were made if the system found that connections are not correct, it immediately informs the users to check and change these inappropriate connections. Once, the system knows that the connections are all connected correctly, the diagram can be drawn according to the connections which have been made. Additionally, now the users have a diagram that is ready to run, by running it, the transaction starts as forward execution. In the case of failure, the system should compensate the executed steps. More importantly, by repeating the compensation, users might get a different set of compensation steps.

Overall, it can be argued that some requirements have been achieved whereas some have not. The system was developed successfully, two policies have been implemented and the compensation mechanism was performed. On the other hand, only two policies have been used which means more policies would be more efficient. Similarly,, multi-threading is reliable and efficient to design such a system, however, this system is not created by them. The main goal has achieved.

It can be identified that further researchs need to be made to improve the system more. Firstly, create a bigger diagram by increasing the number of steps or offering more samples. Additionally, create a system that allows a user to indicate all the transaction step's location not only the first step. Secondly, further research could be done to add more compensation policies to the system. In this study, only distributed interruption and no interruption and distributed compensation were used. Therefore, adding more compensation policies may increase the system's reliability and performance. Additionally, using multi-threading in order to make the system more efficient, insuring and working properly. Finally, when users

intend to use the dynamic diagram they have to move steps and order, but in the new research this can be updated and be changed to create diagrams that do not need to be organized.

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A Survey on Digital Watermarking and its Application

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Abstract—Digital communication plays a vital role in the world of Internet as well as in the communication technology. The secrecy of the communication is an essential part of passing the data or information. One noticeable technique is Digital Watermarking. Copyright owners seek methods to control and detect such reproduction, and henceforth research on digital product copyright protection has significant practical significance for E-commerce & E-Governance. In this paper, a survey on some previous work done in watermarking field is presented. Experimentally evaluated algorithms are collected to focus on the wide scope of encrypted digital watermarking for data transmission security and authentication.

Keywords—Watermarking; Watermarking technique; DCT; DWT; LWM; DFRNT; PSNR

I. INTRODUCTION

Digital Watermarking started back in 1979, but it was not until 1990 that it gained popularity. Its full-fledged application began around 1998. No one is credited with founding or inventing the digital watermark, still it is in its growth stages today, and with cases like Napster, it is showing more and more reasons to have digital watermarking.[12] Digital watermarking is the act of hiding a message related to a digital signal (i.e. an image, song, video) within the signal itself. It is a concept closely related to steganography. But unlike steganography, watermarks typically hide very little information and rely on the part on redundancy of the mark to survive attacks such as cropping. Digital watermarks contain information that may be considered attributes of the covering image such as copyright data, and the cover is the object of communication - not the watermark. It is a digital signal or pattern inserted into a digital document such as text, graphics or multimedia, and carries information unique to the copyright owner, the creator of the document or the authorized consumer. Watermarking leaves the original file/ image intact and recognizable.

Watermarks are embedded into images by changing some bits in image representation. The original data (payload) is first encrypted and watermarked in encoder (sender) and then sent to a decoder (receiver) through the internet to be decrypted and extracted. The most important properties of any digital watermarking techniques are robustness, security, imperceptibility, complexity, and verification. Robustness is defined as if the watermark can be detected after media (normal) operations such as filtering, lossy compression, colour correction, or geometric modifications. Security means

the embedded watermark cannot be removed beyond reliable detection by targeted attacks. Imperceptibility means the watermark is not seen by the human visual system. Complexity is described as the effort and time required for watermark embedding and retrieval. Lastly, verification is a procedure whereby there is private key or public function (Dittmann, Mukherjee & Steinebach, 2000). Each of these properties must be taken into consideration when applying a certain digital watermarking technique. Furthermore, the watermark must be either robust or fragile, depending on the application. By "robust", we mean the capability of the watermark to resist manipulations of the media, such as lossy compression (where compressing data and then decompressing it retrieves data that may well be different from the original, but is close enough to be useful in some way), scaling, and cropping, among others.[13]

Current image-based digital watermarks may be grouped under two general classifications: those that fall into the *image domain* and those that fall into the *transform domain*. These techniques of watermarking are applied to graphic images and text. i) *Image domain* is also called spatial domain watermarking that slightly modifies the pixels of one or two randomly selected subsets of an image. However, this technique is not reliable when it is subjected to normal media operations such as filtering or lossy compression (Berghel, 1998). ii) *Frequency domain* watermarking technique is also called transform domain. Values of certain frequencies are altered from their original to the lower frequency levels. These techniques of watermarking are applied to graphic images and text.

The Zhao Koch Algorithm and The Fridrich Algorithm watermark techniques are applied to MPEG videos. The Zhao Koch Algorithm embeds a copyright label in the frequency domain of the video. The watermark can be easily embedded into the video with minimal operation. The Fridrich Algorithm watermark technique is where a pattern is overlaid in the low frequency domain This watermark technique does not include detail information about the owner when the pattern is created and overlaid, so verification using this algorithm is not reliable. This algorithm is resistant to normal media operations whereas the Zhao Koch Algorithm watermark technique is not robust against normal media operations such as scaling or rotation (Dittmann, Stabenau & Steinmetz, 1998).

Few research works on digital watermarking are discussed in the subsequent section.

II. LITERATURE REVIEW

A. Peyman Rahmati, Andy Adler and Thomas Tran (2013) , “Watermarking in E-commerce” [3]. In this paper a technique is proposed to protect digital identity documents against a Print Scan attack for a secured ID card authentication system. The existing PS operation imposes several distortions, such as geometric rotation & histogram distortion on the watermark location which may cause the loss of information. The proposed system removes distortion of the PS operation: - filtering, localization, binarization, rotation and cropping. The proposed authentication system extracts the watermarks inside the ID card’s holder photo, place in the decoder and then checks it out with the ID card personal number. If the extracted watermark and the ID card personal number are the same, the identity of the user / customer will be verified otherwise identity will be denied. A decision function for extracting a bit of the hidden data at the position (i, j) can be written as:

$$d(i,j) = \text{Corr}(B^{W_{ij}(k,l)}, f1(k,l)) - \text{Corr}(B^{W_{ij}(k,l)}, f0(k,l)),$$

where $B^{W_{ij}(k,l)}$ is the block in the watermarked image; $f1, f0$ are the Hadamard pattern used in encoder. The decision function to find the binary hidden data at the position (i, j) is

$$W(i,j) = \text{sgn}(d(i,i)) = \begin{cases} +1 & \text{if } x = 1 \\ -1 & \text{if } x = 0 \end{cases}$$

where $W(i,j)$ is a bit of the binary hidden data at the position (i,j).

B. Neil F. Johnson, Zoran Duric and Sushil Jajodia, “A Role for Digital Watermarking in Electronic Commerce”. [1] In this paper, a new way of categorizing watermark technique through image modeling is discussed. The image modeling called ‘alpha channel composition’ uses gradual mask. Two images with flat mask and gradual mask are used to create watermark that changes gray values of pixel in the image. A method of watermark recovery by applying inverse transformation to the distorted images is shown. The image is watermarked using the version of Digimarc’s PictureMark watermarking filter that is available with Adobe PhotoShop and the image is distorted by applying the Stirmark tool of affine transformation.

C. Swathi.K, Ramudu.K (2014), “Robust Invisible QR Code Image Watermarking Algorithm in SWT Domain”[2]. A binary image is the watermark here. In the frequency domain, the embedding process on QR code image using watermark is performed. The QR code image is decomposed by one level using one dimensional wavelet transformation. To restore the embedded watermark there is no need of the original QR code image. The secret key for embedding and extracting of the watermark is the pseudo-random sequence (P) where each number can take a value either 1 or -1, randomly generated.

$P = \{ p_i, 1 \leq i \leq N \}$, $p_i \in \{-1, 1\}$, where N is the total number of pixels in the watermark image.

The robustness of the algorithm with some attacks such as Salt and Pepper noise, Gaussian noise, and Scaling and Rotating shows extracted watermark with difference magnitude factors. All extracted watermark images contain some visual noise because of the watermark extracting process did not employed the original QR code image

D. Vinita Gupta, Atul Barve(2014), “Robust and Secured Image Watermarking using DWT and Encryption with QR Codes”[4]. In this Paper, algorithm for embedding watermarking is presented by using DWT and encrypted with QR codes. Here cover image is selected and DWT is applied on it. A key K is selected to generate the QR code as secret key. QR code and watermark image is encrypted by using XOR operation. Then the encrypted watermark is embedded into the cover image and inverse DWT is applied on the embedded watermark image. For extraction, simply apply the DWT on the cover image. This algorithm is quite simple because of the use of simple X-OR operation for encryption. This algorithm is suitable on different kind of attacks on watermarked images like JPEG Compression, Poission Noise Attack, Salt & Pepper Noise and Gaussian Noise.

E. M. Kim, D. Li, and S. Hong(2013), “A Robust and Invisible Digital Watermarking Algorithm based on Multiple Transform Method for Image Contents” [5]. In this paper, algorithm for embedding watermarking is presented. Firstly, the original image is compressed into JPEG image and generates the watermark by using the 2D barcode and scrambling. Secondly, JPEG image is decayed into 3 sub-bands: H, V and D by using 2D DWT. Thirdly, the DFRNT(discrete fractional random transform) is performed on the sub-band coefficients. And then, watermark image is embedded into the sub-band coefficient value using quantization technique. Fourthly, the inverse DFRNT and inverse DWT is performed and lastly watermark JPEG image is obtained. The proposed algorithm has good invisibility and extraction performance, and ensures robustness.

F. Arathi Chitla, M. Chandra Mohan(2012), “Authentication of Images through Lossless Watermarking (LWM) Technique with the aid of Elliptic Curve Cryptography (ECC)”[6]. In this paper, a method for authenticating the image using lossless water marking is being proposed that offers high capacity host signal (information) and non-altered image by implementing the elliptic curve cryptography and LSB method. The proposed LWM image authentication technique involves of four processing stages namely, i) information authentication, ii) data embedding on image, iii) information and image recovery, and iv) verification. These four stages are consecutively performed and thus obtained the watermarked and recovered images. A novel lossless watermarking image authentication technique is proposed in this paper. The technique provides high embedding capacities, allows complete recovery of the original host signal, and the retrieved image have high PSNR value than the conventional technique. The PSNR value of the recovered image proved that the image was not altered and the

lossless watermarking procedure was successfully implemented.

G. Shraddha S. Katariya(2012), “Digital Watermarking: Review”,[11] In this paper, the DCT algorithm is selected to do the application test of digital image copyright protection. The experiment proves that DCT-based watermark can well withstand a variety of image processing, and the watermark can survive after compression, cropping, and other attacks. The two dimension discrete cosine transform is encoded on the Windows platform by using Visual C++ program language. The experiment result shows that the digital watermark is non-perceptible; the watermark information can be extracted even if it has been attacked and the expected effect can be achieved.

H. K.Ganesan and Tarun Kumar Guptha(2010), “Multiple Binary Images Watermarking in Spatial and Frequency Domains”[7] In this paper, watermarking scheme provides 24 binary images to be embedded in the frequency domain and also 12 more binary images in the spatial domain. The capacity of the watermark to be embedded in the host image is much greater. Therefore, not only the size of watermark increases, but also it ensures acceptable level of security and imperceptibility. Hence, by using the combinational scheme 36 images in total can be embedded in a single RGB image.

I. Jeng-Shyang Pan, Hao Luo, and Zhe-Ming Lu(2006), “A Lossless Watermarking Scheme for Halftone Image Authentication”[8] . Authentication watermark is a hidden data inserted into an image that can be applied to detect any unauthorized change of the image. Here a block-based method is used. In this, 512×512 halftone images are selected to test the effectiveness of the method. The halftone image is divided into 4×4 blocks. The original watermark, i.e. the hash sequence of image, is computed by the MD5 hash function.

After translating the string into “0-1” sequence, 128-bit digest is obtained. In authentication, the watermark is extracted from the watermarked image, and the hash sequence is computed from the restored image. When the two sequences are equal, it is confirmed that the watermarked image has suffered no alteration. Both of them are equal to the original watermark.

J. Ali Al-Haj(2007), “Combined DWT-DCT Digital Image Watermarking”[9] In this paper, Watermarking is done by embedding the watermark in the first and second level DWT sub-bands of the host image, followed by the application of DCT on the selected DWT sub-bands. The combination of the two transforms improves the watermarking performance considerably when it is compared to the DWT-Only watermarking approach.

K. Ms. Pradnya, B. Rane and Dr.B.B.Meshram, “XML-Based Security for E-commerce Application”. [10] In this paper, they propose XML watermarking in combination with digital signature for security concerns for payment details of the customers. E-commerce based XML data can be easily captured and tempered as it is presented in plain text. Unlike multimedia data, XML data are diverse in nature where watermarking must be invisible and robust. For watermarking SHA- 512 algorithm is used to get hash of transactional data or payment detail (XML file). First, read XML file, partition XML data. Then this Hash value is embedded in the bit positions of XML file. Then digital signature of watermarked XML is computed. For calculating digital signature of watermarked XML data, they have used SHA1 with DSA algorithm.

To measure the quality of a watermarked image, the peak signal to noise ratio (PSNR) is typically used. A glance on the following table reveals the performance and effectiveness of above mentioned research work. The mentioned PSNR values are also given for a comparative analysis.

Application	Algorithm	Performance
Online Secure ID card Authentication, Online passport Authentication System on Ecommerce model [3]	A Block based algorithm using Hadamard Pattern in spatial domain.	Accuracy is of 99% in average to achieve high quality watermarked images. PS distortion model of halftone effect (variable for scanners and printers) is not required. PSNR ratio is approx. 43 DB
Watermarking Technique applied in a QR code image [2]	Robust Invisible QR Code Image Watermarking Algorithm in SWT Domain (frequency domain)	A novel method to embed the QR code into digital images, lowering the JPEG degradation. It can achieve viable copyright protection and authentication. Most robust to attacks in different considerations. PSNR ratio on various images is approx. 47 DB
Colour Image Watermarking encrypted in QR code [4]	XOR operation for encryption of QR code and watermark, after applying DWT on the Cover image	This algorithm is robust and enhances the security. It does not change the quality of watermarked image. Simple XOR operation is used for encryption. PSNR ratio on various images is approx. 62 DB
Digital Image Watermarking for compressed image format (such as JPEG format) used on the web [5]	Robust and Invisible digital image watermarking algorithm through a 2D barcode and scrambling method based on DWT DFRNT transform. The Watermark extraction process is the inverse of watermark embedding process.	PSNR ratio is approx. 40 DB for various images.
Authentication of Medical Images [6]	Elliptic Curve Cryptography (ECC) algorithm, along with LSB data embedding and through Lossless Watermarking (LWM) Technique.	Lossless Watermarking Image Authentication with high embedding capacity with complete recovery of original images. PSNR ratio is approx. 73 DB for various images.

Digital watermarking for images Copyright Protection, Broadcast Monitoring, Content Authentication, Copy and Playback Control etc. [11]	DCT algorithm for multimedia information Security protection, encoded in Windows platform using visual C++ program.	DCT-based watermark can well withstand a variety of image processing, and it can survive after attacks like compression, cropping etc.
Multiple Binary Images Watermarking in Spatial and Frequency Domains [7]	In frequency domain, DCT is applied to each component of the host image and the watermarked image. For spatial domain LSB bits of the pixel values of the host image are changed.	More data can be inserted into an image and extra level of security is achieved by scrambling image before embedding into the host image. For obtaining better result maximum of 30 binary images can be embedded in a single RGB host image. The PSNR values in the DCT domain for each component after applying the corresponding attack on the mark image are ranging between 31 to 51 DB. This scheme is robust against Gaussian noise and JPEG compression up to 90%. The major advantage is the increase in the capacity with less distortion.
Authentication of Military maps, great works of art, medical images etc. using Lossless Watermarking Scheme for Halftone Images [8]	Digital Halftoning on multi-toning images with hash sequence of original image with MD5 hash function.	Fragile watermarking with low quality distortion is introduced to the halftone images. Original image can be perfectly recovered by reverse process of watermarking application. Only secret key is to be saved.
Combined DWT-DCT Digital Image Watermarking [9]	A combined DWT-DCT (Discrete Wavelet Transform and the Discrete Cosine Transform) digital image watermarking algorithm.	Performance evaluation results show that combining the two transforms improved the performance of the watermarking algorithms that were based solely on the DWT transform. Imperceptibility performance was better and the robustness got improved. PSNR for different sub-bands (HL2 HH2) is approx. 97 DB.
Watermarking Technique for payment database security and XML data security [10]	Digital Signature based SHA1 with DSA algorithm	It shows good result for integrity and authentication of data centric, numeric or document centric, verbose XML data. Even after hijacking digital signature, the hackers get watermarked XML data which is unreadable format for humans.

III. CONCLUSION

In this paper, a brief investigation of several works in past decades on digital watermarking (literature review) is done to overview the development of Digital Watermarking Techniques. The encrypted digital watermarking can not only be used for data authentication but also for secured data transmission. The entrusted algorithms with little modification can be used in various fields starting from media industry to medical science and even for e-commerce transaction. The application area of digital watermarking is very wide. And new novel approaches can be sought. The information provided in this paper on this area may help the new researchers to gather knowledge in this domain. Furthermore, researchers can even improve the existing techniques to make them more effective in various novel applications.

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Database-as-a-Service for Big Data: An Overview

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Abstract—The last two decades were marked by an exponential growth in the volume of data originating from various data sources, from mobile phones to social media contents, all through the multitude devices of the Internet of Things. This flow of data can't be managed using a classical approach and has led to the emergence of a new buzz word: Big Data. Among the research challenges related to Big Data there is the issue of data storage. Traditional relational database systems proved to be unable to efficiently manage Big Data datasets. In this context, Cloud Computing plays a relevant role, as it offers interesting models to deal with Big Data storage, especially the model known as Database as a Service (DBaaS). We propose, in this article, a review of database solutions that are offered as DBaaS and discuss their adaptability to Big Data applications.

Keywords—Cloud Computing; Big Data; Database as a Service

I. INTRODUCTION

The volume of data stored in the world has been doubling every two years, and will reach a dazzling 40 billion terabytes (TB) by the year 2020 [1]. By means of comparison, the total size of data that existed in the digital universe in 2000 is 800 million TB, which means that the volume of data will be multiplied by 50 by 2020. This data is generated by various sources: Social Media, E-Commerce, Internet of Things, Sensors, etc. Organizations are also gathering more and more information, for various purposes: analysis to ameliorate their market position and offer better services to their customers, fraud detection, scientific projects like in genomics, legal reasons (for example, Moroccan firms are required by law to store ten years of financial data), etc.

This flow of data, which has been referred to as a flood or a tsunami, can't be managed using a classical approach and has led to the emergence of a new buzz word: Big Data. Almost all major IT leaders invested in various Big Data projects, from Google's BigQuery and Datastore, to Amazon's Elastic MapReduce, to Facebook's Cassandra, Yahoo!'s PNUTS, etc.

Cloud Computing has a leverage effect on Big Data, providing the computing and storage resources necessary to Big Data applications. The inherent characteristics of Cloud Computing, such as elasticity, scalability, automation, fault-tolerance, and ubiquity offer an ideal environment for the development of Big Data applications.

Cloud Computing is an established computing paradigm that gained in importance in the last decade. It refers to the utilisation of storage and computation resources as a utility.

There is a great tendency to opt for using IT as a service. It is estimated that more than 80% of Internet users use Cloud Computing in one form or another, from email services to different business applications as a service, all through data storage, development platforms, etc [2]. This usage percentage is even greater when it comes to companies: In a survey conducted by RightScale in January 2015, 93% of respondent companies confirmed using Cloud Computing [3], which shows that the latter is steadily advancing to become an integral part of companies and individuals use of IT.

Although the emergence of Cloud Computing is relatively new, the idea of delivering computing as a utility dates back to as far as the 1960s, when pioneers like John McCarthy, Leonard Kleinrock, and Douglas Parkhill predicted that, just like water, electricity, or the telephone, computing resources will someday be used as a public utility [4, 5, 6].

There is no consensual definition of Cloud Computing, yet. Many works have proposed their own as discussed in [7, 8]. One of the most cited definition is the NIST's, where Cloud Computing is defined as being a “*model for enabling ubiquitous, convenient, on-demand network access to a shared pool of configurable computing resources that can be rapidly provisioned and released with minimal management effort or service provider interaction*” [9].

Through the plethora of definitions, it emerges that cloud computing has several major characteristics, especially the following:

- Virtualization: physical resources are virtualized in order to optimize their utilization;
- Pooling: multiple users share access to the same pool of virtualized resources. This results in optimizing costs of infrastructure, installation, hosting, and maintenance for providers, who benefit from the economy of scale, and can offer more competitive prices;
- Ubiquity: cloud services are always accessible, anytime, anywhere, and from various computing devices;
- Remote access: cloud services are accessible via a network. It can be the Internet for cloud services that are destined to the general public, or LAN for private ones;
- Automation: users can get the resources they need without having to interact with the provider or require their intervention;

- **Elasticity:** resources are automatically and rapidly increased or decreased to accommodate the workload: when it increases, more resources are added to support it, and when it decreases, superfluous resources are removed. Thus, available resources are directly proportional to workload requirements, ensuring that client applications will have the exact amount of resources needed at any given time;
- **Pay-as-you-go:** users don't need to make any upfront investment in infrastructure, software licenses, etc. They pay only for the resources they consumed, without surplus. Although these resources are multi-tenant, providers strictly measure each client's resource consumption and bill them accordingly. Many billing plans are proposed, some based on the volume of resources used, others on the duration of usage (usually in hours), and others on "commitment" (paying per month, for example).

Cloud Computing's major deployment models are public, private, community, and hybrid (Fig. 1).

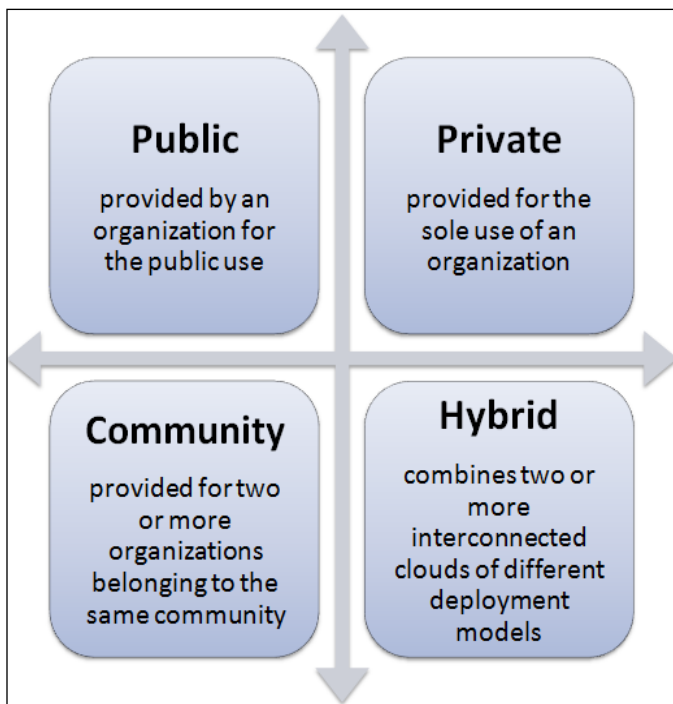


Fig. 1. Cloud deployment models

A Public Cloud is a deployment model in which cloud services are provided via a public network, usually the Internet. Examples include Amazon's Elastic Compute Cloud (EC2), Google's App Engine, and Microsoft's Azure.

A Private Cloud is provided for the sole use of an organization that can either choose to be responsible for managing it or delegate its management to a third-party. The organization can also choose to host it on-premise or off-premise. A variation of this deployment model is the On-Site Private Cloud, where the cloud is hosted and managed by the organization to which it is destined. The main advantage of both models is that there are no restrictions in bandwidth or

resources, since all resources are exclusively intended for the sole use of the organization. It also allows organizations to manage themselves the security aspect of the cloud.

A Community Cloud is a private Cloud that is shared by organizations belonging to the same community, for examples, many departments belonging to the same University, or many companies that want to use a specific application that the provider is going to offer solely to them.

A Hybrid Cloud is composed of two or more of the Cloud models previously presented, interconnected by standard or proprietary technologies.

As for service models, the major ones are Infrastructure as a Service (IaaS), Platform as a Service (PaaS), and Software as a Service (SaaS) (Fig. 2).

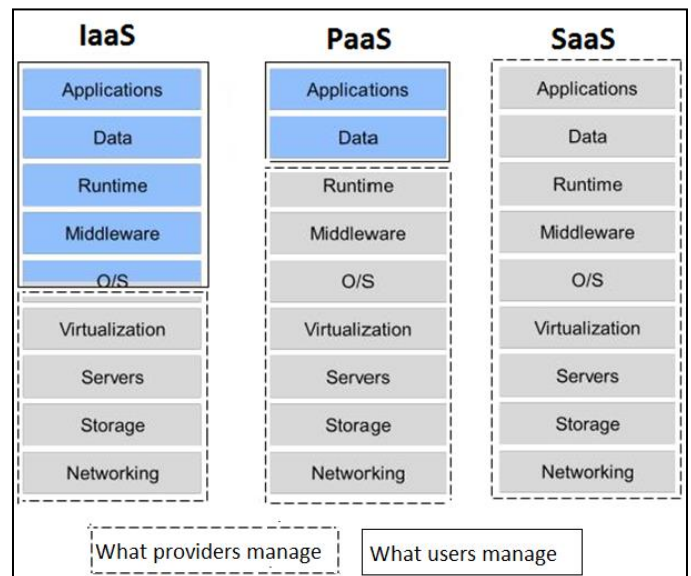


Fig. 2. Components of the main Cloud services models

IaaS provides basic virtualized resources, namely networking (network connections, bandwidth, IP addresses), virtual servers and virtual storage space. This infrastructure will be completed by clients with the various blocks necessary and used to run their applications. The provider manages the underlying infrastructure, while it is up to the user to handle anything other than the hardware part of the architecture. Although IaaS management is majorly incumbent to users, it is the model that satisfies best interoperability and portability needs, since users can compose the various blocks of the infrastructure used [10]. It is also used to build the other cloud service models. Prominent IaaS include Amazon Elastic Compute Cloud (EC2), Google App Engine, and Microsoft Azure.

PaaS is built on top of IaaS by adding a software layer to offer a development environment that can be used by clients to build and deploy their applications. It provides various development tools, such as APIs, for users to develop their applications. Clients can control the deployment and hosting environment of their applications without having to manage the underlying infrastructure. Prominent PaaS include Salesforce's Force.com, Google App Engine, and Microsoft Azure.

SaaS is arguably the most known and used cloud service model. It offers remote access to applications running in the Cloud, through various devices. Users seamlessly access “ready-to-go” applications without needing to invest or manage the underlying infrastructure, to buy software licenses, to handle updates and patches, etc. The provider is responsible for the smooth running of the applications and the maintenance of the underlying infrastructure. Prominent SaaS include Google Drive and Salesforce CRM.

Other service models are increasingly used, among which there is Network as a Service (NaaS), Logging as a Service (LaaS) for log files management, Security as a Service (SECaaS), Recovery as a Service (RaaS), etc. And one of the most promising service models is DataBase as a Service (DBaaS): a report by CISCO showed that if users had the choice to move only one application to the cloud, 25% would choose data storage [11].

Many factors contributed to the rise of Cloud Computing. The widespread use of mobile devices, for example, with their limited storage and processing capacities, led to delegating storage and processing to third parties. The various advantages that come from using the Cloud are also encouraging its rise, especially regarding elasticity, scalability, ubiquity, and cost efficiency, etc.

With Cloud Computing unlocking the barrier of storage and processing resources, developers could focus on their applications without fearing limitation. This led to an expansion of data-intensive applications where datasets are measured in terms of terabytes or petabytes, and the enhancement of Big Data.

We propose, in this work, a review of Cloud Computing solutions for Big Data storage, more precisely the model of DataBase as a Service (DBaaS).

Our paper is organized as follows. We present the definition and characteristics of Big Data in the next section. In section 3, we present some of the storage solutions for Big Data. Section 4 presents a review of several databases as a service, ensued by a discussion of the reviewed features in section 5.

II. BIG DATA: DEFINITION AND CHARACTERISTICS

Throughout the last decade, the increasing use of new technological trends, such as Social Media, E-Commerce, E-Learning, video streaming, etc., resulted in a flood of data. For example, it is estimated that YouTube stores 1 000 TB of new data per day [12], Facebook 600 TB [13], eBay 100 TB [14], and Twitter 100 TB [15], to name but a few. Data thus generated can't be gathered, stored and analyzed easily using traditional storage and analytics tools. This data is referred to as Big Data.

One of the earliest works mentioning Big Data was in the 1990s, where Big Data is referred to as multisource, distributed data that is “*too large to be processed by standard algorithms and software*” [16]. This definition is also adopted by authors in [17], who define Big Data as “*information that can't be processed or analyzed using traditional processes or tools*” and in [18] where Big Data is a set of “*datasets which could*

not be captured, managed, and processed by general computers within an acceptable scope”.

Another definition of Big Data is proposed in [19] as a “phenomenon” that aims “maximizing computation power and algorithmic accuracy to gather, analyze, link, and compare large data sets” to “identify patterns in order to make economic, social, technical, and legal claims”, while authors in [20] talk about “a set of techniques and technologies that require new forms of integration to uncover large hidden values from large datasets that are diverse, complex, and of a massive scale”, a definition that doesn't confine Big Data to the generated data only, but includes both the technology and the architecture related to data.

Cuzzocrea et al. [21] define Big Data as “*enormous amounts of unstructured data produced by high-performance applications*” belonging to various domains, from social media, to e-government, to medical information systems, etc. This data is highly-scalable and requires the applications that handle it to be highly-scalable as well.

Notorious consulting groups also attempted to define Big Data. McKinsey [22] talks about large datasets that can't be “*captured, communicated, aggregated, stored, and analyzed*” using traditional tools, while Experton Group [23] defines it as a “*collection of new information which must be made available to high numbers of users in near real time, based on enormous data inventories from multiple sources, with the goal of speeding up critical competitive decision-making processes*”. Hortonworks defines Big Data as an ensemble of transaction data, interaction data, and observation data [24]. Transaction data is usually structured and stored in SQL databases, and results from applications such as ERP, CRM, transactional web applications, etc. Interaction data results from the interaction between users and applications, or users/applications with each other. This includes logs, social feeds, click streams, etc. As for observational data, it results from the Internet of Things, such as sensors, RFID chips, ATM machines, etc. Gartner [25] defines Big Data as being “high-volume, high-velocity and high-variety information assets that demand cost-effective, innovative forms of information processing for enhanced insight and decision making.” This led to associating Big Data with the 3 Vs: Velocity, Variety, and Volume (Table I).

1) *Volume*: data sets easily reach hundreds of gigabytes, or terabytes. According to IBM, 2.5 million TB of data is created every day [26]. However, volume isn't always quantified by the size of data, but also by the number of transactions, the number of records, the number of files, etc.;

2) *Velocity*: data is generated and delivered at a very rapid pace. Sensors alone, for example, generate thousand TB of data every hour [27], and Wal-Mart is reported to collect 2 500 TB of customer transactions data per hour [28]. This flow of data can be in real time, near real time, batch, or streaming;

3) *Variety*: data comes from various sources, such as social media, blogs, business applications, sensors, mobile devices, etc. This data has different forms. It doesn't always have a specific format or respect a certain schema.

TABLE I. CLASSIFICATION OF THE 3 VS OF BIG DATA

Big Data's V	Classification	Definition
Volume	–	Data is characterized by a large volume, easily reaching Terabytes, or even Petabytes. This data deluge is due to, inter alia, the multiplication of data sources (where data is both human and machine induced), the widespread use of smartphones and applications in an increasingly connected world
Velocity	Real time	Data that is collected and then instantaneously made available for processing or analysis, such as data from GPS or ATM machines
	Near real time	Data that is collected and then is made available for processing or analysis with some delay. An example is data from Geographic information systems
	Batch	Data that is collected at a rather slow rate over a given period time of time, before being processed. Billing systems are an example of batch data
	Streaming	Data that has an interrupted flow, such as data from sensors
Variety	Structured	Data that respects a predefined data model, which makes it easy to collect and store. An example is data stored in relational databases
	Semi structured	Data that doesn't conform with a predefined formal data structure, but that has a certain level of data description, using tags (XML, HTML) or implementing a hierarchy (JSON) [29]
	Unstructured	Data that cannot be represented with a schema, such as text messages, tweets, blog entries, videos, etc.
	Hybrid	Data that combines two or more of the other data types

Other works emphasize on a fourth V, Veracity, to avoid the risk of obtaining a huge amount of poor quality data, or “data garbage” [30, 31, 32]. Authors in [32] define Big Data as “the capture, management, and analysis of data that goes beyond typical structured data” to “any data not contained in records with distinct searchable fields” and characterize it by the four Vs, namely Volume, Variety, Velocity, and Veracity. Thus, it is important to ensure good data quality by verifying its comprehensibility, completeness, and reliability. This represents a challenge because it is not always possible to validate data first-hand, especially as it is highly varied and comes from different sources, and in many cases entered by users.

Gantz et al. define Big Data in [33] as “a new generation of technologies and architectures, designed to economically extract value from very large volumes of a wide variety of data, by enabling high-velocity capture, discovery, and/or analysis”. This definition highlights a fifth V related to Big Data, namely Value, as it is not enough to store a large amount of data, but it is important to analyze it in order to extract value from it.

The NIST introduces another V, Variability, which describes any data change [34]. Thus, Big Data is defined as “extensive datasets - primarily in the characteristics of volume, variety, velocity, and/or variability - that require a scalable architecture for efficient storage, manipulation, and analysis”.

Authors in [35] emphasize on the fact that Big Data has two important sides, namely the storage of large volume of data as well as the analysis of said data, while authors in [36] state that Big Data is a “cultural, technological, and scholarly phenomenon” that originates from the belief that the bigger the volume of data is, the more insight it would provide. It relies on technology and analysis to gather, store, analyze, and identify patterns in large datasets.

Deriving from these various definitions, we propose to define Big Data as large-scale datasets that originate from a plurality of sources at a rapid pace, aren't necessarily structured in a specific schema, can't be stored using typical database management systems, and can't be analyzed using conventional analytics tools.

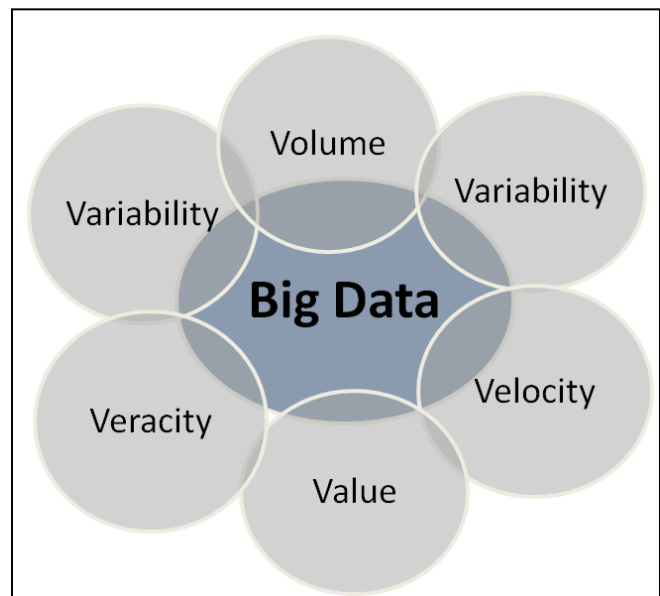


Fig. 3. Some of the V characterizing Big Data

Many factors influence the growth of the Big Data market. Horton identified seven key drivers falling into three categories, namely business drivers, technical drivers, and financial drivers [24]. Among these key drivers, there is the fact that Big Data enables innovative new business models to find adapted solutions to their needs, without requiring big investments in hardware or software, as it runs on commodity computers and offers a multitude of open source software. In fact, Big Data's influence is so tangible in business that some go as far as calling it a “management revolution” that challenges established conceptions of expertise, experience and management practice [37]. Many works have been trying to understand the source and nature of Big Data, and come up with new ways to address the challenges encountered in its different phases, from data collection to archiving, all through storage and analytics. Each one of Big Data's lifecycle's phases called for new solutions to be developed, as shown in Fig. 4.

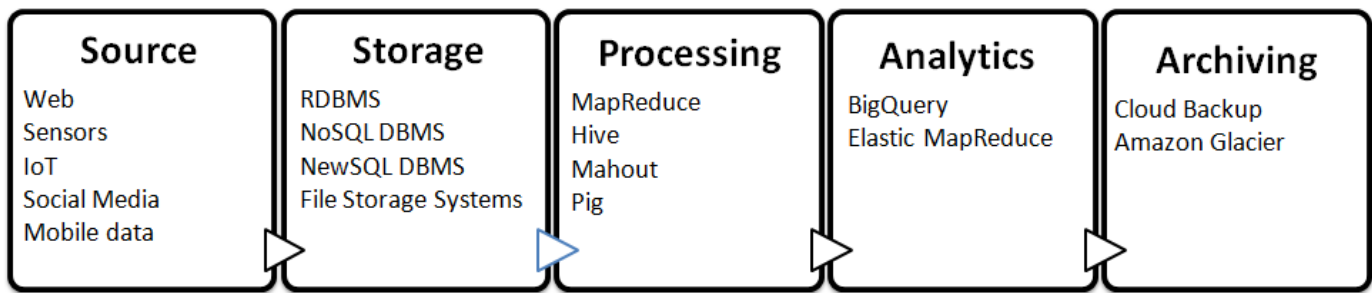


Fig. 4. Big Data's lifecycle

One of the challenges that rose with the growth of Big Data is the storage of the huge volume of generated data. We present in the next section the main storage systems used.

III. BIG DATA STORAGE

One of the challenges that face organizations dealing with Big Data is how and where to store the tremendous amount of data.

The most widespread data management technology is relational database management systems (RDBMS). However, with the rise of Big Data, these RDBMS became unfit for large, distributed data management, especially regarding data Velocity and Variety, since they require data to respect a relational schema before being imported in the database, while Big Data is about managing data of various formats and flow rate (streaming, real-time, etc.). Regarding data Volume, RDBMS are required to be distributed over multiple clusters, sometimes geographically distant. While most proprietary RDBMS scale to large amounts of data, open source ones, such as MySQL and PostgreSQL, are still far behind [38].

First approaches tried to adapt traditional RDBMS by using replication to scale reads, adding a caching layer, using vertical scaling (scale up) or horizontal scaling (scale out) to cope with said volume. Vertical scaling adds more resources to the machine that stores data. This needs powerful machines and can be expensive. Moreover, there is a physical storage limit that can't be exceeded (the current maximum size of a hard disk drive is 8 TB, with the project to reach 10 TB by 2017 [39]). Horizontal scaling, on the other hand, adds more machines to cope with the increasing data volume. Now that the cost of hardware is significantly less than it used to be, it is more interesting to add new servers to the cluster, whenever resources are needed. However, users would ultimately need to shard data across many clusters, which they would have to manage in the application layer.

A real-world example is the expansion of Twitter. Launched in 2006, Twitter knew an exponential growth leading to an average of 500 million tweets per day [40]. In order to manage the expansion of data volume, Twitter had to rethink its architecture, which was relying on MySQL for data storage, when sharding couldn't keep up with the increasing data traffic. This called for developing new adapted solutions used internally by Twitter, such as T-Bird and Snowflake [41]. In general, alternative database solutions are increasingly used in order to provide advantages in terms of performance, scalability, and suitability for Big Data environments. Among

these solutions, there are NoSQL databases, NewSQL databases, and file storage systems like HDFS [50] and GFS [49].

A. NoSQL database systems

The term NoSQL, or Not Only SQL, was first coined in 1998 as the name of a relational database, based on the Unix Shell, and conceived to give better flexibility and optimize the use of resources compared with existing relational databases [42]. It was revived in 2009 with the rise of Cloud Computing and the presentation of Google's Bigtable [43], and has since been generalized to describe databases that model, store, and retrieve data in a different way than traditional relational databases. Many NoSQL databases are well-known today, such as MongoDB, HBase, Facebook's Cassandra, LinkedIn's Voldemort, etc. One of the main features of NoSQL databases is that they are schema free, which means that the structure of data can be easily and quickly modified without needing to rewrite tables. This aims to overcome the inflexibility of traditional relational databases schemas. And while many NoSQL databases don't implement certain relational functionalities, such as JOINS, ordering, and aggregation, many offer support for SQL-like querying.

While relational databases permit handling data storage and management simultaneously, especially with implemented SQL-querying interfaces, NoSQL databases handle them separately. Data storage is done according to the adopted data model (key-value, document, etc.) with a primary focus on scalability. Data access is done using APIs. This renders NoSQL databases flexible for data modelling and easy for application development and deployment updates [44].

Relational databases guarantee ACID (Atomic, Consistent, Isolated, and Durable) transaction properties. However, CAP theorem (Fig. 5) states that at most two out of the three properties (Consistency, Availability, and Partition tolerance) can be achieved simultaneously in distributed environments [45]. While RDBMS do well on Consistency and Availability, they don't scale well. The main idea behind NoSQL databases is to loosen up on one of these two properties, namely Consistency and Availability, in order to enhance scalability. They provide what can be called BASE (Basically Available, Soft state, and Eventually consistent) [46] properties, in contrast with ACID. NoSQL database systems differ in which of the two properties they loosen, and how much they do loosen it. Many however provide eventual consistency to ensure high scalability and availability.

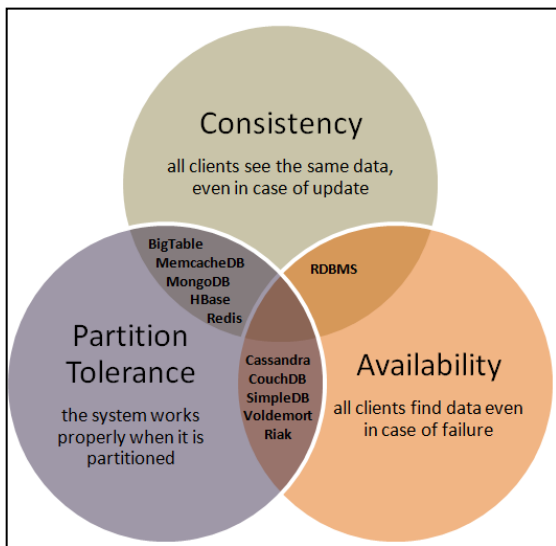


Fig. 5. The CAP theorem

NoSQL databases have many data models: Key-Value, Document, Column, and Graph, as shown in *Table II*.

Key-value databases store data as a collection of (*key, value*) pairs where a unique identifier, *key*, is used to access and retrieve data. They are schema-free, as values are independent from each other, with no restriction on their nature. As data is completely opaque to the system, the only way to access and retrieve it is by using the unique *key*. They support basic insert, read, and delete operations. Most are persistent while others like Memcached cache data in memory. Notorious examples include Redis, Memcached, and DynamoDB.

Document databases store data as documents that are based on a specific encoding (JSON, BSON, XML, etc.) and identified by a unique “ID”. Document databases being schema-free, documents can store attributes of any kind. Most document databases generally support more complex data (such as nested documents) and offer more indexing and

querying functionalities, but relatively less performance, than Key-Value ones.

Column databases are modelled after Google’s Bigtable [43]. They store data using tables (columns and rows) but without any association between them. Columns consist of a unique identifier, a value, and a timestamp used for versioning. They are grouped in column families that have to be predefined, which affects flexibility.

Graph databases store data nodes interconnected with edges where each node and edge consists of key-value pairs. This allows graph databases to store not only data, but also relationships between data nodes. They are the tool of choice when dealing with heavily linked data. Some examples include Neo4J database, which supports ACID properties, and OrientDB.

Although they differ in their data model, all NoSQL databases allow a relatively simple storage of unstructured, distributed data and achieve high scalability. They are best adapted for applications that don’t use a fixed schema, or don’t require ACID operations, and for intensive read and update OLTP workloads [47].

B. NewSQL database systems

NewSQL originated from the affirmation that the relational model can be implemented to scale by retaining its key aspects and removing some of the general purpose ones [48]. NewSQL databases aim to answer Big Data storage needs, especially regarding volume and scalability, while providing the traditional functionalities of relational databases, especially regarding ACID transactions, querying operations such as JOINS and aggregations, etc. They are an attempt to realize the three properties featured in the CAP theorem, proving that Consistency and Availability can be achieved simultaneously in distributed environments.

NewSQL databases provide an SQL query interface, and clients (users and applications) interact with them the same way they interact with relational databases. They manage read/write conflicts using non-lock concurrency control [48].

TABLE II. NOSQL DATA MODELS

Data model	Definition	Use case	Advantages	Limitations	Examples
Key-Value	Stores data as a collection of (key,value) pairs	Applications with only one kind of object where search is performed based on one attribute	Simple to use	Relationships between data must be explicitly managed in the application layer	Memcached Redis DynamoDB
Document	Stores data as encoded documents	Applications with many kinds of objects where search is done on multiple attributes	Management of complex data structures	Relationships between data must be explicitly managed in the application layer	CouchDB MongoDB
Column	Stores data as columns consisting of a key, a value, and a timestamp	Applications with many kinds of objects where search is done on multiple attributes and that need data to be partitioned both horizontally and vertically	Allows high throughput and low latency	Less flexibility	Bigtable HBase Cassandra
Graph	Stores linked data as graphs	Applications that handles heavily connected data (social networks, location based services, etc.)	Seamless manipulation of graphs	Relatively high complexity and less scalability	Ne04j GraphDB OrientDB

Many NewSQL solutions extend existing relational databases to support high scalability, like Infobright, TokuDB, and MySQL cluster NDB, which are all built on MySQL. Other solutions retain existing relational databases and add a middleware for achieving high scalability through shading or clustering, such as ScaleArc, ScaleBase, dbShards, etc. There are also solutions that were developed from scratch to provide relational features in distributed environments, such as NuoDB.

NewSQL databases are relatively new compared to NoSQL ones. They are most adapted to use case scenarios that call for relational databases with more scalability. They try to combine the advantages of both relational and NoSQL databases, as detailed in Table III.

TABLE III. COMPARISON OF RELATIONAL, NOSQL, AND NEWSQL DATABASES

Feature	Relational databases	NoSQL databases	NewSQL databases
Relational schema	Yes	No	Yes
SQL Querying	Yes	No	Yes
ACID transactions	Yes	No	Yes
Big Data compatibility	No	Yes	Yes
Availability	Yes	Yes	Yes
Strong Consistency	Yes	No	Yes
Scalability	No	Yes	Yes

C. File Storage Systems

File storage systems are another solution to deal with large volume of data in distributed environments. The major ones are Google File Storage (GFS) [49] and Hadoop Data File Storage (HDFS) [50].

GFS is a scalable distributed file system developed by Google to meet the needs of its large distributed data-intensive applications [49]. It is designed for environments that are prone to failures, that manipulate huge data files by frequent read/append operations, and that need to process data in batch rather than in real-time. Thus, it is highly fault-tolerant and reliable, and emphasizes on high throughput rather than low latency.

GFS has a master-slave architecture (Fig. 6), a typical cluster consisting of one master and many chunkservers to which clients access directly after consulting the master. The master divides each file into 64 MB chunks and manages the mapping and replication of said chunks through the different chunkservers.

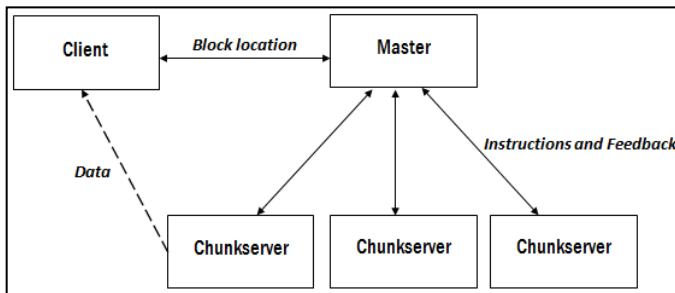


Fig. 6. GFS architecture

GFS maintains multiple replicas of each file, which leads to higher reliability and availability.

HDFS [50] is an open source implementation of GFS. It is part of the Apache Hadoop, an open source framework for distributed storage and distributed processing of large data sets. The biggest clusters implementing Hadoop are composed of 45 000 machines and store up to 25 petabyte of data [51].

HDFS is one of the four modules composing Hadoop, which are Hadoop commons, Hadoop YARN, and Hadoop MapReduce, the open source implementation of Google’s Map/Reduce for the parallel processing of large distributed data.

HDFS is implemented based on the fact that moving computation is cheaper than moving data, providing interfaces to client applications to move where data is stored. Like GFS, HDFS has master-slave architecture (Fig. 7) consisting of a single master node, NameNode, and a slave for each node in the cluster, DataNode.

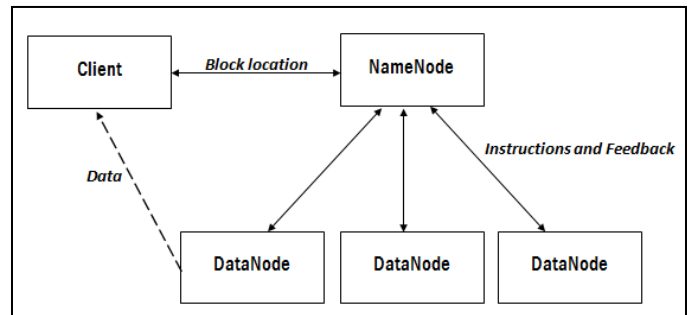


Fig. 7. HDFS architecture

The NameNode is the coordinator of HDFS. It divides files into fixed-sized blocks and maps them to DataNodes, and client applications consult it to know where to access data. The DataNode manages data storage in the node where it is installed. It can also create, delete, and replicate blocks when instructed by the NameNode.

The adoption of NoSQL, NewSQL and File Storage systems is mainly driven by six key factors, regrouped in the acronym SPRAIN [52]. These key drivers, which are the weak points of traditional RDBMS, are Scalability, Performance, Relaxed consistency, Agility, Intricacy, and Necessity. And while these new database systems are becoming the tool of choice to meet the demands of Big Data applications, it can be complicated and costly to run and manage them, especially at scale. One solution is to move them to the Cloud in order to take full advantage of the elasticity, scalability, availability, and performance of the latter, and meet the ever-growing storage and processing requirements of Big Data applications. And one of the currently most adapted Cloud Computing models to Big Data storage requirements is DataBase as a Service (DBaaS), as it can combine many of the aforementioned storage systems to offer scalable, on-demand, pay-as-you-go storage resources to organizations without any upfront investment.

We present, in the next section, a review of several DBaaS and discuss their suitability for Big Data storage.

IV. DATABASE AS A SERVICE (DBaaS) FOR BIG DATA

An ever growing number of companies found themselves swamped with the large amount of data generated and stored for different purposes (user based preference suggestions, business analysis...). Storing and retrieving data becomes a costly and complex operation, involving investments in infrastructure and database managers. It is only normal then that the question of outsourcing data was one of the earliest to surface with the emergence of Cloud Computing, which led to the DataBase as a Service (DBaaS) model.

DBaaS can be simply defined as “a paradigm for data management in which a third party service provider hosts a database and provides the associated software and hardware support” [53]. Companies using this model outsource all database management operations, from installation to backups, to the provider, and focus on developing applications. They can access their databases instances on-demand, using querying interfaces or programming tools.

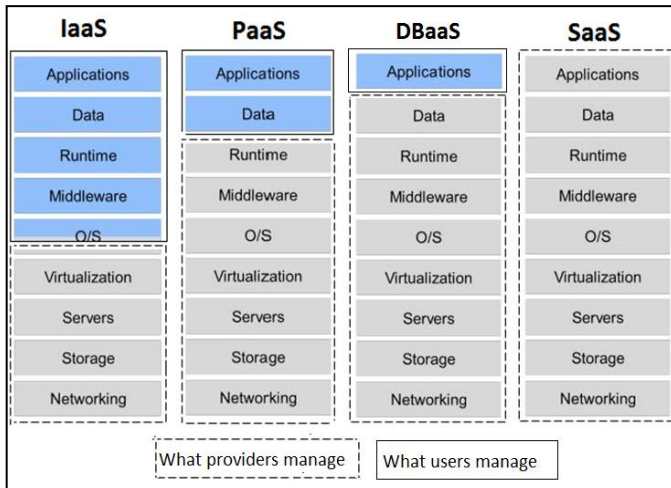


Fig. 8. DBaaS components

The increasing use of Cloud Computing, and especially SaaS, called for rethinking the persistency layer. The inherent characteristics of cloud computing, such as elasticity, scalability, self-service, and easy management make traditional RDBMS not fully adapted for applications that run in cloud environments. Early solutions tried extending existing DBMS to support high-scalability, but it only led to complex solutions with poor performance [54]. Leader IT operators, such as Google, Yahoo!, and Facebook, chose to implement their own data management solutions, respectively Bigtable, PNUTS, and Cassandra. Various other databases provided as DBaaS were developed from scratch to integrate the advantages of the cloud, with the exception of few providers who offer established relational or NoSQL databases, such as MySQL, PostgreSQL, MongoDB, and Redis, as a service.

Database as a Service (DBaaS) is one of the Cloud Computing models that is most suitable for Big Data. In this model, it is possible to use a database as a service and benefit from the high-scalability and storage capacity offered by the Cloud, without having to install, maintain, upgrade, backup or manage the database or the underlying infrastructure.

DBaaS is a different concept from the concept of cloud databases, which is beyond the scope of our paper. In this concept, users can either upload their machine image, with the database installed, to the cloud infrastructure or use a ready one offered by the provider. In both scenarios, the various database management operations are incumbent to users. Datawarehouse Cloud solutions are also beyond the scope of this paper.

We propose to review some of the most prominent databases that are DBaaS and discuss their adaptability to Big Data uses.

A. Cloud Bigtable

Cloud Bigtable is a DBaaS based on Bigtable [43], a highly-scalable, distributed, structured, and highly-available column database developed by Google that has been used internally since 2003 to store the data of numerous Google projects (Google Finance, Google Analytics, Google Earth, etc.). Bigtable was made publically available as Cloud Bigtable in May 2015 [55].

Bigtable stores data in tables, which are “sparse, distributed, persistent sorted” maps. [43]. These tables are sharded into tablets containing blocks of adjacent rows. Each cell is referenced by three dimensions: a row key, a column key, and a timestamp.

A row key is an arbitrary string and is the unit of transactional consistency in Bigtable. Rows with consecutive keys are grouped into tablets, which are the unit of distribution and load balancing. A column key is also an arbitrary string, and column keys are grouped into columns families, the unit of access control. Timestamps are used to manage data versioning. A cell can store different versions of the same data, each referenced by a timestamp. Older data is garbage-collected depending on the user’s specifications.

Bigtable relies on Google File System (GFS), a scalable distributed file system presented in Section 4, for storing data in SSTable [43] file format. An SSTable is a file of key/value string pairs that is sorted by keys. It is used to map keys to values. Bigtable also uses Chubby, a highly-available and persistent distributed lock service, for synchronizing data access [56]. A Chubby service has four replicas and one master replica. The latter is used to serve requests. Bigtable architecture is composed of one master server, many tablet servers, and a library, as shown in Fig. 9.

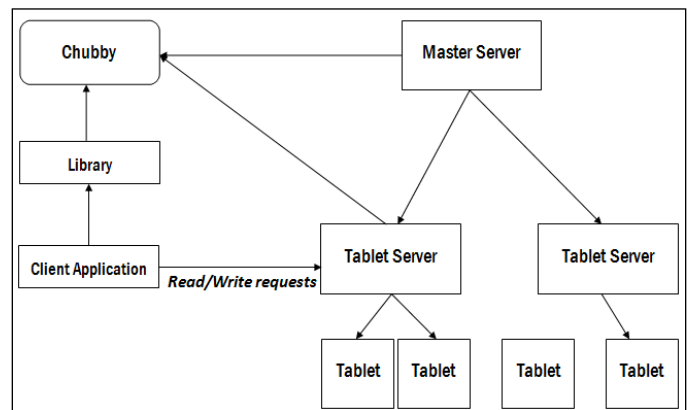


Fig. 9. Bigtable architecture

The library is linked to client applications and is used to retrieve the location of tablets. The master server performs many tasks: assigning tablets to tablet servers, load balancing, detecting new or expired tablets, detecting schema changes, and GFS garbage collection. A tablet server is responsible for managing a set of tablets, receiving read /writes requests from client applications, serving client requests that are directed to the tablets it manages, and splitting tablets when their size exceeds 1 GB.

Each tablet is assigned to one tablet server at a time. Tablet servers use Chubby to obtain an exclusive lock on the tablets they manage. The master server consults Chubby to discover tablet servers.

While being manipulated, tablets are stored in memory in a buffer called memtable. When the size of a memtable reaches a certain level, it is stored as an immutable SSTable in GFS. Tablet servers perform write operations on tablets in memtable, and read operations on views obtained from merging SSTables and the memtable.

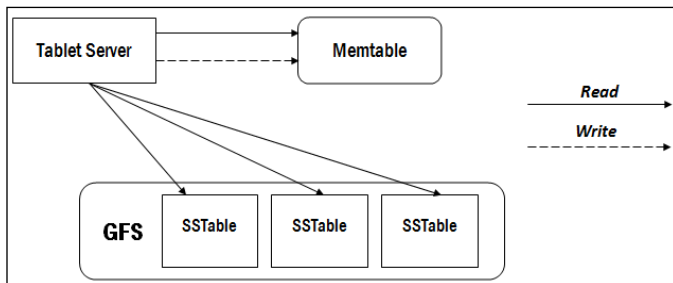


Fig. 10. Management of Read and Write operations

Bigtable maintains a high level of consistency. Reads are strongly consistent, since SSTables are immutable. As for writes, memtables perform a row copy each time there is a write operation in a row, ensuring that updates are seen by reads.

Client applications can connect to Cloud Bigtable using the Cloud Bigtable HBase client. The latter supports HBase shell, which can be used to perform queries and administrative tasks.

Cloud Bigtable was designed for Big Data applications that handle terabytes of data in clusters composed of thousands of nodes. Google recommends it for applications where the volume of data exceeds 1 TB. For Big Data applications with less than 1 TB data volume, Google recommends another solution, namely Cloud Datastore.

B. Cloud Datastore

Cloud Datastore is a NoSQL, schemaless, highly-scalable, and highly-reliable database for storing non-relational data developed by Google as a part of the App Engine. The main motivation for its development is to answer the need for high-scalability that couldn't be met by traditional relational databases. It supports basic SQL functionalities, including filtering and sorting. Other functionalities like table joins, sub queries and flexible filtering are not supported. Cloud Datastore is based on another Google's solution, namely Megastore, which is built on Bigtable. Thus, Cloud Datastore architecture is as shown in Fig. 11.

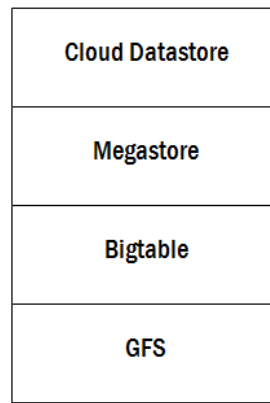


Fig. 11. Cloud Datastore architecture

Megastore [57] is a distributed data store that combines the scalability of NoSQL databases and some key features of relational databases, especially in terms of consistency and ACID transactions. It allows users to define tables just like in traditional SQL databases, and then maps them to Bigtable columns. It is used by more than 300 applications within Google [58].

Megastore ensures strong consistency. It replicates data across multiple geographically distributed datacenters using an algorithm based on a distributed consensus algorithm, Paxos [59], for committing distributed transactions. It also implements two-phase commit (2PC) [60] for committing atomic updates. Unlike 2PC, Paxos doesn't require a master node for committing transactions. Instead, it ensures that only one of the proposed values is chosen and, when it is, that all the nodes forming the cluster get the value. Thus, all future read and/or write access to the value will give the same result.

For each new transaction, Megastore identifies the last transaction committed and the responsible node then uses Paxos to get a consensus on appending the transaction to the commit log. Megastore is built on Bigtable to overcome the difficulty to use in applications that have relational schemas, or that need to implement strong consistency [86]. An amelioration to Megastore is Spanner [86], a highly-scalable, globally-distributed, semi-relational database where queries are done in an SQL-like language and offers better write throughput. Though Spanner is not offered as a service to developers, it is used internally by Google as the backend of F1, Google's distributed RDBMS supporting its online ad business. However, there is a project for building an open source version of Spanner, CockroachDB.

Cloud Datastore relies on Megastore to support transactions, ensuring strong consistency. The entity data, which is the equivalent of a row in relational databases, is written in two phases: the commit phase and the apply phase. In the commit phase, data is recorded in the transaction logs of a majority of replicas. It is also recorded in the transaction logs of all replicas in which it was not recorded and that are not up-to-date. In the second phase, the entity data and its index rows are written in each replica.

Cloud Datastore also relies on Bigtable's automatic sharding and replication to ensure high-scalability and

reliability. Performance is ensured by reducing lock granularity and allowing collocation of data to minimize the communication between nodes.

In Cloud Datastore, client applications perform queries and manipulate data using APIs, third-party implementations of the Java Data Objects (JDO) and Java Persistence API (JPA), or third-party frameworks such as Objectify, Slim3 or Twig.

Google intends to prove, with Cloud Datastore, that scalability can be achieved while keeping some features of traditional relational databases, especially transactions, ACID semantics, schema support, etc. It thus provides a highly-scalable and reliable cloud database that is adequate for Big Data applications that need to implement strong consistency.

C. Cloud SQL

Cloud SQL is a fully-managed, highly-available MySQL database hosted in Google's cloud and offered as DBaaS. It allows users to easily create, run, and manage MySQL databases in Google's infrastructure, with a promise of 99.95% uptime SLA [61]. It is simple to use and gives users the possibility to control the geographical location where their data is stored, the RAM capacity they need (ranging from 0.125 to 16 GB), the billing plan they prefer (based on the number of hours the database is accessed or based on the number of days the database exists), the backup frequency, the replication mode, the connection encryption mode, etc. Many companies opted for migrating their data into Cloud SQL, such as CodeFutures and KiSSFLOW.

Cloud SQL is distributed, and it replicates data across multiple datacenters in order to be fault-tolerant, using both synchronous and asynchronous replication. It supports all MySQL features with some exceptions (user defined functions, LOAD_FILE function, installing and uninstalling plugins). It is accessible via MySQL clients, standard MySQL database drivers, App Engine applications written in Java or Python, and third-party tools such as Toad for MySQL.

In Cloud SQL, the maximum size of an instance is 10 GB, with a total size limit of 500 GB. Moreover, it doesn't scale automatically, but it is up to the user to handle scalability, and it is not adapted to applications where data schema changes frequently. This makes Cloud SQL unsuited for Big Data applications.

D. Cloudfant

Cloudfant [62] is a scalable, distributed, NoSQL database as a service provided by IBM, with the assurance, through SLAs, of uninterrupted, highly-performant access to data. Cloudfant's infrastructure consists of over 35 datacenters distributed in more than 12 countries all over the world. Data is stored in server nodes, grouped into clusters that can either be multi-tenant or single-tenant. Cloudfant also offers users the possibility to deploy it on-premise, or to select other hosting providers such as Rackspace, SoftLayer, and Microsoft Azure. This is done in the optic of bringing Cloudfant near to users' data, in the case where it is already hosted in the cloud. As for the billing, it is adaptable to the growth of the user's applications, offering a "pay-as-you-grow" billing plan.

Cloudfant is interoperable with many open source solutions, which enhances its capabilities and features, as shown in Fig. 12.

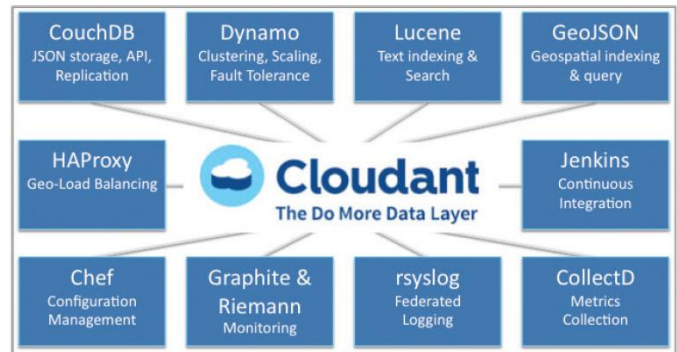


Fig. 12. An overview of Cloudfant interaction with various open source solutions [62]

Cloudfant is based on Apache CouchDB, with some additional features regarding data management, advanced geospatial capabilities, full-text search, and real-time analytics. It stores data as JSON documents (Fig. 13), which is a lightweight data-interchange format that is built on a collection of name/value pairs, and an ordered list of values.

```
{ "menu": {
  "header": "SVG Viewer",
  "items": [
    { "id": "Open",
      { "id": "OpenNew", "label": "Open New",
        null,
        { "id": "ZoomIn", "label": "Zoom In",
          { "id": "ZoomOut", "label": "Zoom Out",
            { "id": "OriginalView", "label": "Original View",
              null,
              { "id": "Quality",
                { "id": "Pause",
                  { "id": "Mute",
                    null,
                    { "id": "Find", "label": "Find..."},
                    { "id": "FindAgain", "label": "Find Again"},
                    { "id": "Copy",
                      { "id": "CopyAgain", "label": "Copy Again"},
                      { "id": "CopySVG", "label": "Copy SVG"},
                      { "id": "ViewSVG", "label": "View SVG"},
                      { "id": "ViewSource", "label": "View Source"},
                      { "id": "SaveAs", "label": "Save As"},
                        null,
                        { "id": "Help",
                          { "id": "About", "label": "About Adobe CVG Viewer..." }
                    ]
                ]
            ]
          ]
        ]
      ]
    ]
  }
}
```

Fig. 13. An example of JSON-formatted documents

JSON documents are accessed using an HTTP-based RESTful API. Querying is done using Cloudfant query, a declarative system based on MongoDB's declarative query. Cloudfant assigns a unique identifier to each JSON document and uses a MapReduce-based framework to query data. Users write MapReduce functions in JavaScript, where the Map function defines which JSON documents are concerned by the Reduce function that specifies the operations to perform. Then Cloudfant distributes the MapReduce functions to all nodes forming the cluster. It is noted that Cloudfant allows MapReduce functions to be "chainable", meaning that the output of a MapReduce job can be used as input for other MapReduce jobs in the chain.

Data distribution is done by multi-master replication, ensuring a high fault-tolerance, and reducing latency by connecting users to data that is geographically closest. Users can replicate data not only through all nodes forming the cluster, but also to CouchDB, being able to benefit from an open source data storage solution to increase their datacenter size.

Cloudant is adapted to Big Data uses, especially for web, mobile, and the Internet of Things [63]. It is also suitable for applications that deal with unstructured data or that need to synchronously replicate data across multiple datacenters.

E. MongoLab

MongoLab is a fully-managed, highly-performant, highly-available MongoDB database offered as DBaaS that runs in major cloud infrastructures: Amazon WS, Google Cloud Platform, Rackspace, and Windows Azure, etc. It is also possible to integrate it with users' applications that run on other PaaS providers' platforms, like AppFog, Heroku, OpenShift, etc.

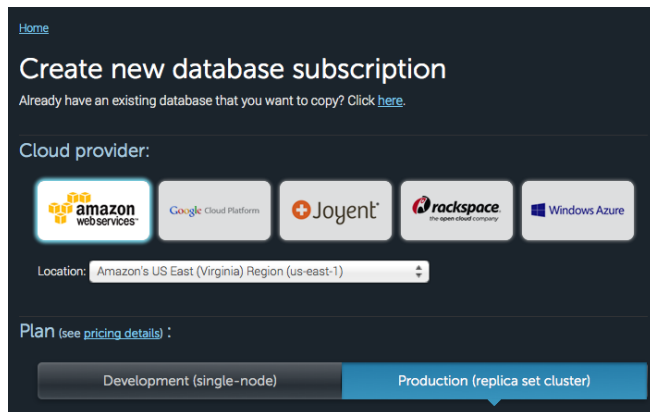


Fig. 14. MongoDB control panel

MongoDB is a schema-free, scalable document database that offers, along with the basic CRUD functions of traditional relational databases, many features such as indexing, aggregation, session-like data expiration management, native support of geo-spatial indexing, etc. Other features specific to relational databases, such as JOINS, are not supported.

MongoDB stores data as BSON documents, a lightweight, binary interchange format based on JSON. BSON represents data efficiently, optimizing storage space and scan speed, and rendering encoding and decoding data simple and fast. Data access, data requests and background management operations are performed by *mongod*, the primary daemon process of MongoDB.

Users can browse their data stored in MongoLab via the management portal, or the MongoDB shell, which is an interactive JavaScript shell. Applications can be connected to the MongoLab databases using a MongoDB driver, or MongoLab RESTful APIs.

MongoDB defines its own query language. Users can perform ad hoc queries using two functions like *find()* and *findOne()* that return a subset of documents. Queries can be performed with complex criteria (such as ranges or negatives),

conditions, sorting, embedded documents, etc. It is also possible to use indexing, like in relational databases, which allows performing faster queries. In addition, MongoDB offers a wide range of commands to be used to manage servers and databases.

MongoDB handles replication using a master-slave strategy. Users define a replica set, which is composed of a primary server and many secondary servers. The primary server gets the requests from applications and users, and secondary servers store copies of the data contained in the primary server. This way, if the primary server becomes unavailable, one of the secondary servers is chosen by its peers to replace it. MongoDB also offers an interesting feature, slave delay, which sets a secondary server to lag by a predefined number of seconds to allow retrieving an earlier version of damaged data.

Scalability in MongoDB is ensured by autosharding. Mongos, MongoDB's routing service, is used to keep track of the location of data in the different shards. Applications connect to Mongos and send their queries the way they'd do with a stand-alone MongoDB instance, as shown in Fig. 15. This allows MongoDB to handle higher throughput in read and write operations than what a stand-alone instance can handle [64].

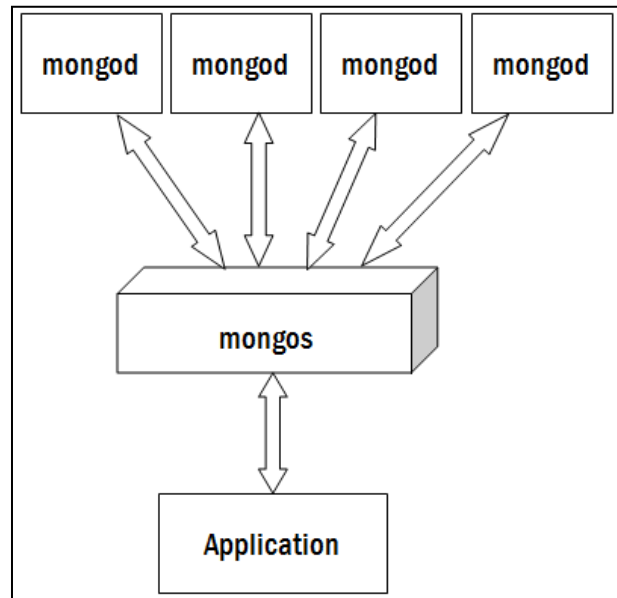


Fig. 15. Access by applications to sharded data in MongoDB

MongoDB's design makes it suitable for storing large volumes of heterogeneous, evolving collections of data.

F. Morpheus

Morpheus is a fully managed, highly-available DBaaS that provides access to SQL (MySQL), NoSQL (MongoDB), and cache (Redis) databases. It also offers a fully managed access to Elasticsearch, a full-text search engine.

As mentioned above, Morpheus offers a fully managed access to four databases. MongoDB and MySQL have been presented in previous chapters. We will present Elasticsearch and Redis.

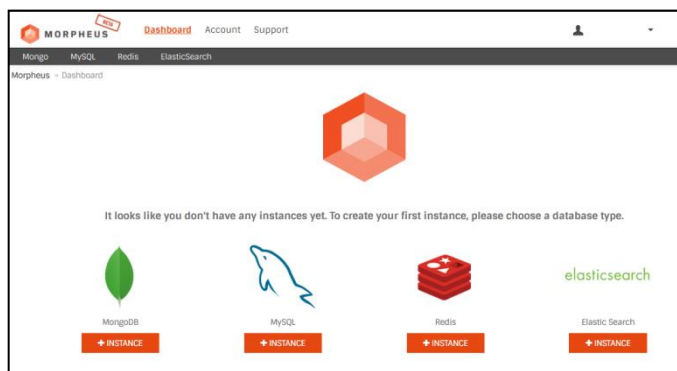


Fig. 16. Available databases in Morpheus⁴¹

Elasticsearch is an open source distributed, scalable, highly-available full-text search engine. It is built on Apache Lucene, an open source library for data retrieval.

Redis is an open source key-value cache and store that keeps data in memory for faster treatment, handling over 100 000 read/write operations per second [65]. Redis can also store data on hard disk asynchronously using snapshots or append-only logs.

Morpheus allows users to easily select one of the available databases and create an instance with a size ranging from 1 to 200 GB, as shown in Fig. 17. It supports many versions of each database and gives users the possibility to select one. Users can create many instances using disparate databases.

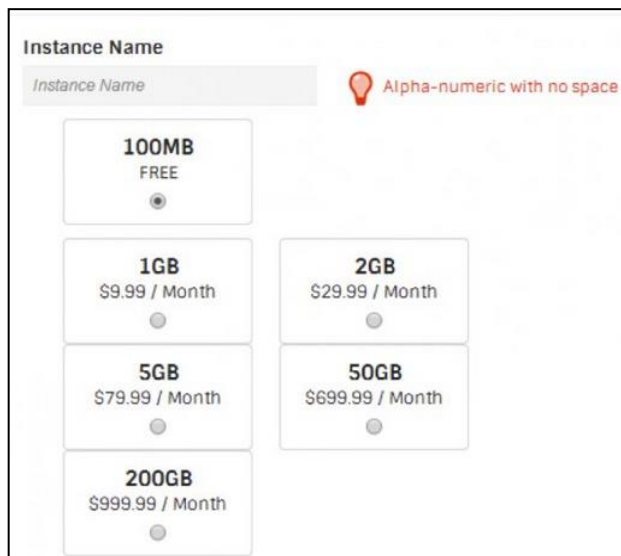


Fig. 17. Available instance sizes on Morpheus and their cost⁴¹

Morpheus uses Solid State Drives (SSD) for data storage, which improves the speed of data access. It also uses Amazon’s datacenters. Replication is done using a master-slave strategy to ensure availability and fault-tolerance. Scalability is achieved using autosharding.

Use cases show that Morpheus allows creating up to 2000 instances, with a total data size of 400 TB [66]. This, along with its scalability and high availability, makes Morpheus suitable for Big Data uses.

G. Postgres Plus Cloud Database

Postgres Plus Cloud Database (PPCD) [67] is a fully-managed, highly-performant, highly-available, scalable access to PostgreSQL, an object-relational database management system. It supports relational databases ACID transactions, as well as NoSQL databases features.

The architecture of PPCD is composed of one server, and clusters, as shown in Fig. 18.

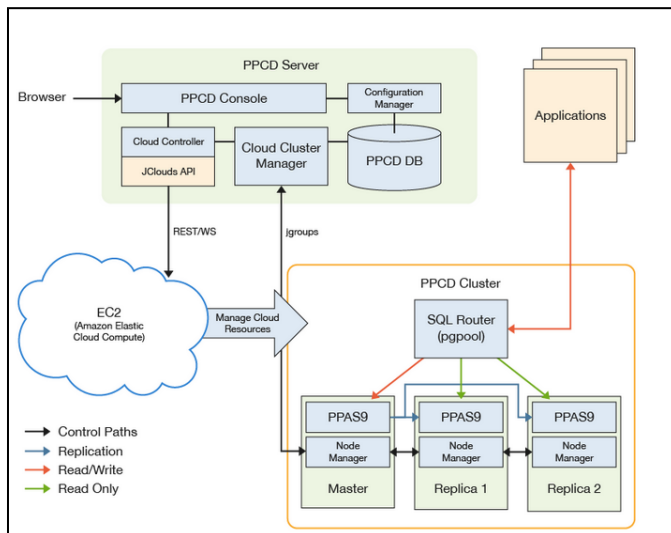


Fig. 18. The architecture of PPCD [67]

This architecture is for each cloud region. Users in a cloud region connect to a centralized console, the PPCD Console, to create clusters. The PPCD server deploys these clusters to the instances hosted by a Cloud provider (Amazon’s EC2 [67], Amazon’s VPC [68], etc) and connects to the cloud using JCloud APIs. The console uses jgroups, a toolkit for nodes messaging, to communicate with the various Cloud environments where clusters are deployed.

PPCD ensures reliability and availability using master-slave replication. The first database deployed by the console is designed as the master database, the other replicas are slaves and used for read-only operations. So PPCD clusters consist of a master and one or more replicas. They have built-in load balancers that receive incoming requests from applications and distribute them through the nodes.

The PPCD server manages the instances in the clusters using the Cloud Cluster Management (CCM). In case of failure, the CCM initiates automatic failover.

Automatic failover is implemented in two ways, as shown in Fig. 19. One way is to switch to a replica, which minimizes downtime, another is to migrate data from the failed master to a new one, which minimizes data loss.

PPCD offers, as a service, PostgreSQL databases that are hosted in the cloud, especially using Amazon’s WS. This lets PPCD benefit from Amazon’s powerful resources and makes it suitable for Big Data applications.

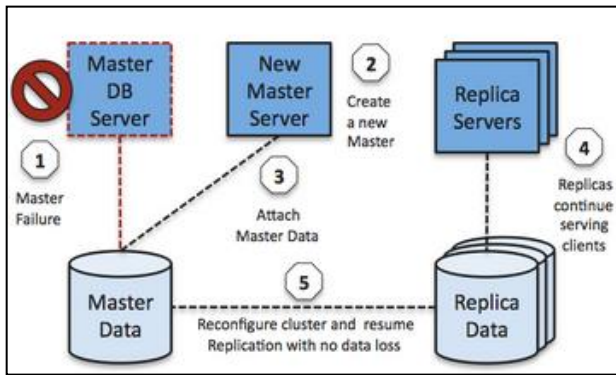


Fig. 19. Automatic failover scenario [67]

H. SimpleDB

SimpleDB is a highly available, scalable, schemaless non-relational document database that is part of Amazon's Web Services. It provides many of the functionalities provided by relational databases as a service in the cloud. SimpleDB is designed to run on other web services provided by Amazon. Developers that use SimpleDB can run their applications using Amazon's Elastic Compute Cloud (EC2) and store their data in Simple Storage Service (S3).

Data is structured in domains, which are the equivalent of tables in relational databases. Each domain is composed of attributes and items, and each attribute has one or more values for a given item, as shown in Fig. 20. Currently, users can store up to 10 GB of data per domain, and can create up to 250 domains [69]. However, they can request to create additional domains if needed.

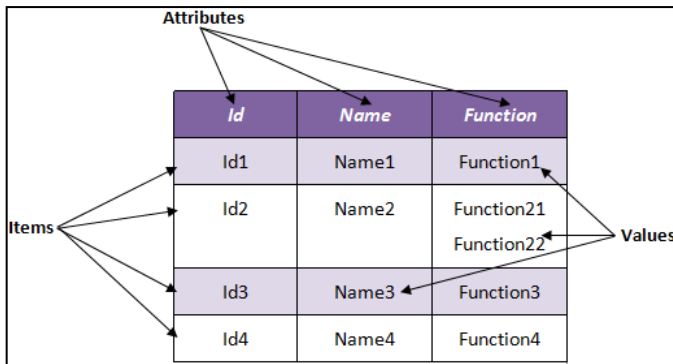


Fig. 20. Example of a domain in SimpleDB

SimpleDB provides a group of API calls to build applications [69], such as *CreateDomain* for creating domains, *DeleteDomain* for deleting domains, *PutAttributes* for adding, modifying, and removing data in domains, etc. Querying domains is done using an SQL-like *Select* query, but multi-domain querying is not supported.

SimpleDB implements automatic data indexing for a better performance. To ensure high-availability, asynchronous replication is implemented, and multiple copies of the domain are done after a successful write. Two consistency options are supported for read operations, namely strong consistency and eventual consistency. Strong consistency requires a majority of replicas to commit writes and acknowledge reads. Eventual

consistency asynchronously propagates writes through the nodes, and any replica can acknowledge reads. Automatic data sharding is not supported, so users have to manually partition their data across multiple domains for better scaling. SimpleDB is optimized for parallel-queries.

SimpleDB is designed for fast reading and is a simple way to store data in a schema-free database offered as a DBaaS. However, it has many drawbacks, such as the storage limit of 10 GB per domain, the maximum attribute values of 256 per item, the limit response size of 1 MB per query [70], the performance setback due to the automatic indexing of all attributes, etc. For all these reasons, Amazon built upon SimpleDB to develop DynamoDB, which can be considered an improved version of SimpleDB that is more adapted to Big Data applications.

I. DynamoDB

Amazon's DynamoDB is a fully-managed, highly-available, highly-scalable, distributed NoSQL database. It is an answer to Amazon's need of a performant, reliable, efficient database able to scale up to meet the ever growing load on their servers, which simultaneously serve, at peak times, more than tens of millions of customers [71], with all the economical issues at stake. DynamoDB is fast and flexible, and supports document and key-value data models.

Since strong consistency and high availability are complementary (according to the CAP theorem), and one must be sacrificed in order to achieve the other in distributed environments, Amazon chose to privilege high availability. Thus DynamoDB supports eventual consistency, which is achieved by asynchronously propagating updates, and considering each update to be a new version of data. This versioning is done by using vector clocks [72]. DynamoDB uses sloppy quorum, a quorum-based technique, and hinted handoff, a decentralized replica synchronization protocol, to achieve consistency among replicas while ensuring availability in case of server failures [71].

Conflicts during updates needed to be addressed too. The classical approach is to resolve these conflicts during writes, committing them only when the majority of replicas can be reached. To be more suitable for Amazon's services, where rejecting a write can be prejudicial from the customer's perspective, DynamoDB opts for resolving conflicts during reads. However, DynamoDB leaves it up to developers to implement their own conflict resolution strategy at the application level. By default, DynamoDB uses "the last write wins" strategy [71].

DynamoDB scalability is designed using a variant of consistent hashing in order to partition data and scale incrementally [71]. This variant dynamically partitions data over all the nodes in the clusters, knowing that each node communicates with its immediate neighbours. Some of these nodes are used as coordinators to replicate data on many nodes. DynamoDB optimizes throughput and latency at any scale by using automatic partitioning and Solid State Drive (SSD).

As for querying and manipulating stored data, it is done using two functions: *get(key)* to retrieve all the versions of the object associated with the key "key" along with their context,

and put(key, context, object) to determine where to store the replicas of the object “object” and to write them to the disk. Data is stored as binary objects, or blobs.

Problem	Technique	Advantage
Partitioning	Consistent Hashing	Incremental Scalability
High Availability for writes	Vector clocks with reconciliation during reads	Version size is decoupled from update rates.
Handling temporary failures	Sloppy Quorum and hinted handoff	Provides high availability and durability guarantee when some of the replicas are not available.
Recovering from permanent failures	Anti-entropy using Merkle trees	Synchronizes divergent replicas in the background.
Membership and failure detection	Gossip-based membership protocol and failure detection.	Preserves symmetry and avoids having a centralized registry for storing membership and node liveness information.

Fig. 21. A list of techniques used by DynamoDB as a response to some encountered problems and their advantages [71]

In DynamoDB, each node shares the routing table with the other nodes in the cluster in order to know what data is stored by which node. In the case of large clusters composed of thousands of nodes, the size of the routing table is significantly large. An improvement is suggested in [71] by using hierarchical extensions.

DynamoDB is Amazon’s NoSQL solution for Big Data storage. It has been used by Amazon’s services and given good performance, especially regarding availability and data loss. It is well-suited for many Big Data applications, from gaming to the Internet of Things.

J. Azure SQL Database

Azure SQL Database is a highly-available, scalable, relational database built on Microsoft SQL Server and hosted in Microsoft’s cloud. It offers the main features of traditional relational databases (tables, views, indexes, procedures, complex queries, full-text search, etc.) as a service in the cloud. It also supports Transact-SQL, ADO.net, and ODBC. Azure SQL Database supports Microsoft SQL Server only, though it is not completely compatible with it. However, a recent version offers a near total compatibility [73].

Azure SQL Database is a TDS [74] proxy endpoint that routes the requests of client applications to the SQL server node that contains the primary replica of data. It has a four-layer architecture, as shown in Fig. 22. First, the infrastructure layer, which is Microsoft Azure datacenter, provides powerful computing and storage resources on which the other layers are built. Then there’s the platform layer that contains at least three

nodes of SQL server running in the infrastructure layer. Then there’s the services layer that controls Azure SQL Database in terms of partitioning, billing, and connection routing. Last there’s the client layer that contains various tools to allow client applications to connect to Azure SQL Database.

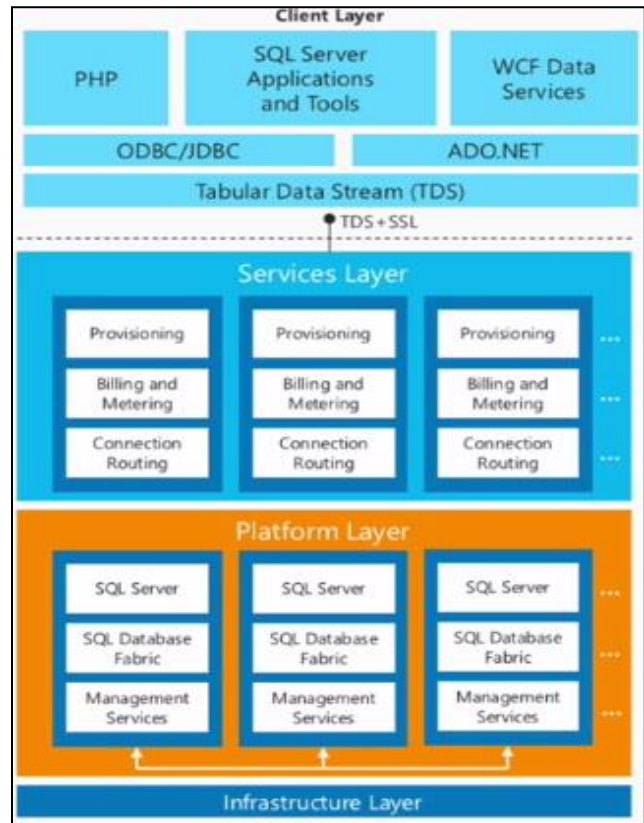


Fig. 22. Microsoft Azure SQL Database architecture [75]

Azure SQL Database organizes data in table groups, which are the equivalent of databases in SQL Server. A table group can be keyless or keyed. All tables in a keyed table group must have a common column called partitioning key. Rows that have the same partitioning key are grouped into row groups. However, Azure SQL Database doesn’t support executing transactions on more than one table group and, if the table group is keyed, on more than one row group.

Azure SQL Database performs automatic scalability when the table groups are keyed. Each table group is partitioned based on its partitioning key in a way that each row group is contained in one partition. To ensure availability, partitions are replicated using a Paxos-based algorithm, and each partition is stored on a server.

As for consistency, it is ensured by taking snapshots of the table group to verify that committed transactions are reflected in the table group, and uncommitted ones aren’t.

Azure SQL Database is used by many companies, including Xerox, Siemens, and Associated Press. However, it suffers from many limitations that render it unsuitable for Big Data applications. For example, the maximum database size supported is 500 GB, and the maximum database number

supported by a server is 150. So for Big Data applications, Microsoft's more adopted solution is DocumentDB.

DocumentDB is a fully-managed, scalable, NoSQL document database offered as a service. It supports SQL querying of JSON stored documents, which are all indexed by default to optimize query performance. Users can also query databases using JavaScript.

DocumentDB supports four levels of consistency, configurable by users. In addition to strong and eventual consistencies, there is session consistency, which is the default mode, and bounded staleness consistency. Session consistency asynchronously propagates writes, and sends read requests to the one replica that contains the requested version. Bounded staleness consistency asynchronously propagates writes, while reads are acknowledged by a majority of nodes, but may be lagged by a certain number of time or operations.

DocumentDB is still at its early stages and lacks many important features, such as backups and replication. Another solution developed by Microsoft and adapted to Big Data is SQL Server in Azure VM, which is not a DBaaS, but an IaaS to run SQL Server databases on virtual machines in the cloud.

K. Amazon RDS

Amazon Relational Database Service (RDS) offers a highly-available access to five distributed relational database management systems (MySQL, Oracle, Microsoft SQL Server, PostgreSQL, and Amazon Aurora) as a service in Amazon's Cloud. RDS aims to make setting up, running, and scaling relational databases simpler and easier, and to automate administrative tasks such as backups, point-in-time recoveries, and patching.

Scalability in RDS is achieved horizontally and vertically. RDS relies on sharding and read replicas to achieve horizontal scalability. As for vertical scalability, users can perform it by using command line tools, APIs, or AWS Management Console.

RDS supports automated backups. These backups can be used as point-in-time recoveries. In addition, users can program backups in the form of snapshots and that can be manually restored afterwards.

RDS replicates data synchronously using the Multi-AZ deployment [76] feature, where data is replicated between a primary instance and a standby instance, as shown in Fig. 23. Each one of these instances is stored in a different Availability Zone (AZ) to minimize downtime. If the primary instance fails, RDS performs an automatic failover to the standby instance.

RDS is most adapted to applications that already use one of the five supported database systems, or new applications that work with structured data and need relational features not supported by NoSQL databases, such as join operations [78]. It is also optimized for databases that support heavy I/O workloads. The size of databases stored in RDS can reach up to 3 TB and 30 000 IOPS [79], which makes it suitable for Big Data applications.

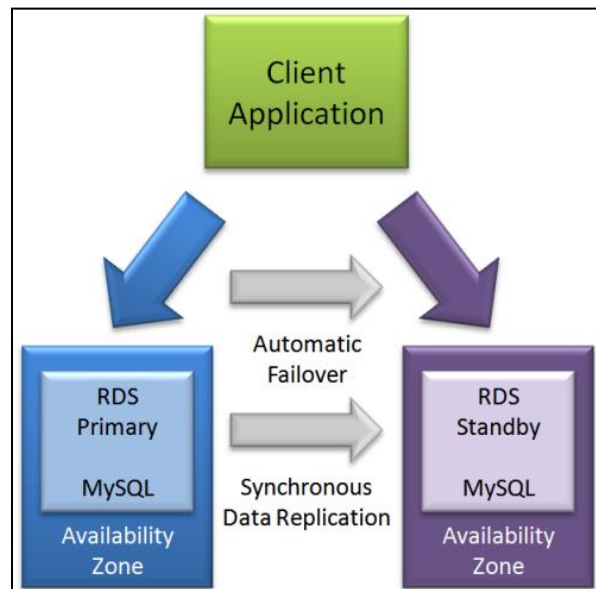


Fig. 23. Example of replication in RDS [77]

L. Other DBaaS solutions

There are various other DBaaS solutions, such as ClearDB, Clustrix, CumuLogic, Heroku, Percona, etc. They are meant for relatively small cloud deployment projects, not Big Data applications. Two prominent DBaaS solutions are HP Cloud Relational Database and Rackspace Cloud Database, two fully-managed, highly-available databases. Both support MySQL, with Rackspace Cloud Database supporting Percona Server, MariaDB also.

HP Cloud Relational Database is provided by HP and hosted in HP Helion Public Cloud. It is still in its early development stages, available in a beta version only for the users of HP Helion Public Cloud. Rackspace Cloud Database is provided by Rackspace. Both databases use OpenStack, an open source cloud computing platform. Users can manage their databases via the native OpenStack command-line interface tools, or APIs. HP Cloud Relational Database supports automated backup/restore operations to enhance fault-tolerance. Both databases offer the possibility for users to initiate backups. Availability is ensured by implementing snapshots and keeping replicas in different availability zones. Both databases are not suitable for Big Data applications, especially regarding data volume, HP Cloud Relational Database having a limiting size of 480 GB per database instance, and Rackspace Cloud SQL supporting a maximum size of 150 GB per database instance.

Rackspace acquired another DBaaS solution, Objectrocket, which is a fully-managed, highly scalable database that supports MongoDB and Redis. It offers the possibility of having instances of multiple TB. Another prominent DBaaS is Salesforce's Database.com, a fully-managed, highly-scalable relational database. It was first used as part of Salesforce's PaaS, force.com, before being available in a stand-alone version.

Database.com uses one large Oracle instance as the main data storage system. It arguably stores data in one wide table composed of hundreds of flex columns, which are columns storing various data types [80]. Salesforce doesn't disclose much of the technical details of Database.com's functionalities and architecture. For example, there are no resources detailing

how Database.com handles scalability, replication, or consistency. The maximum supported data size isn't specified either.

We present, in tables IV, V, and VI hereafter, a summary of the databases as a service reviewed in this section.

TABLE IV. COMPARISON BETWEEN THE REVIEWED DATABASES (PART 1)

Name	Provider	Data Model	Supported databases	Data Storage Type	Querying
Cloud Bigtable	Google	Column database	N/A	Tables and Tablets	HBase Shell
Azure SQL Database	Microsoft	Relational	Microsoft SQL Server	Tables	SQL
Cloud Datastore	Google	NoSQL	N/A	Kinds (equivalent of relational tables)	API
HP Cloud Relational Database	HP	Relational	MySQL	Tables	OpenStack CLI API
Cloud SQL	Google	Relational	MySQL	Tables	SQL
Cloudant	IBM	Document datastore	N/A	JSON documents	API
DynamoDB	Amazon	Key-value store	N/A	Key-Value objects	API
MongoLab	MongoDB	Document datastore	MongoDB	BSON documents	MongoDB driver API
Morpheus	Morpheus	Relational or NoSQL	MySQL MongoDB Redis Elasticsearch	Tables JSON documents Key-Value objects	API
Postgres Plus Cloud Database	EnterpriseDB	Relational	PostgreSQL	Tables	API
Rackspace Cloud Database	Rackspace	Relational	MySQL Percona Server MariaDB	Tables	CLI API
RDS	Amazon	Relational	MySQL Oracle Microsoft SQL Server PostgreSQL Amazon Aurora	Tables	SQL
SimpleDB	Amazon	Key-value store	N/A	Key-Value objects	API
Database.com	Salesforce	Relational	N/A	Organizations (equivalent of relational tables)	SOQL SOSL
Objectrocket	Rackspace	Document and Key-value store	MongoDB Redis	Key-Value objects BSON	API

N/A: NOT APPLICABLE X: NOT AVAILABLE

TABLE V. COMPARISON BETWEEN THE REVIEWED DATABASES (PART 2)

Name	Consistency	Replication	Scalability	Deployment Model	Clusters Tenancy	Interoperability with other cloud platforms	Geographical region choice
Cloud Bigtable	Strong Row Consistency	X	Horizontal	Cloud	Multi-tenancy	No	Yes
Azure SQL Database	Strong	Asynchronous	Horizontal	Cloud	Multi-tenancy	No	Yes
Cloud Datastore	Strong Eventual	Asynchronous	Vertical Horizontal	Cloud	Multi-tenancy	No	Yes
HP Cloud Relational Database	X	X	Vertical Horizontal	Cloud	Multi-tenancy	No	Yes
Cloud SQL	X	Synchronous Asynchronous	Vertical	Cloud	Multi-tenancy	No	Yes
Cloudant	Eventual	Synchronous	Horizontal	Cloud On-premise Hybrid	Single-tenancy Multi-tenancy	Rackspace Microsoft Azure SoftLayer	Depending on the provider
DynamoDB	Eventual	Asynchronous	Vertical Horizontal	Cloud	Single-tenancy Multi-tenancy	No	Yes
MongoLab	Eventual	Asynchronous	Horizontal	Cloud	Single-tenancy Multi-tenancy	Amazon Web Services Google Cloud Platform Windows Azure Rackspace Joyent	Depending on the provider

Name	Consistency	Replication	Scalability	Deployment Model	Clusters Tenancy	Interoperability with other cloud platforms	Geographical region choice
Morpheus	X	Asynchronous	Horizontal	Cloud On-premise Hybrid	Single-tenancy Multi-tenancy	Yes (through Morpheus Virtual Appliance)	No
Postgres Plus Cloud Database	Eventual	Asynchronous	Vertical Horizontal	Cloud On-premise	Single-tenancy Multi-tenancy	Amazon Web Services HP Cloud	Yes
Rackspace Cloud Database	Eventual	Asynchronous	Vertical	Cloud On-premise	Single-tenancy Multi-tenancy	No	X
RDS	Strong	Synchronous Asynchronous	Vertical Horizontal	Cloud	Single-tenancy Multi-tenancy	No	Yes
SimpleDB	Strong Eventual	Asynchronous	Vertical Horizontal	Cloud	Single-tenancy Multi-tenancy	No	Yes
Database.com	X	X	X	Cloud	Multi-tenancy	No	X
Objectrocket	Eventual	Asynchronous	Horizontal	Cloud On-premise	Single-tenancy Multi-tenancy	No	Yes

N/A: NOT APPLICABLE X: NOT AVAILABLE

TABLE VI. COMPARISON BETWEEN THE REVIEWED DATABASES (PART 3)

Name	Client libraries	License	billing	SLA commitment	Initial release date	Big Data compatible	Used by
Cloud Bigtable	HBase Client	Commercial	Per hour	Yes	2015	Yes	Google
Azure SQL Database	PHP ODBC/JDBC .NET SQL Server Tools WCF Data Services TDS	Commercial	Per hour	Yes	2009	Yes	Samsung easyJet
Cloud Datastore	Java JavaScript PHP Python .NET Objective-C	Commercial	Per hour	Yes	2013	Yes	Ubisoft GenieConnect
HP Cloud Relational Database	RESTful API	Commercial	Per hour	No	2012 (beta version)	No	X
Cloud SQL	Java JavaScript PHP Python .NET Objective-C	Commercial	Package Per hour	Yes	2013	No	BeDataDriven CodeFutures
Cloudant	Java Node.js	Commercial	Per month	Yes	2008	Yes	Adobe DHL
DynamoDB	Java PHP .NET Ruby	Commercial	Per month	Yes	2012	Yes	Elsevier Amazon Cloud Drive
MongoLab	Java PHP Python Ruby Node.js	Commercial	Per month	Yes	2011	Yes	Toyota Lyft
Morpheus	Not Available	Commercial	On estimate	No	2014	Yes	Spireon Rowmark
Postgres Plus Cloud Database	Java .Net	Commercial	Per hour Per month One Year subscription	Yes	2012	Yes	Bouygues Telecom Los Angeles Times
Rackspace Cloud Database	Java PHP .Net RESTful API	Commercial	Per hour	Yes	2012	No	InferMed
RDS	Java PHP .Net	Commercial	Per month	Yes	2009	Yes	Airbnb Unilever

Name	Client libraries	License	billing	SLA commitment	Initial release date	Big Data compatible	Used by
SimpleDB	Java PHP Python Ruby .NET	Commercial	Per hour	Yes	2007	Yes	MyMiniLife.com Issuu
Database.com	RESTful API SOAP API	Commercial	On estimate	Yes	2010	X	Cirrus Computing
Objectrocket	Java PHP Python Node.js cURL	Commercial	On estimate	Yes	2012	Yes	SponsorHub SumAll

N/A: NOT APPLICABLE

X: NOT AVAILABLE

V. DISCUSSION

As presented in the previous section, there are various databases offered as a service by many Cloud providers. This model of use, namely DBaaS, offers many advantages both to users and providers. Users find themselves exempt from up-front investments and relieved from the burden of installing, running and administrating their databases. As for providers, the costs of providing their service are optimized, especially in the case of multi-tenancy.

However, there are several points to take into consideration when selecting a DBaaS, few of which we discuss hereafter.

A. Provider's reputation

Within the last decade, Cloud Computing has positioned itself as a primordial technology with an ever growing market, although big IT names still have a dominating position. In the first quarter of 2015 [81], Amazon held 29% of the market share, followed by Microsoft (10%), IBM (7%), Google (5%), Salesforce (4%), and Rackspace (3%). Every one of these providers has a DBaaS solution that benefit from their established Cloud platforms, whether relational (Amazon's RDS and SimpleDB, Microsoft's Azure SQL Database, Google's Cloud SQL, and Rackspace's Cloud SQL) or NoSQL (Amazon's DynamoDB and SimpleDB, IBM's Cloudant, and Google's Cloud Datastore).

In addition to these providers, other ones have positioned themselves quite successfully in the DBaaS market, such as Mongo inc. (MongoLab), Morpheus, and EnterpriseDB (Postgres Plus Cloud Database).

Users may be more confident confiding their data to well-established Cloud "pioneers", or choose to rely on other users' feedback, which every provider has on their website in the form of use cases.

B. Deployment

Users who are looking for a DBaaS should consider the deployment model to know whether their data will be stored on-premise or off-premise. For example, some users would choose to keep their data on-premise, for security concerns. Many providers don't offer the choice, as their databases are hosted in the Cloud only. This is the case for Amazon, Google, Microsoft, MongoDB, and Salesforce. Other providers give the possibility to choose between using their database as a hosted service in their Cloud or on-premise. This is the case of EnterpriseDB, HP, IBM, Morpheus, and Rackspace.

Another point regarding deployment is the interoperability of the DBaaS with other Cloud providers' solutions. In many cases, users' applications are already deployed, whether internally or in the Cloud. Thus, it would be more convenient when a DBaaS provider enables users to select the cloud platform they want to use, even if it is provided by another Cloud provider. This is not the case for providers like Amazon, Google, Microsoft, HP, Rackspace, and Salesforce, who compel customers to use their specific Cloud platforms, as their databases can't be used elsewhere.

Tenancy mode is also a point to consider when selecting a DBaaS. Customers desiring to optimize their database performance may want to opt for single-tenancy, where they get dedicated clusters and don't share resources with other customers. Not all DBaaS have this option. Database.com, for example, was specifically designed to be multitenant. Providers like Microsoft, Google, and HP don't offer this possibility either.

C. Database model

Providers who support many database systems give users the possibility to select a database to use from available databases. This way, customers can choose the database to which they are used or that they are most comfortable with. This can be particularly interesting for users who already have their applications deployed and running, because when a DBaaS offers access to a traditional database (MySQL or PostgreSQL for example), the codes that were designed to work with these databases can work seamlessly in the cloud, exempting users from rewriting their code.

Another point to study before choosing a DBaaS is the data model. Customers must have a clear idea of how they project to use their database, and especially the type of data they deal with. Although developers may benefit from the flexibility of NoSQL databases, due to their being schema free, they will have to explicitly manage data coherence in the application layer (relationships between data, for example, as there are no defined foreign keys in the database). Thus, if data is variably structured and can't be represented using the relational schema, then NoSQL databases will be more adapted. If not, then some relational DBaaS can offer good performance for Big Data applications, like Amazon RDS or Microsoft Azure SQL database.

D. Law and regulations

Data collection and storage are increasingly subject to regulations, whether directly, such as the "Data Protection

Directive” (DPD) [82] in the European Union, or indirectly, such as the “USA Patriot Act” in the United States of America. Such legislation affects the storage of data. The DPD, for example, requires personal data to be stored inside the EU, or only in countries outside the EU that ensure a certain level of data protection.

DBaaS physically store data in various datacenters in different locations. Moreover, to ensure availability, data is replicated across geographically distributed datacenters. Users in some cases may need to choose the geographical location where their data will be stored. This possibility is offered by the majority of the reviewed providers (except Morpheus), who have datacenters mainly in the USA and the EU. Other providers, like Salesforce and Rackspace, don’t give details about the location of their datacenters. Another possibility is to opt for keeping data on-premise, which is possible for DBaaS like Cloudant, Postgres Plus Cloud Database, Rackspace Cloud Database, and Objectrocket.

E. Payment mode

One of the main characteristics of Cloud Computing is the concept of pay-as-you-go, where users strictly pay for the resources they consume. DBaaS users pay for the volume of data they store, according to several purchasing options. The majority of providers adopt a billing by the hour plan, where users pay for the volume of data stored during one hour. Examples include Google and Microsoft. Amazon, IBM, and MongoDB enlarge the time period to a month, while other providers like Morpheus, Salesforce, and Rackspace tailor their payment to customers, on a case-by-case basis.

F. Data volume

Choosing a DBaaS for Big Data applications implies to carefully consider the maximum supported size in order to ensure that it can scale to handle terabytes of data. While most reviewed DBaaS verify this condition, HP Cloud Relational Database, Cloud SQL, and Rackspace Cloud Database only offer a maximum instance size of 500 GB. Salesforce doesn’t disclose information about Database.com maximum storage size.

G. Data consistency

Consistency, availability, and partition tolerance being complementary (as stated by the CAP theorem), most reviewed DBaaS chose to relax consistency in order to achieve high-availability in distributed environments. This is the case for Cloudant, DynamoDB, MongoLab, Postgres Plus Cloud Database, Rackspace Cloud Database, and Objectrocket. For applications that can’t relax consistency, strong consistency is offered by DBaaS like Azure SQL Database, SimpleDB, and Cloud Datastore. The two latter ones implement both strong and eventual consistency, allowing users to choose the most adapted mode.

H. Scalability

Scalability allows adjusting computing resources and storage space to meet the increasing needs of applications. It is one of the inherent characteristics of cloud computing and one of the necessary requirements for Big Data applications.

Most reviewed databases scale horizontally to meet the levels required by Big Data applications. Databases like Cloud Datastore, DynamoDB, Postgres Plus Cloud Database, Amazon RDS and SimpleDB implement both vertical and horizontal scalability. Cloud SQL and Rackspace Cloud Database scale only vertically, which, added to their size limitations, makes them further unsuitable for Big Data applications. As for Salesforce’s Database.com, there is no information on how it handles scalability.

I. SLA

A Service-Level Agreement (SLA) is a contractual document that governs the client’s use of the provider’s services.

SLAs help providers manage the services contracted and maintain the overall level of quality agreed on with their customers. The providers of the reviewed databases use SLAs, except for HP and Morpheus, who don’t disclose their SLA policy. They all guarantee high availability, with an uptime of 99.9% at least.

J. Security and Privacy

One of the main concerns that keep organizations and individuals from moving their data to the cloud is the security and privacy aspects. Recent leaks and hacks (iCloud and Sony, to name but a few) only reinforced their reluctance to entrust data to the Cloud [83, 84].

The concern of security and privacy in cloud environments is enhanced by the large volume of datasets managed by Big Data. And just like DBaaS removes the burden of database installation and management, it also ensures the security of data. DBaaS providers implement different levels of security, starting from identity and access management, to data encryption, all through assuring the physical security and monitoring of datacenters. In addition to securing data while being stored in datacenters, it is crucial to ensure its transfer to and from client applications, which can be implemented using cryptographic protocols like TLS or SSL.

Providers like Amazon, Google, Microsoft, IBM, and Rackspace have achieved the ISO/IEC 27001 certification for their cloud platforms.

VI. CONCLUSION

Big Data has emerged as one of the most important technological trends for the current decade. It challenges the traditional approach to computing, especially regarding data storage. Traditional clustered relational database environments prove to be complex to scale and distribute to adapt to Big Data applications and new solutions are continually being developed.

One of the most adapted answers to Big Data storage requirements is Cloud Computing, and more specifically Database as a Service, which allows storing and managing tremendous volume of variable data seamlessly, without need to make large investments in infrastructure, platform, software, and human resources. In this context, our article presents a benchmark of the main database solutions that are offered by providers as Database as a Service (DBaaS). We studied the

features of each solution and its adaptability to Big Data applications.

Cloud Computing and Big Data are entwined, with Big Data relying on Cloud Computing's computational and storage resources, and Cloud Computing pushing the limits of these resources. New extensions of Cloud Computing are emerging to further enhance Big Data, especially Fog Computing and Bare-Metal Cloud. Fog Computing uses edge devices and end devices, such as routers, switches, and access points to host services, which minimizes latency. This proximity to end-users, along with its wide geographical distribution and support for mobility makes Fog Computing ideal for Big Data and the Internet of Things applications [85]. As for Bare-Metal Cloud, it aims to optimize performance for applications with high workloads by eliminating the virtualization layer and delivering "bare" servers without hypervisors installed. This way, there won't be too many virtual machines competing for physical resources and impeding the overall performance.

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Analysis of the SNR Estimator for Speech Enhancement Using a Cascaded Linear Model

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Abstract—Elimination of tainted noise and improving the overall quality of a speech signal is speech enhancement. To gain the advantage of individual algorithms we propose a new linear model and that is in the form of cascade adaptive filters for suppression of non-stationary noise. We have successfully deployed NLMS (Normalized Least Mean Square) algorithm, Sign LMS (Least Mean Square) and RLS (Recursive Least Square) as the main de-noising algorithms. Moreover, we are successful in demonstrating that the prior information about the noise is not required otherwise it would have been difficult to estimate for fast-varying noise in non-stationary environment. This approach estimates clean speech by recognizing the long segments of the clean speech as one whole unit. During experiment/implementation we used in-house database (includes various types of non stationary noise) for speech enhancement and proposed model results have shown improvement over conventional algorithms not only in objective but in subjective evaluations as well. Simulations present good results with a new linear model that are compared with individual algorithm results.

Keywords—Least Mean Square (LMS); Normalized Least Mean Square (NLMS); Recursive Least Square (RLS); Speech Enhancement; Non-stationary

I. INTRODUCTION

The goal of speech enhancement is to improve the quality and intelligibility of speech that has been degraded by noise [11]. The speech varies according to the needs of specific applications, such as to increase the overall speech quality or intelligibility. In real-life contexts, there are wide variety of situations in which we need to enhance speech signals. Speech enhancement techniques have been successfully applied to problems as diverse as correction of disrupted speech due to pathological problems of the speaker, pitch and rate modification, restoration of hyperbaric speech, and correction of reverberation, but, noise reduction is probably the most important and most frequently studied Issues [19]. Prior to designing algorithms that cope with adverse conditions, it is crucial to understand the noise characteristics and the differences between the noise sources in terms of temporal and spectral characteristics. Noise can be impulsive, continuous, and periodic and its amplitude may vary in frequency range [19]. In previous literature various speech enhancement techniques are given for noise reduction [13]. Some are given as spectral subtraction, modified spectral subtraction, wiener and gain based method like MMSE STSA, log-MMSE, P-MMSE [15] etc. All methods give improvements in speech

quality but do not improve intelligibility up to satisfactory levels [20]. In the real environment speech may be distorted by more than one noise source. Most of the time it is not possible to consider only single noise in communicated speech or number of noises in speech signal [20]. Most noise estimation algorithms work well for stationary or slowly varying noise, but very few are working for non stationary noise. This is because of the weak predictability of fast-varying noise. Because of the non stationary nature of the speech signal, most current enhancement algorithms operate on a frame-by-frame basis [16]. Many algorithms ignore the temporal constraints between adjacent speech frames. Without context, and without specific knowledge about the noise, it is difficult to separate the speech from noise in the duration of a frame (typically about 20 ms). This is especially true when the noise is a form of speech (e.g., a crosstalk sentence). Previous research has revealed the importance of imposing cross-time spectral constraints in improving speech enhancement quality [16]. In many such cases only a single-channel speech signal is available. Of the available solutions to the single-channel speech enhancement problem, Short Time Fourier Transforms (STFT) based methods achieve relatively good performance and compromise the majority [15]. It is appropriate to further categorize this class of speech enhancement algorithms into the sub-categories of spectral subtraction, wiener filter and statistical approaches [15]. Wiener filter is used for linearity whereas spectral subtraction is used for simplified mathematical expressions [15]. Almost all the papers work on speech enhancement with added known amount of noise and then use their proposed algorithm to enhance the speech or reduce the noise level. In this case, mostly noise is assumed to be white, Gaussian noise and colored noise [11]. However, if one records speech on the road or in the market, there is no guarantee that the noise is Gaussian. For this enhancement algorithm to be really useful, it must improve the quality of speech that was originally noisy due to some environmental conditions like railway station, fan, vehicle, machine gun, tank, factory etc that create distortions in clean speech signal and not due to explicit addition of noise by the researcher [13]. However, we have to bear this constraint in mind that the enhancement must be prominent in both quantitative as well as qualitative manner and at the same time, we should not be overlooking the complexity and the ease with which these algorithms can be implemented on hardware platforms [10]. In this paper, we aim to reduce the requirement of priori information about the noise as this can be difficult to estimate with fast varying noise [16]. We propose a class of two stage

adaptive architecture to address some properties of non-stationary noise by calculating energy for original speech and then calculate energy for processed speech with SNR, MSE. In the speech enhancement process, the estimation of the a priori signal-to-noise ratio (SNR), Mean square error (MSE), energy, Power Spectral Density (PSD) is one of the most important parts, especially in non-stationary environments [5].

This paper is divided into 6 sections. Section 1 gives the overview of speech enhancement. Section 2 is all about the basic principle of suppression of non stationary noise. Section 3 contains details about the Least Mean Square Algorithm, Normalized Least Mean Square Algorithm and Recursive Least Square Algorithm. The methodology, set up for problem solution presented in section 4. All the details about the data set used for analysis and results presented in section 5. At the end discussion and conclusion are included in section 6, respectively. In this paper, we study the problem of retrieving speech from non stationary noise assuming minimal noise prior.

II. SUPPRESSING NON STATIONARY NOISE

Noise can generally be classified into three major categories based on its characteristics: Stationary noise, Pseudo or Non-stationary noise and transient noise [19]. The spectral and temporal characteristics of pseudo or non-stationary noise change constantly. The task of suppressing this type of noise is more difficult than that of suppressing stationary noise [19]. Another distinctive feature of noise is their spectrum shape, particularly the distribution of noise energy in the frequency domain. For instance, most of the energy of car noise is concentrated in the low frequencies, i.e., it is low-pass in nature. In most speech enhancement methods, the estimation of power of the noise is a requirement. Fortunately, the bursty nature of speech makes it possible to estimate the noise during speech pauses [14]. Moreover, it should be mentioned that it is easier to deal with additive noise than convolutive noise [14]. For practical reasons, the estimation of the noise is almost performed in the spectral domain. Actually, spectral components of speech and noise are partially uncorrelated. Besides this, perception/hearing and psycho-acoustic models are well understood (and adapted) in the spectral domain [19]. Realizing the limitations of traditional speech enhancement methods in the presence of non stationary noise, Research efforts have been directed over the past decades to devise the new solutions. The solutions find into two categories: improvements to the noise estimators and modifications of the suppression rule [14]. The former class of methods essentially targets the limitations of Voice activation detection (VAD) based noise estimation. VAD is based on some prior knowledge of the speech signal. As discussed earlier, accurate estimation of noise spectrum would make effective with single channel speech enhancement methods in any background noise conditions. More research has been focused on improving the noise estimation. Due to the limitations of VAD, a number of methods are available for noise spectrum estimation. These methods are based on tracking some statics of power spectral values for each frequency bin over several frames. The test file has a continuous stretch of speech frames by high and low pulses in the plot. Noise is non stationary with random bursts has been detected by high and low pulses [14]. A different

approach to carry out the adaptation of noise during both speech presence and absence is by estimate of SNR. Any sudden increase in the background noise level is not easily distinguished from speech and results in high estimated SNR making the method less effective in highly non stationary noise.

III. PROPOSED METHOD

As mentioned earlier, we focus on a common form of non stationary noise characterized by randomly occurring noise bursts in a stationary background. These noise bursts lead to the partial or complete corruption of the spectrum of the speech. A traditional STSA algorithms based on a simple noise estimator can effectively suppress only the stationary background noise leaving mainly the speech together with residual noise bursts in the enhanced signal. The proposed processing algorithm involves identifying regions in the spectrogram of the enhanced speech that are dominated by the residual noise. These cascaded adaptive algorithms contribute to improve the overall quality of the enhanced speech

A. Least Mean Square Algorithms

An adaptive filter is a filter that self-adjusts its transfer function according to an optimizing algorithm. Because of the complexity, most adaptive filters are digital filters that perform digital signal processing and adapt their performance based on the input signal [9]. Least mean squares (LMS) algorithms is such an adaptive filter used to find the filter coefficients by equation (1) that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). In stochastic gradient descent method, the adaptive filter is only adapted based on the current error [9].

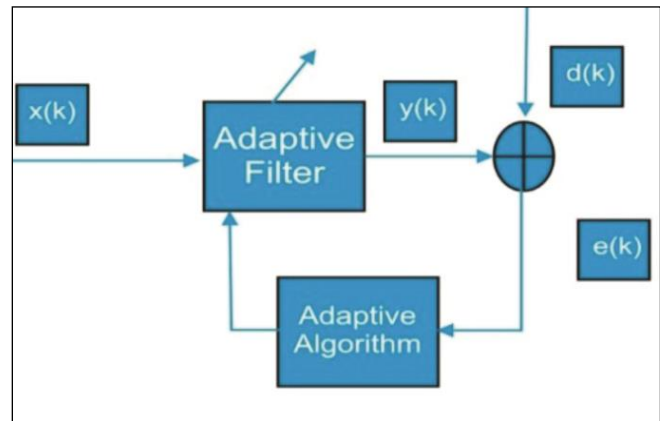


Fig. 1. Adaptive filter

$$W(k+1) = W(k) + \mu e(k) \quad (1)$$

$$e(k) = d(k) - W^H(k)X(k) \quad (2)$$

Equation (1) is used for calculating the updated weights and equation (2) is used for calculating error signal, where $X(k)$ represents the input signal vector. The least mean square (LMS) algorithm uses the statistical properties of the signals. The main objective of this method is to minimize the mean square error. The LMS algorithm is widely used in the adaptive algorithm because of its simplicity in structure and its robustness for numerical analysis.

B. Normalized Least Mean Square (NLMS) Algorithm

NLMS is widely used algorithm because of its simplicity and robust performance. The stability of the basic NLMS is controlled by a step size. This parameter also governs the rate of convergence, speed, tracking ability and the amount of steady-state excess mean-square error (MSE) [9] aimed at solving conflicting objectives of fast convergence and low excess MSE. It achieves a certain degree of success that converges slowly with colored input signals. In the standard LMS algorithm if $x(n)$ is large, it suffers from gradient noise amplification. But normalized LMS algorithm seeks to avoid gradient noise amplification. The step size is time varying and optimized to minimize error [9].

$$w(n+1) = w(n) + \mu(n) \nabla J(n) \quad (3)$$

$$= w(n) + \mu(n) [p - R w(n)] \quad (4)$$

C. Recursive Least Square (RLS) Algorithm

Recursive algorithm is used for the design of adaptive transversal filters which provides the least square estimate of the tap weight vector of the filter at iteration (n-1) and also computes the updated estimate of the vector at the iteration n upon the arrival of new data [9]. RLS algorithms calculate $J(n)$ by using the following equation:

$$J(n) = 1/N \sum \lambda^i e^2(n-i) \quad (4)$$

In this Equation where N is the filter length and λ is the forgetting factor. This algorithm calculates not only the instantaneous value $e^2(n)$ but also the past values, such as $e^2(n-1), e^2(n-2), \dots, e^2(n-N+1)$. The range of the forgetting factor is (0, 1). When the forgetting factor is less than 1, it specifies that this algorithm places a larger weight on the current value and a smaller weight on the past values [9].

The resulting $E[e^2(n)]$ of the RLS algorithms is more accurate than that of the LMS algorithms [10]. The LMS algorithm require fewer computational requirements and memory as compare to RLS algorithm.

However, the Eigen value of the input correlation matrix, might affect the convergence speed of the resulting adaptive filters [9]. The convergence speed of the RLS algorithms is much faster than that of the LMS algorithms. However, the RLS algorithms require more computational complexity than the LMS algorithms.

$$\begin{aligned} E(i) &= d(i) - y(i) \\ &= d(i) - w^H(n)u(i) \end{aligned} \quad (5)$$

$$w(n) = w(n+1) + k(n)\xi^*(n)$$

IV. CASCADE

The LMS algorithm provides only enhancement, NLMS has got the problem of musical tones, and in case of RLS algorithm rate of convergence is typically an order of magnitude faster than that of simple LMS filter. So RLS filter whitens the input data by using inverse correlation matrix of the data with zero mean, but this improvement in performance

increase the computational complexity of the RLS filter [10]. After evaluating them, we come to a conclusion that NLMS-RLS is the optimum cascade for speech enhancement.



Fig. 2. Block Diagram

It has two stages: Normalized Least Mean Square algorithm and Recursive Least Square algorithm and the main idea of this approach are that of enhancing the original speech that is the combination of speech with non stationary noise.

V. EXPERIMENTAL RESULTS

In this section, we present the experimental results of the proposed cascaded system that may use Output energy, SNR and MSE for the comparison. A comparison with the single algorithms will be used to study the merits and demerits. Due to limitations, we will replace single algorithm by cascaded algorithms.

A. Setup

For performance evaluation of the proposed method, English Language speech patterns have been recorded in different situations. The English speech patterns have been added with noise patterns. The listeners participated in different conditions. For each condition speech processed with circuit of adaptive filter algorithm. Ten sentences from the IIT-H database are taken for clean speech that is produced by male and female speakers. Railway station and Restaurant noise were added to the clean speech files. Each male and female provided 2 to 6 minutes of "conversational speech" that is a story of anything. All recordings were originally digitized at a sampling rate of 16kHz. Then down sampled according to system requirement. Each story was cut to have different lengths of 6 sec and 3 sec.

We first show the performance of the traditional single channel filter which provides a benchmark for studying other noise reduction filters. Using a large $\zeta = 1$ there is large variation in the value of SNR. SNR is increasing with the large value of leakage factor. But in case of RLS if we decrease forgetting factor delay is more .so best performance is achieved at moderate value of ζ . The output SNR reaches its peak when $\zeta = 1.0$. The noise corrupted speech was processed by different circuits of adaptive filter algorithms that included cascaded version of NLMS algorithm and RLS algorithm based on a priori SNR and energy estimation. Different conditions are 1.Clean speech 2.speech with noise 3.Only noise. These listening experiments were conducted using headphones at a comfortable listening level. Each subject listened to a set of noise corrupted sentences to be formalized with the testing procedures.

The performance of the proposed method is compared in terms of parameters like Input energy, output energy, MSE, SNR. The values of these parameters are given in table for comparative analysis. In table 1 we calculated SNR with $K=32, 64, 128$ & 512 with new linear model and table 2 we calculated SNR with NLMS algorithm. It shows that SNR is improved in new linear model as compare to NLMS Circuit. In the fig

spectrograms of speech signal voice 001.wav for 6 sec is shown and then that signal is processed by NLMS, RLS and combination of NLMS-RLS by setting the buffer values according to number of samples and each algorithm parameters are also adjusted according to system requirement.

The babble noise was recorded in the factory and cut that noise for 6 sec and 3 sec that are shown in fig. and that the

signal was processed by NLMS, RLS and combination of both the algorithms. Such a NLMS filter with $\mu = 0.00095$, $N=512$, $w=0.1$ and $\zeta= 0.85$. In this experiment we let the Output SNR is increasing and MSE is also increasing. But if we reduce the size of the input signal MSE should be same but SNR is still increasing. Consider leakage factor ($\zeta= 0.72, 0.85$ and 1.0) we examine that performance varies with variation in the

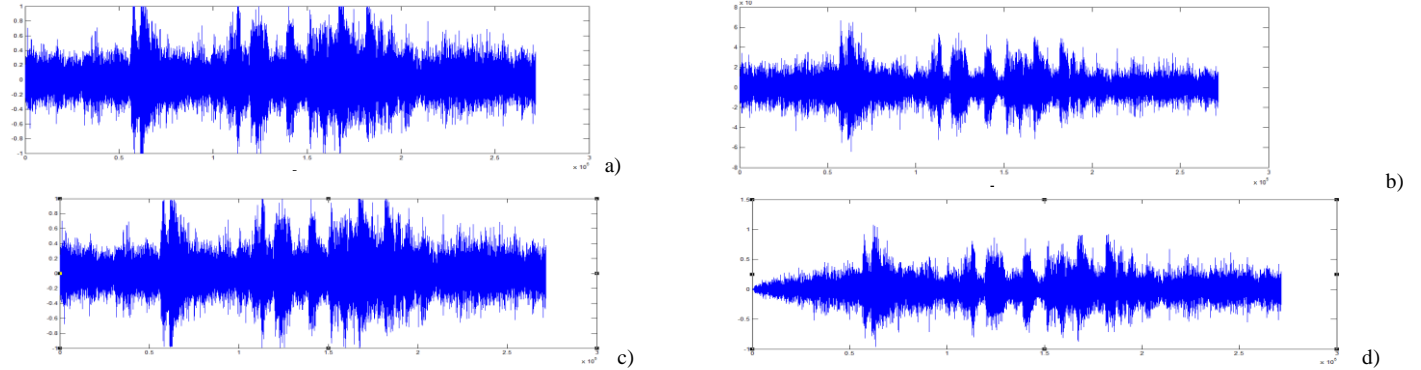


Fig. 3. (a) Speech corrupted by factory noise for 6 sec (b) Speech enhanced by NLMS (c) Speech enhanced by RLS (d) Speech enhanced by NLMS-RLS

B. Results

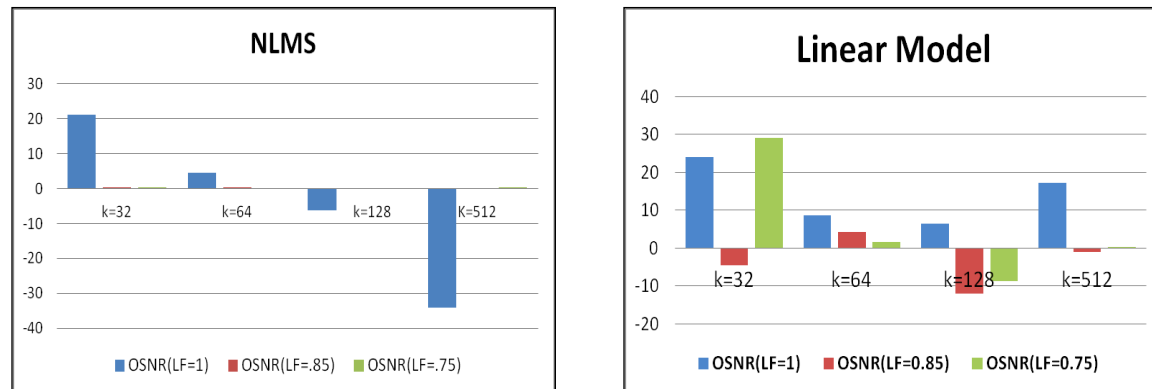


Fig. 4. Full band performance by forgetting factor 0.75, 0.85, 1 with window length 32, 64, 128, 512 by using a linear model and NLMS

TABLE I. ANALYSIS OF OSNR WITH DIFFERENT WINDOW SIZES WITH NEW LINEAR MODEL

Window size k	OSNR(LF=1)	OSNR(LF=0.85)	OSNR(LF=0.75)
k=32	24.13	-4.561	29.14
k=64	8.635	4.228	1.656
k=128	6.378	-12.09	-8.778
k=512	17.28	-0.9701	0.3721

TABLE II. ANALYSIS OF OSNR WITH DIFFERENT WINDOW SIZES WITH NLMS ALGORITHM

Window length k	OSNR(LF=1)	OSNR(LF=.85)	OSNR(LF=.75)
k=32	21.08	0.1903	0.03602
k=64	4.572	0.09006	-0.005574
k=128	-6.347	-0.2226	-0.08932
K=512	-34.14	-0.0125	0.0009499

TABLE III. COMPARISON TEST RESULTS FOR SIGNAL (6 SEC)

Parameters	NLMS	RLS	NLMS+RLS
In Energy	13970	560.7	13970
MSE	-0.04871	-0.004243	8749
SNR	-0.0125	20.47	17.28

TABLE IV. COMPARISON TEST RESULTS FOR SIGNAL (3 SEC)

Parameters	NLMS	RLS	NLMS+RLS
In Energy	5442	5442	5442
MSE	-0.01898	-0.04129	-0.01898
SNR	-0.00562	0.1718	0.2286

TABLE V. POST PROCESSING ALGORITHM VALUES FOR DIFFERENT TYPES OF SIGNALS

Parameters	Duration	No of samples	Energy	Power
Voice 001.wav	6 sec	286720	0.316694439	80.86 dB
Voice 001(1)	3 sec	132300	0.123402411	80.12 dB
Voice 006.wav	3 Sec	132300	0.238506168	82.98 dB
Voice 0012	3 Sec	132300	0.123402411	80.12 dB

The performance of the processing algorithm is evaluated for the real environmental noises like factory noise, canteen, digging noise. All these tables show that energy and SNR of the cascaded signal is higher than others individual Adaptive filters. All these noises are highly fluctuating and characterized by random energy bursts. Noise corrupted speech at selected SNR is generated by adding speech and noise digitally. For lower level of noise no need of using cascaded circuit. But as the noise ratio in the input increases, cascade of different filters are required.

VI. CONCLUSIONS

During the course of experiments, we have found that SNR tests alone can't reflect the effectiveness of a de-noising system and hence results are to be confirmed with listening tests. LMS algorithm can't be used alone as it provides trivial improvement. However it can be used as a preprocessing algorithm owing to its noise cancellation and channel equalization features.

Normalized Least Mean Square eliminates a good deal of noise however the Residual noise is heavy and undesirable as a good amount of vital speech information sometimes get subtracted. But this algorithm stops processing as soon as we achieve the target. From the SNR values of algorithms, we learn that SNR decreases drastically with increasing noise. By cascaded systems we can eliminate large amounts of noise whereas for lower levels of noise, any single algorithm (preferably Kalman Filter) will do. By pipelining NLMS with RLS, we slightly improve the efficiency of the system, resulting in providing stability with increasing noise proportions. Though cascade may show appreciable SNR improvements. NLMS-RLS is far better than any other algorithm. Moreover, the efficiency of this system varies only a little with increasing noise.

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A Game Theoretic Framework for E-Mail Detection and Forgery Analysis

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Abstract—In email forensic, the email detection and forgery conflict is an interdependent strategy selection process, and there exists complex dynamics between the detector and the forger, who have conflicting objectives and influence each other's performance and decisions. This paper aims to study their dynamics from the perspective of game theory. We firstly analyze the email basic structure and header information, then discuss the email detection and forgery technologies. In this paper, we propose a Detection-Forgery Game (DFG) model and make a classification of players' strategy with the Operation Complexity (OC). In the DFG model, we regard the interactions between the detector and the forger as a two-player, non-cooperative, non-zero-sum and finite strategic game, and formulate the Nash Equilibrium. The optimal detection and forgery strategies with minimizing cost and maximizing reward will be found by using the model. Finally, we perform empirical experiments to verify the effectiveness and feasibility of the model.

Keywords—email detection; email forgery; game theoretic model; Nash Equilibrium; the optimal strategy

I. INTRODUCTION

E-mail is ubiquitous in the contemporary commercial environment. Because of its convenience, low cost and rich content, it becomes one of the most widely used applications for people to transmit information on the Internet. However, the widespread use of the email has made it a common tool and carrier for criminals to commit criminal activities. Meanwhile, the forensic investigators are more often to take the email as evidence of criminal cases. Therefore, technical appraisal of email plays an increasingly important role in solving cases and providing evidence in the court.

To verify the authenticity of email evidence, a scientific appraisal technology is of great concern. However, as email is complicated, with different protocols for receiving and sending, various email server software and client, research on email technical appraisal is almost a blank in nowadays. Genwei Liao proposed the basic ideas to identify the authenticity of email from following parts: email header, server log file, email sending environment, email content abnormal and the logic between emails[1]. M.T Banday and Hong Guo et

al. studied the working principle of an email, discussed the construction mechanism of keywords that commonly used in the header field, and applied the analysis to email forensic[2, 3]. Based on the email header information, Preeti and Surekha et al. provided an algorithm to identify the data, time, and address spoofing[4, 5]. Email authenticate is challenging due to not only the flexibility of composing, editing, deleting of emails by using offline or online applications, but also the various fields that can be forged by hackers or malicious users. However, the current researches study the emails authentic identification only from the perspective of detecting but not forging.

To have a better performance on emails authentic identification, detectors can consider what method had falsifiers taken to fabricate emails. The game theoretical analysis is useful for analyzing, modeling and deciding for the interdependent and antagonistic relationship. The player can have a prediction of other players' action and strategy in the game theory model. Recently, many literatures on the forensic and anti-forensic with game theoretical framework have been proposed. Mauro Barni et al. used the game model to solve the optimum forensic and counter-forensic strategies in source identification with training data[6].The model is used to derive the Nash equilibrium and the condition under which the false negative error probability tends to zero. Matthew C Stamm et al. developed a theoretical understanding of interactions between a falsifier who uses anti-forensics and a forensic investigator, and decided the optimum decision rule by predicting the falsifier's best anti-forensic strength [7]. Xiangui Kang et al. defined a VIF (Video Inter-frame Forgery) game to analyze the interplay between the forensic investigator and the falsifier, and used the Nash equilibrium strategy to decide under which false alarm rate can the detection rate reach 100%[8]. These studies demonstrate the efficiency of game model in solving the optimal strategy. However, the previous studies are almost based on the multimedia forensic, and the similar research on email forensic is still a blank. And there is no related research had a discussion of how can a forensic investigator predict the forger's action by introducing the game theory into email forensic.

Our work is different from the state-of-the-art studies in several aspects. Firstly, we make a new classification and cost-benefit quantification for the existing email forgery methods and authenticity appraisal technologies. Before we make a prediction with someone’s action, we need to have a basic understanding.

The strategies classification and cost-benefit quantification can help detectors know the factors that will affect forger’s decision and action. Secondly, we take both email forensic and game theory into consideration simultaneously and propose a DFG game model to analyze the dynamics between email detection and forgery for the first time. The DFG model aims to help the detector and forger find out the trade-offs that depend upon the actions of another. Thirdly, we propose an algorithm to solve the Nash equilibrium of the DFG model. Then the optimal strategy with the maximum benefits and minimum risks can be found for the players.

The rest of paper is organized as follows. In section 2, we study the email header fields and information, and have a discussion on the email forensic and forgery technologies. In section 3, we will give a formalization definition to DFG model, make a strategy classification and cost-benefit quantification for the detector and forger, and introduce an optimal strategy selection algorithm. The experimental work is discussed in section 4. The conclusion and future work are discussed in section 5.

II. E-MAIL DETECTION AND FORGERY ANALYSIS

Electronic mail, often called email, is a method of exchanging digital messages from an editor to one or more recipients. The email relevant rules are defined by the RFC (Request for Comments), a series of number ranked memorandums issued by the IETF(Internet Engineering Task Force)[9].

A. E-mail Header Analysis

An Internet email messages consist of two major sections: header fields and body. The email header is divided into several fields and each field has a name and value. The email header contains the sender and recipient information, time and data information, email sever information, email transfer information and other relevant information, which plays an essential role to ensure the authenticity of an email. The basic header fields that have been defined in RFCS are show in Table I[3].

In addition to the basic fields, there are some non-standard, custom fields generated by different mail client, which begin with X-, such as ‘X-Sender’, ‘X-Mailer’, ‘X-SMAIL-MID’, ‘X-Received’, ‘X-Originating-IP’ and so on. Expecting for these custom fields, some fields generated because of the security technology used by the mail server, such as ‘DKIM-Signature’, ‘Received-SPF’, ‘Sender-ID’. All these header fields are of great importance for email authenticity appraising.

TABLE I. BASIC FIELDS

Field name	Description
From	The email address, and optionally the name of the sender.
To	The email address, and optionally the name of the message recipient.
Date	The local time and date when the message was written.
Subject	A brief summary of the topic of the message.
Message-ID	An automatically generated field, it uniquely identifies this message.
Reply-To	Address that should be used to reply to the message.
Received	Tracking information generated by mail servers that have previously handled a message, in reverse order.
Content-Type	The type of the message content
Content-Transfer-Encoding	The transfer and encoding ways of message content.

B. E-mail Detection and Forgery Analysis

Email spoofing is one of the biggest challenges that threatens email security, and the main important forms of email spoofing are data and time spoofing, address spoofing and content spoofing. Generally, an email may be required to be appraised in following three conditions: firstly, the sender or recipient does not recognize they sent or received the email; Secondly, the sender and recipient have objections on the email date and time. Thirdly, the litigants don’t reach an agreement on the email content. The Fig.1 shows the header message of an email which has a question on the sender address.



Fig. 1. An email’s header message, there are two different senders 543954686@qq.com and cqdyt2009@163.com in the header message while the sender on the email envelop is 543954686@qq.com

To find out the real sender, we can analyze the email header information. There are many fields include sender information, such as ‘X-Sender’, ‘Authentication-Results’, ‘From’, ‘Message-id’ and ‘Sender’. Among these fields, only the ‘From’ field refers to 543954686@qq.com, and the ‘From’ field is created by the author. Meanwhile, there are four fields referring to cqdyt2009@163.com, especially the ‘Message-id’, it was an automatically generated field and not easy to be changed.

After the multi-fields correlation analysis of sender, we can appraise the email sender 543954686@qq.com is forged and the real sender is cqdydt2009@163.com.

In fact, the email forgery and detection methods are various, and the example above only represents one situation. For example, modifying the system properties is the most convenient way to falsify an email, and we can falsify the email data by modifying the system time. The Simple Mail Transfer Protocol (SMTP) is an email transfer protocol, and we can use Telnet command to tamper the email address by logging in the SMTP server. We can also use the off-the-shelf software and website to forge emails. The most complex method is to steal someone's email password and imitate him to send emails, but it is not easy to know others' password because of the Encryption software and algorithms. Most people fabricate emails with a certain purpose, may be just a joke but the more is for some profits.

To protect people's profits from the email forgery, various detection strategies need to be taken. Viewing the email header information is the simplest method to detect an email. The multi-fields correlation analysis denotes to analyze a series of fields including one message. For example, the 'Received', 'Data', 'Message-ID', and 'Boundary' filed are all including the email data and time. And we appraise the email by contrasting the times of these fields.

We can also use the sender related fields to identify the true address. Making use of external resources means we can take use of off-the-shelf software like 'nslookup' to analyze the IP and DNS, or other information like login and server files to identify the email. Multi-emails correlation analysis indicates that we can identify if the emails are authentic by analyzing the logical relationship among emails, comparing the client, writing habits, IP, address and so on.

Since the methods are so various, how can the detector know which detection strategy is the most effective, and how can the forger know which forgery strategy can bring him the maximum benefits and minimum risk?

III. DETECTION-FORGERY GAME MODEL

Game theory is a study of strategic decision making. Specifically, it is "the study of mathematical models of conflict and cooperation between intelligent rational decision-makers". It attempts to determine mathematically and logically the actions that "players" should take to secure the best outcomes for themselves in a wide array of "games"[10].

A. Detection-Forgery Game Model Definition

A game theoretical model includes three basic elements: Player, Strategy set and Payoff function. The strategic form of a detection-forgery game is a 3-tuple $DFG=(N,S,U)$ [11]:

- $N = (N_1, N_2 \dots N_n)$ is a set of players. Players are the decision-makers who decide the action and strategy to maximize their own interests. And in this game model, the players are detector N_d and forger N_f .
- $S = (S_1, S_2 \dots S_n)$ is a set of players' strategies. $\forall i \in n, S_i \neq \emptyset, S_i = (S_1^i, S_2^i, \dots, S_m^i)$ is the strategy set

of player i. And here we define the strategy sets as $S_d = (S_1^d, S_2^d, \dots, S_n^d)$ and $S_f = (S_1^f, S_2^f, \dots, S_m^f)$.

- $U = (U_1, U_2 \dots U_n)$ is the payoff function set of the players. It reflects the gain and utility the players can gain from the game. We define the detector's payoff as U_d , and the forger's payoff as U_f .

Definition1: Nash Equilibrium (NE) is a solution concept of a non-cooperative game, it means each player gains the maximum benefits. In $DFG = (N_d, N_f), (S_d, S_f), (U_d, U_f)$, the strategy group (S_*^d, S_*^f) is a Nash equilibrium if and only if for $\forall S^d \in S_d, U_d(S_*^d, S_*^f) > U_d(S^d, S_*^f)$, and for $\forall S^f \in S_f, U_f(S_*^d, S_*^f) > U_f(S_*^d, S^f)$.

In a complete information game model, we can use the definition 1 to solve all the possible Nash equilibrium. In the DFG model, $\forall S_i^d \in S_d, \forall S_i^f \in S_f, U_d(S_i^d, S_j^f), U_f(S_i^d, S_j^f)$ represents the detector and forger's payoff while the detector selects the strategy i to detect the email which is forged by strategy j. The Fig.2 shows the corresponding strategy game, where each row represents the detector's strategy and each column represents the forgers' strategy, and the values in the matrix are the payoffs associate to the players.

$$U = \begin{matrix} & S_1^f & S_2^f & S_3^f & S_4^f \\ \begin{matrix} S_1^d \\ S_2^d \\ S_3^d \\ S_4^d \end{matrix} & \begin{pmatrix} U_{d11}, U_{f11} & U_{d12}, U_{f12} & U_{d13}, U_{f13} & U_{d14}, U_{f14} \\ U_{d21}, U_{f21} & U_{d22}, U_{f22} & U_{d23}, U_{f23} & U_{d24}, U_{f24} \\ U_{d31}, U_{f31} & U_{d32}, U_{f32} & U_{d33}, U_{f33} & U_{d34}, U_{f34} \\ U_{d41}, U_{f41} & U_{d42}, U_{f42} & U_{d43}, U_{f43} & U_{d44}, U_{f44} \end{pmatrix} \end{matrix}$$

Fig. 2. The DFG payoff matrix

B. The Classification of Strategies

In the DFG model, the players' strategy set is a necessary component. In this paper, we mainly discuss the forgers' and detectors' strategies based on the email header, and classify the strategies according to Operation Complexity (OC). Richard E Overill[12] used the Operation Complexity(OC) to enable the complexity of both the cognitive and the computational components of a process, and the more complex a process is, the less likely it is to occur accidentally, unintentionally or spontaneously. Similarly, we use the operation complexity to measure the complexity or difficulty of a detection or forgery strategy. Generally, the more complex a strategy is, the higher costs it takes. This can be evaluated according to the amount of extra resources or the steps the players take. For any detection or forgery strategy i, the operational complexity of that strategy can be given by:

$$OC_i = KLM_i + R_i \tag{1}$$

Where OC_i comprises a cognitive complexity component KLM_i and an extra resource component R_i . The KLM is specified by the GOMS-KLM model for measuring the human involvement in the operational process[13] and the R represents the size of files for sending an email. The basic unit of the GOMS-KLM characterization of cognitive information processing is taken to be the mouse button press or release;

similarly, the basic unit of information processing used in characterizing the resource is the byte. In addition, the cognitive component should be scaled by the ratio of the processing rates of the human and computer, typically $\approx 10^6$. Table II shows the KLM operators and normal values[12] and Table III shows an example of the frequent KLM actions and values of modifying the system time to send a false email, and the total value is 62.6. Since this strategy needs no extra resource expect an email client or login an email website, such as Foxmail7.2, then the R is 15,624,827, so the OC=78,224,827.

TABLE II. KLM OPERATORS AND NORMAL VALUES

KLM operators	normal values(sec)
K(key press & release)	0.2
P (point mouse)	1.1
B (button press/ release)	0.1
H (hand to/from keyboard)	0.4
M (mental preparation)	1.2

TABLE III. THE FREQUENT KLM ACTIONS AND VALUES

Action	M	P	B	K	H	Total
1.point with mouse to the target	1	1	2	0	0	2.5
2. single click	4	4	8	0	0	10
3. Modify the time	1	1	2	12	2	5.7
4. open the foxmail	1	1	4	0	0	2.7
5. log in(username and pw)	4	2	4	28	4	14.6
6. write the email (the email content is the least)	6	3	6	66	6	26.7

In order to have a better strategy classification, we can divide the operation complexity into three relative levels:

- L1: The cost is very small and the $OC < 10^9$. For example, modifying the system time needs only an email client, and the operation needs simple steps;
- L2: The operation needs some time and resource and the $OC < 10^{10}$. For example, using telnet command to falsify the email address needs little resources, and the operational step is complex and time-consuming;
- L3: The operation needs much more resource and the $OC < 10^{11}$. For example, stealing other’s password not only needs many resources but the operational step is more complex and time-consuming.

Based on the email detection and forgery analysis and the definition of three level operation complexity above, the email detection- and forgery- strategy taxonomy according to the operation complexity are show in Table IV and Table V.

TABLE IV. DETECTION STRATEGY TAXONOMY

Category	Example	OC
No detect	*	0
View the email header	Received, X-Sender	L1
Multi-fields correlation analysis	Time analysis:Received, Date, Message-ID, Boundary	L2
Make use of external resources	nslookup, login files, server information	L3
Multi-emails correlation analysis	Logic, client, writing habit, IP, DNS...	L3

TABLE V. FORGERY STRATEGY TAXONOMY

Category	Example	OC
No forge	*	0
Modify the system properties	Modify the system time	L1
Make use of transfer protocol	Use telnet to falsify the address	L2
Make use of external resources	Email forgery software or website	L2
Steal password	Implant Trojan to the target computer	L3

C. Cost-Benefit Quantification

In order to make the payoff function more exactly and actually, we need to quantify the costs, risks and benefits. The relevant cost factors are defined as follows[14]:

Definition2: Detect Cost (DC) characterizes the amount of resources of implanting a detect action, such as hardware and software resource, expertise, time and so on.

Definition3:Detect Damage (DD) characterizes the amount of damage or risks inflicted by the detector if he can’t identify the forged email or treat the forged email as the authentic one. (Expressed in negative values)

Definition4:Detect Benefit (DB) characterizes the amount of benefits inflicted by the detector or the extra-reward if he successfully detected the forged email.

Definition5:Forge Cost (FC) characterizes the amount of resource of implanting a fake action, such as hardware and software resource, expertise, time and so on.

Definition6:Forge Damage (FD) characterizes the amount of damage or the legal penalties to the forger which is inflicted by the detector if he can identify the forged email successfully (Expressed in negative values).

Definition7:Forge Benefit (FB) characterizes the amount of benefits if the forger escaping from the forgery detection.

Definition8:Detection Rate P indicated the possibility that a detect method can successfully identify the forged email as forgery. If there is an email E that has been manipulated using an editing operation $m(*)$, then we assume the null hypothesis H_1 is that E is unaltered and authentic; The alternated hypothesis H_2 is that E is a manipulated version of another email E_1 and E is forged, i.e.

$$\begin{aligned} H_1 : E &= m(E_1) \\ H_2 : E &= m(E_1) \end{aligned} \quad \square\square\square(2)$$

Then we do experiments on a large number of emails that include forged and authentic, and the detection rate P defined as follows:

$$P = P\left(\frac{H = H_2}{E = m(E_1)}\right) \quad \square(3)$$

Where $(H = H_2)$ means we appraise the email is forged, and $E = m(E_1)$ means the email is actually forged. Based on the definitions above, we can define the detailed rewards of detector and forger as follows:

$$U_d = DB * P + DD * (1 - P) - DC \quad (4)$$

$$U_f = FB * (1 - P) + FD * P - FC \quad (5)$$

From the (4) and (5), we can find that if the detector and forger want to maximize their rewards, the detector needs to maximize the detection rate P while the forger needs to minimize it. However, the detection rate depends not only on the cost of detecting but on the cost of forging. The more detection cost, the higher P, and the more forgery cost, the lower P. So we need to find out an optimal P to maximize both the forger's and detector's rewards.

D. Optimal Strategy Selection

The detection strategy with highest possibility and maximum rewards to appraise emails, and the forgery strategy with maximum possible and rewards to falsify emails can be selected from the candidate sets through DFG model. And the detail of the optimal strategy selection process is presented as follows:

Input: detect and forge strategy set

Output: optimal strategy

Algorithm:

- 1) Construct the detector's strategy set $S_d = (S_1^d, S_2^d, \dots, S_n^d)$;
- 2) Construct the forger's strategy set $S_f = (S_1^f, S_2^f, \dots, S_m^f)$;
- 3) Initialize the $DFG = ((N_d, N_f), (S_d, S_f), (U_d, U_f))$;
- 4) For all $s_i^d \in S_d, s_j^f \in S_f$, compute the defection rate P according to the (2) and (3), and compute the rewards of detector and forger according to the (4) and (5), and get the payoff matrix U_d, U_f ;
- 5) Set the utility matrix $U = U_d - U_f$;
- 6) Compute the Nash Equilibrium of DFG. Processes as follows:
 - a) Test for the saddle point in the utility matrix;
 - b) If there is a saddle point, the saddle point is the Nash Equilibrium;
 - c) If there is no saddle point, solve it by linear program. Processes as follows:
 - i). maximize Z, minimize W;
 - ii). subject to
 - iii). for all $s_i^d \in S_d, s_j^f \in S_f$;
 - iv). $\sum_{i=1}^m p_i^d U_d(S_i^d, S_j^f) \geq Z$;
 - v). $\sum_{j=1}^n p_j^f U_f(S_i^d, S_j^f) \leq W$;
 - vi). $\sum_{i=1}^m p_i^d = 1, p_i^d \geq 0$;
 - vii). $\sum_{j=1}^n p_j^f = 1, p_j^f \geq 0$;
 - viii). Get the Nash equilibrium (p_d^*, p_f^*) ;
- 7) Decide the optimal strategy.

IV. NUMERICAL ANALYSIS

In order to verify the effectiveness and feasibility of the DFG model, we need to introduce the model into the actual email authenticity identification cases. Since the email address spoofing case is the most widely happened in civil disputes, so we consider the counterwork between the detector and forger as follows: A and B are business partners, and A used to order

goods from B through emails. However, one day B received an email and the sender on the envelope is A, but A denied to have sent the email and received the goods. We consider the envelop of all the emails shown as Fig.3.

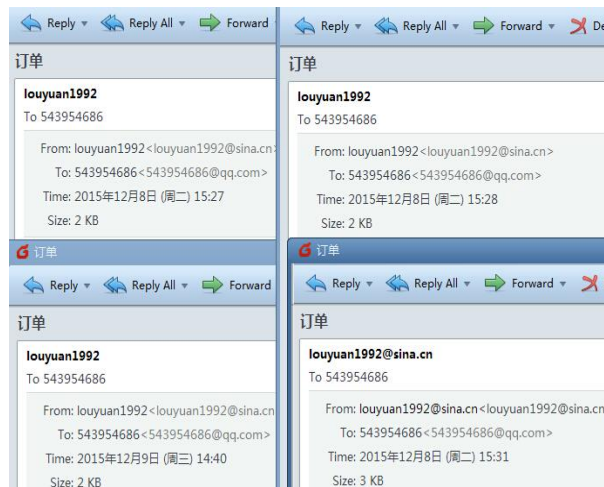


Fig. 3. The envelop of emails, the emails above have the same sender and recipient shown in the envelop, but not all the emails are authentic, some of them are forged, the sender is not the real one

In this case, we considered there are different strategies for a forger to fabricate an email with false sender, and the methods to identify the email's real sender are also various for the detector. In real society, the mostly related money of E-mail appraisal cases is almost from ten thousand to ten million, and the influence of the appraisal results is from 0% to 100%. In this case, the influence of appraisal result is 100%, and the basic unit of the cost and benefit is one thousand dollar. Since the more complex, the higher cost, here we set the cost of L1 from 0-10, the cost of L2 from 10-40, the cost of L3 from 40-100. Table VI, VII shows the strategies, benefits, costs of the email forger and detector.

TABLE VI. SUMMARY OF FORGERY STRATEGY

Forger	strategy	FB	FD	FC
S_1^f	Use telnet to login in SMTP for address tampered	100	-100	15
S_2^f	Send a forgery email by an email fake website	100	-100	15
S_3^f	Use off-the-shelf software to send a forgery email	100	-100	20
S_4^f	Build a local email server to send a forgery email	100	-100	30

TABLE VII. SUMMARY OF DETECTION STRATEGY

Detector	strategy	DB	DD	DC
S_1^d	No detect	0	-100	0
S_2^d	View the email header, X-sender	100	-100	5
S_3^d	Multi-fields correlation analysis, such as From, X-Sender, Received, Message-ID	100	-100	20
S_4^d	Multi-emails correlation analysis	100	-100	50

According to the tables above, we fabricated numbers of email with the forgery strategy set and appraised emails with the given detection strategies. Then we use the optimal strategy

selection algorithm to solve the optimal solution. According to the (2),(3),(4),(5), we can get the detection rate P matrices and detection-forgery payoff matrices U_d, U_f as follows:

$$P = \begin{matrix} & S_1^f & S_2^f & S_3^f & S_4^f \\ \begin{matrix} S_1^d \\ S_2^d \\ S_3^d \\ S_4^d \end{matrix} & \begin{pmatrix} 0 & 0 & 0 & 0 \\ 0.9 & 0.15 & 0.1 & 0.2 \\ 0.95 & 0.2 & 0.3 & 0.25 \\ 0.95 & 0.85 & 0.8 & 0.78 \end{pmatrix} \end{matrix} \quad U_f = \begin{matrix} & S_1^f & S_2^f & S_3^f & S_4^f \\ \begin{matrix} S_1^d \\ S_2^d \\ S_3^d \\ S_4^d \end{matrix} & \begin{pmatrix} 85 & 85 & 80 & 70 \\ -95 & 55 & 60 & 30 \\ -105 & 45 & 20 & 20 \\ -105 & -85 & -80 & -86 \end{pmatrix} \end{matrix}$$

$$U_d = \begin{matrix} & S_1^f & S_2^f & S_3^f & S_4^f \\ \begin{matrix} S_1^d \\ S_2^d \\ S_3^d \\ S_4^d \end{matrix} & \begin{pmatrix} -100 & -100 & -100 & -100 \\ 75 & -75 & -85 & -65 \\ 70 & -80 & -60 & -70 \\ 40 & 20 & 10 & 6 \end{pmatrix} \end{matrix} \quad U = \begin{matrix} & S_1^f & S_2^f & S_3^f & S_4^f \\ \begin{matrix} S_1^d \\ S_2^d \\ S_3^d \\ S_4^d \end{matrix} & \begin{pmatrix} -185 & -185 & -180 & -170 \\ 170 & -130 & -145 & -95 \\ -175 & -125 & -80 & -90 \\ 145 & 105 & 90 & 92 \end{pmatrix} \end{matrix}$$

An equilibrium $P_d = (p_1^d, p_2^d, p_3^d, p_4^d) = (0, 0.0083, 0, 0.9917)$, $Z = 90.4545$; $P_f = (p_1^f, p_2^f, p_3^f, p_4^f) = (0, 0, 1, 0)$, $M = 90$ can be found by the optimal strategy selection algorithm. Therefore, the detector plays the strategy S_2^d with the possibility 0.0083 and the strategy S_4^d with the possibility 0.9917, and the strategy S_3^f is the optimal strategy for the forger. From the above results, we can find that the strategy set (S_4^d, S_3^f) is an optimal strategy for the example case. This result indicates that the forger is most likely to use off-the-shelf software to fabricate an email with false sender, and the forensic investigator will have a maximum reward by making correlation analysis with multi-emails when facing such cases.

V. CONCLUSION

In this paper, we have proposed a DFG game model for analyzing optimal detect and forge strategy decision in email authenticity identification. We regard the interactions between a forensic investigator and a forger as a two-player, non-cooperative, nonzero-sum game and formulated the DFG game model. Based on the strategies' cost-benefit quantification and DFG model, we selected the optimal strategy from the given sets. And finally, we used a practical case study to verify the effectiveness of the DFG model. Nevertheless, there are still some problems of the DFG model, such as the cost-benefits quantification and payoff functions we adopted in this paper is

not very comprehensive. We will pay more attention to improving it in the future work.

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Review of Energy Reduction Techniques for Green Cloud Computing

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Abstract—The growth of cloud computing has led to uneconomical energy consumption in data processing, storage, and communications. This is unfriendly to the environment, because of the carbon emissions. Therefore, green IT is required to save the environment. The green cloud computing (GCC) approach is part of green IT; it aims to reduce the carbon footprint of datacenters by reducing their energy consumption. The GCC is a broad and exciting field for research. A plethora of research has emerged aiming to support the GCC vision by improving the utilization of computing resources from different aspects, such as: software optimization, hardware optimization, and network optimization techniques. This paper overviews the approaches to GCC and classifies them. Such a classification assists in comparisons between GCC approaches by identifying the key implementation approaches and the issues related to each.

Keywords—Cloud Computing; Green Computing; Energy Efficiency; Power Management; Virtualization

I. INTRODUCTION

Cloud computing is a new computing paradigm that relies on a business model whereby services, such as servers, storage and applications, are delivered to users' devices from the Internet[1][11][12]. To provide and deliver the services to users' devices, cloud computing utilizes a large number of datacenters [13].

Each datacenter consists of hundreds or thousands of physical machines arranged in hundreds of racks that can run millions of virtual machines (VMs)[5]. For instant, Google, one of the most famous cloud-based companies, delivers all of its services through the cloud, such as Gmail and Google Earth. To deliver this content to end users in real-time, videos, pictures and other data are stored in huge datacenters [8]. It has at least 14 datacenters around the world [9], and more than two million servers, as estimated in [10].

Virtualization is becoming a hot topic in Information Technology. The concept began in the 1960s, when IBM partitioned mainframe computers to increase processor utilization [14]. In cloud computing, virtualization is the process of logically dividing a server's resources. Each physical server is partitioned to contain multiple independent logical servers, as VMs [12]. Once the physical server is partitioned, each virtual server runs an independent operating system and applications.

The main advantages of VMs are improved portability, manageability, maintenance effort, and security [15]. VMs also provide isolation, meaning that a VM does not affect any other VM in the physical machine. Also, it prevents the guest operating system from directly accessing the real hardware. Furthermore, it improves hardware utilization by up to 70% [2], by reducing the number of physical servers necessary to store and process data. In contrast, VMs suffer from performance degradation due to the overheads associated with creating, running, and maintaining VMs on the physical machine [14]. Also, they suffer from single point of failure problems, because many VMs are dependent on one physical machine; failure in one physical machine will cause failure of many VMs [12].

A datacenter is a massive facility that consumes large amounts of energy for data processing, storage and communication, which negatively impacts the environment [6]. The environmental impact is the resulting carbon emission; one datacenter can produce 170 million metric tons of carbon per year. The expected carbon emissions by datacenters worldwide in 2020 is 670 million metric tons annually [7]. Additionally, the huge energy consumption in datacenters causes high operational costs. The total estimated energy for datacenters in 2013 was 91 billion kilowatt-hours of electricity [6]. As indicated in [5], the typical datacenter consumes as much energy as 25,000 households per year. As a result of such potential impacts to the environment, the green cloud computing initiative has emerged as part of the green IT vision [16]. The overall objective of green IT is to increase energy efficiency and reduce CO₂ emissions to save the environment. Since 2007 it has received growing attention when the United States Environmental Protection Agency (EPA) submitted a report to the United States Congress about the expected energy consumption by datacenters. In 2010, President Obama invested 90 million dollars in green initiatives via the American Recovery and Reinvestment Act (ARRA). The United States Energy Department granted \$47 million of the ARRA money towards projects that optimize datacenter software and hardware [17][18].

The rest of this paper is structured as follows: **Section II** presents an overview of energy reduction techniques. In **Section III**, the software techniques and related research are presented. **Section IV** discusses various research into hardware optimization techniques. In **Section V**, network techniques and

related studies are presented. **Section VI** presents our conclusions.

II. OVERVIEW OF ENERGY REDUCTION TECHNIQUES

In general, green cloud computing can be implemented via three approaches: software optimization [2] [19] [11] [20] [21] [22], hardware optimization [23][24][25][26], or network optimization [6][27][28][29][30] in order to reduce the power consumption, as illustrated in Fig. 1. Table 1 summarizes the related studies of green cloud implementation techniques.

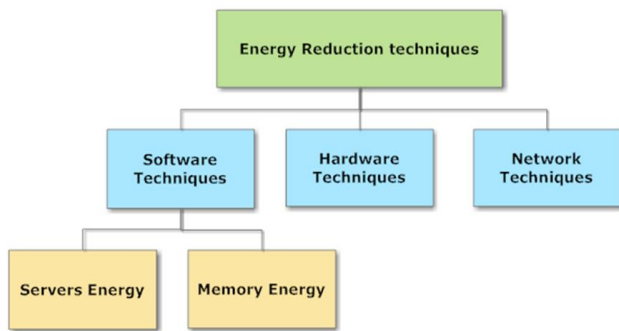


Fig. 1. Energy Reduction Techniques for GCC

III. SOFTWARE TECHNIQUES

There are two software approaches for energy consumption reduction: reducing the energy consumed by servers (by reducing the number of active servers), and reducing the energy consumed by memory (by reducing the number of running memory nodes).

A. Reducing Server Consumed Energy

The energy consumption of servers can be decreased by reducing the number of active servers. This is usually implemented by scheduling optimization, which is a common approach for green clouds and is considered [31] more efficient than hardware optimization, in terms of cost, consumed resources and scalability. It depends on finding a suitable mapping between requests for VMs and physical servers to minimize the amount of consumed power [31].

One of the important issues for energy efficiency in virtualized cloud environments is where to place new VM requests within the physical servers. In [2], they proposed a heuristic-based energy-efficient approach for VM placement in cloud based datacenter which relies on statistical analysis of historical data. It uses the multiple correlation coefficient (MCC) method; i.e. measuring the strength of association between a given variable and other variables; to select the server that provides a suitable trade-off between power efficiency and SLA violation.

Higher the correlation coefficient of CPU utilization between selected VMs running on a server, the higher the risk of SLA violation in the datacenter. They used the CloudSim 3.0 toolkit to simulate a datacenter with heterogeneous physical hosts and computed the energy consumption. However, this algorithm requires information from the hardware level [2].

An inspired approach is proposed in [19] based on the behavior of real ants. It uses the Ant Colony Optimization (ACO) meta-heuristic for VM placement. It aims to minimize the number of active servers by maximizing the resources utilization. VM placement is computed dynamically according to the current server load. In this algorithm, each ant (server) receives all VMs, and starts scheduling them to the servers. After all ants have built their solutions, the solution with lowest value for objective function is chosen. The pheromone trails are updated for only this solution. They have developed their own java-based simulation toolkit [19]. As the case with all ACO techniques, this algorithm takes a long time to converge [32]. It also wastes a lot of resources as each ant must compute its own solution, but then only the one with the lowest value of objective function will be considered.

A server consolidation algorithm (Sercon) is proposed in [11]. It aims to reduce the energy consumption in homogeneous datacenters by minimizing the number of active servers. The algorithm inherits some properties from First- and Best-Fit (FF and BF) bin packing problems. First, it sorts the servers in decreasing order based on their load. Then the VMs in the least loaded server become candidates for migration. Thereafter, those VMs are sorted in decreasing order based on their weights. After that, those VMs are allocated one-by-one to the most loaded servers. Therefore, the least loaded server will be idle. So the number of running servers is reduced by switching off idle servers, reducing energy consumption. A simulation software is developed using the .NET 3.5 framework to evaluate an experiment of the proposed method. However, the algorithm is fully centralized, considers homogenous servers, and prevents the server's processor from being fully utilized.

VM migration technique can be used to optimize scheduling. This focuses on transferring VMs between servers via the network. It is used as a solution for improving energy efficiency, by consolidating the VMs on fewer physical servers [20][21]. In [20] both scheduling and VM migration methods were used to reduce the energy consumed by servers. It proposed the Energy-aware Scheduling algorithm using Workload-aware Consolidation Technique (ESWCT). The algorithm aims to consolidate the VMs in minimum amount of servers based on balancing the integrated resources (processor, memory and network bandwidth) which are shared concurrently among users in cloud datacenters. It considers heterogeneous workloads of various resource consumption characteristics. The aim of this algorithm is to reduce the power consumption by improving resources utilization based on the fact that heterogeneous workloads have different resource consumption characteristics. The algorithm is centralized and consists of two phases: the first phase shows where to place the VM to get a better balanced utilization of resource among physical servers. First, it computes the processor, memory, network capabilities for each physical server, and the requirements of the VM. Then it selects the server with the smallest value of Imbalanced Utilization Value (IUV) and assigns the new VM to that sever. The IUV is calculated by a mathematical function that considers different resources utilizations (processor, memory, network bandwidth) considering the integrated resources utilization and the current

average resources utilization of the server. The second phase of the algorithm, ELMWCT, optimizes the current VM allocation and deals with where to migrate VMs. It migrates running VMs from underutilized servers to others which are more fully utilized. At the first step, the algorithm chooses the VMs to be migrated (VMs in servers where utilization of processor, memory or network are below threshold). At the second step, the chosen VMs are allocated to the selected physical servers. The performance measures are the IUUV and the integrated resource consumption. CloudSim is used as a simulation platform to evaluate the algorithms. The simulation results of the two algorithms showed that multi-dimensional resources have well-balanced utilizations and good power savings compared to other methods. However, the algorithm is fully centralized and includes computing overheads associated with calculating IUUV for each server before every VM allocation.

In [21], they propose an approach for VM placement and migration to deal with both over-utilized and under-utilized servers. In the VM placement, they applied a modification of the Best Fit Decreasing algorithm called Power Aware Best Fit Decreasing (PABFD), to allocate a new request for a VM to a server that provides the least increase in power consumption caused by the allocation. They modeled the VM placement problem as a bin packing problem with variable bin sizes and prices; bin sizes are the available processor capacities of the nodes; and prices correspond to the power consumption by the nodes. Bins represent the physical nodes, items are the VMs that have to be allocated. They apply the PABFD method by sorting all VMs in decreasing order of their current processor utilizations and allocate each VM to the server that provides the least increase of power consumption caused by the allocation. This migration approach is proposed to overcome the problem of under loaded servers. The system finds the server with the minimum utilization compared to the other servers, all of its VMs are selected for migration. The target server is selected based on a Best Fit Decreasing heuristic: the migrating VMs are sorted in decreasing order by processor utilization and placed in the server that provides the least increase in power consumption, the source server is then switched to the sleep mode (power-saving mode) once all migrations have been completed. If all the VMs from the source server cannot be placed on other servers, the server is kept active. They used the CloudSim toolkit simulation tool to evaluate the proposed algorithm. In order to evaluate the efficiency of the algorithms, the total energy consumption and the number of VM migrations are used as performance metrics.

B. Reducing Memory Consumed Energy

The article [22] proposed a technique for reducing memory energy consumption using virtual machine scheduling in multicore systems. It presented two scheduling policies: Biggest Cover Set First (BCSF), and Biggest memory Node First (BNF). Each policy makes a scheduling decision based on the currently used memory nodes at time t ($C(t)$), and the access set of the VMs (memory nodes accessed by these virtual machines) in run queues. To reduce the currently running memory nodes, it tries to find the biggest access set that is completely covered by $C(t)$ and schedules the corresponding VM. BNF schedules the VMs based on the popularity of

individual memory nodes, the number of VMs that use the memory node in the entire computer system. Memory power simulator, called MPSim, is developed to evaluate the scheduling algorithms. The consumed energy, the average elapsed time to schedule a VM and the average waiting time of VMs in running queue are measured.

IV. HARDWARE TECHNIQUES

Other technique reduce the consumed energy by utilizing flexible hardware that varies the server computing capability via controlling the frequencies and voltages in the server, which affects the energy consumption [31][33]. However, as with all other hardware techniques, this approach to green cloud is costly and suffers from poor scalability because of the special hardware requirements.

A power-aware scheduling algorithm is presented in [23]. It implements Dynamic Voltage Frequency Scaling (DVFS) technique, which is applied with a number of special processors that can to operate at different voltage and frequency levels. It selects the appropriate supply voltages and frequencies of processing elements to minimize energy consumption without violating the SLA, based on the VMs workload. Each VM is allocated to the First Fit server, and each server applies the DVFS to save the energy while complying with the SLA requirements. The result shows a reduction in energy consumption without violating the SLA, and is compared with a non-power aware algorithm. It is implemented by using CloudSim toolkit and it is provided as an example in the simulator.

The study in [24] follows the same approach but they consider the SLA based on the task level (task deadline), and scaling only the supply voltage. The scheduling algorithm applies Dynamic Voltage Scaling (DVS) to save energy while testing the ability of each scheduled task to meet its deadlines. They consider two DVS scheduling policies: one a space-shared policy and the other a time-shared policy. They simulate the proposed algorithms by using GridSim toolkit. The DVS-server is required to control its supply voltages, so it needs special or additional hardware.

The study in [25] applied DVFS technique to find the optimal frequency for each task of a scientific workflow without affecting its performance. A multi-step heuristic workflow scheduling algorithm is proposed, namely Energy-Aware Resource Efficient workflow Scheduling under Deadline constraint (EARES-D). In the first phase, they calculate the estimated earliest completion time for a workflow, in all datacenters. Then the optimal frequency for executing each task is determined by scaling down the processor frequency under the deadline constraint. The datacenter is selected based on the first and second phase. Thereafter, the task is forwarded to the selected datacenter for scheduling. The resource utilization rate is improved by reusing VM and shrinking the idle time between tasks if the deadline is still guaranteed. They used CloudSim toolkit as a simulator tool to evaluate the scheduling algorithm. The simulation results showed an improvement in energy consumption and resources utilization rate.

In [26], they develop six green task scheduling strategies for sequential tasks to minimize the total energy consumption, which are:

- Shortest Task First for Computer with Minimum Energy (STF-CME)
- Longest Task First for Computer with Minimum Energy (LTF-CME)
- Random Task for Computer with Minimum Energy (RT-CME)
- Shortest Task First for Random Computer (STF-RC),
- Longest Task First for Random Computer (LTF-RC) and
- Random Task for Random Computer (RT-RC).

The strategies have two main steps: consolidate the tasks as much as possible, and setting the same optimal speed for all tasks in each server. These green strategies are developed for heterogeneous and adjustable speeds servers to effectively reduce energy consumption and finish all tasks before the deadline. They evaluate those strategies by using the simulation, but they did not indicate which simulation. The simulation results indicate that the best policy among these policies is STF-CME strategy.

As indicated earlier, the special hardware requirements [24] limit the scalability of the hardware optimization techniques. Also, they are more costly and not feasible in all datacenters, compared with software and network optimization techniques.

V. NETWORK TECHNIQUES

The communications between VMs consumes energy in the datacenter [6]. Reducing the network traffic between servers reduces energy consumption. The studies [6][27][28][29][30] consider the network traffic of the VMs placements to reduce the energy consumption.

In [7], two heuristics for VMs migration are presented based on the communication graph and other resource requirements such as processor, memory etcetera. The communication graph is represented as a weighted graph. The weight for each edge in the communication graph shows the amount of traffic between two VMs. So the connected component means those VMs communicate with each other while disconnected components means there is no network traffic between these components. The algorithm identifies the under-loaded servers and the heavily-loaded servers. Then it identifies the physical servers with sufficient residual capacity and sorts them in ascending order according to load. From the lightly loaded servers, it identifies the set of VMs whose load can be accommodated by these physical machines and constructs the communication graph of those VMs. After that, it sorts the components in decreasing order of their size. The algorithm migrates the largest and least connected component first. Each component is migrated as a whole to a single physical machine based on the load of the VMs and the residual capacity of the target physical machine. So the VMs

with high communications with each other will be in one server. If this is not possible, either a modified breadth-first search algorithm or a modified Prim's maximum spanning tree algorithm is used to partition the VMs. Then the partitions are migrated to physical machines in proximity to each other based on their distance matrix. Thus, the network traffic between the servers is reduced resulting in less power consumption in the datacenter. They didn't implement these algorithms for the evaluation.

Study [27] optimizes the VM placements by consolidating the VMs to the minimum number of servers, and reducing the network traffic between those, to decrease the energy consumption. It consolidates the VMs with high communication flow together in order to reduce the network traffic between racks, number of active servers and number of active network elements (links and switches). It evaluated in terms of simulation to estimate the number of active servers and intra-rack traffic.

Datacenter Energy-efficient Network-aware Scheduling (DENS) is proposed in [28]. It aims to reduce the energy consumption in a datacenter by optimizing the tradeoff between task consolidation and traffic pattern distribution. The proposed DENS selects the best-fit server to execute a job based on weighted computational function that considers the load and the communicational potential at server, rack, and module levels. The proposed function converges VMs towards the maximum loaded server in the least-utilized rack with low network traffic.

The study in [29] proposes two techniques for flow migration and VMs migration. The proposed flow migration technique is called Disjoint Edge Node Divided Spanning Tree with traffic-aware Flow Migration (DENDIST-FM). It aims to generate various disjoint spanning trees to avoid overlapping paths, and chooses the least utilized path to reroute the flow. The second technique is Energy-and-Topology Aware VM Migration (ETA-VMM). It detects and migrates VMs from under- or over-utilized servers to the nearest machine based on the network distance. It evaluates a simulation using Network Simulator NS2 and CloudSim simulator. The results show an improvement in the throughput but it increase the energy consumption by 2.2% comparing with Shortest Path Bridge (SPB)[29].

The study in [30] proposes VM placement algorithm that aims to provide a balance between server energy consumption and network energy consumption. It considers the datacenter as a dependency graph, similar to [6]. It employs fuzzy logic to combining those two conflicted objectives. A detailed review of network optimization techniques for green clouds is presented in [34].

In summary, network optimization techniques provide a reduction in the energy consumption with the ability to meet the SLA. On the other hand, a datacenter is usually constructed with a fixed network topology, which limits the scalability and the flexibility in the datacenter. This approach needs to be aware of network topology to decide the flow route as in [28] [29].

TABLE I. A COMPARISON BETWEEN GREEN CLOUD COMPUTING TECHNIQUES

Optimization technique	Study	Strategy	Performance measure	Simulation tool	
Software technique	Server Energy	[2]	Utilizes MCC method to provides a balance between power consumption and SLA.	<ul style="list-style-type: none"> •Total energy consumption •SLA violation 	CloudSim toolkit.
		[19]	Utilizes ACO meta-heuristic for VM placement.	<ul style="list-style-type: none"> •Energy consumption by a placement •Average execution time 	They developed their own simulation toolkit.
		[11]	Applies VMs consolidation method by utilizing FF and BF bin packing.	<ul style="list-style-type: none"> •Number of utilized servers •Number of migrations 	They developed their own simulation toolkit.
		[20]	Considers load balancing of physical resources in VMs placement. Migrates the VMs from lightly loaded servers to heavy loaded servers.	<ul style="list-style-type: none"> •Power consumption •Imbalance utilization value •Integrated resources utilization 	CloudSim toolkit.
		[21]	Applies VMs consolidation method by utilizing PABFD based on bin packing problem. Prevents the servers to be fully utilized by maximum load threshold.	<ul style="list-style-type: none"> •Total energy consumption •Number of VM migrations. •SLA violation 	CloudSim toolkit.
	Memory Energy	[22]	Propose BCSF and BNF policies.	<ul style="list-style-type: none"> •Energy consumption. •Waiting time in run queue •Elapsed time to schedule a virtual machine 	MPSim simulator.
Hardware technique	[23]	Applies DVFS to provide a balance between power consumption and SLA.	<ul style="list-style-type: none"> •Energy consumption •SLA violation 	CloudSim toolkit.	
	[24]	Applies DVS to provide a balance between power consumption and tasks deadline.	<ul style="list-style-type: none"> •Energy consumption •Job acceptance ratio 	GridSim toolkit.	
	[25]	EARES-D utilizes DVFS to schedule DAG workflow based on earliest completion time for a workflow.	<ul style="list-style-type: none"> •Energy consumption •Resources utilization rate 	CloudSim toolkit.	
	[26]	Six different algorithms that aim to set the same optimal speed for all tasks in the server and increase its utilization under the deadline constraint.	<ul style="list-style-type: none"> •Energy consumption 	NA.	
Network technique	[6]	Applies VMs consolidation method. Reduces the network traffic between servers by using weighted graph.	<ul style="list-style-type: none"> •NA 	NA.	
	[28]	DENS provides a tradeoff between tasks consolidation and traffic patrons distribution.	<ul style="list-style-type: none"> •Server load •Traffic load 	GreenCloud simulator.	
	[29]	DENDIST-FM and ETA-VMM.	<ul style="list-style-type: none"> •Throughput •Energy consumption 	CloudSim toolkit.	
	[30]	Applies fuzzy logic to provide balance between servers energy consumption and network energy consumption.	<ul style="list-style-type: none"> •Communication cost •Resources utilization •Number of utilized servers 	NA.	

VI. CONCLUSION

In general, the growth of cloud computing has led to uneconomical energy consumption in data processing, storage and communication. The massive energy consumption is unfriendly to the environment because of the huge carbon footprints of the datacenters. Therefore, green cloud computing is required to support the environment. Green computing produces environmental-friendly and cost-efficient cloud computing by using computing resources more efficiently.

This paper overviews the GCC approaches and classifies them. This classification assists in comparisons between GCC approaches by recognizing the key implementation techniques and the related issues Three approaches can be followed to implement the green cloud computing: software optimization, hardware optimization, and network optimization.

The software optimization is easy to implement and most scalable, usually not requires special network topology or special hardware. But in the software optimization techniques,

SLA compliance and energy consumption have a negative correlation.

Hardware optimization provides a reduction in the energy consumption while complying with the SLA. On the other hand, it more costly and has a limitation in scalability because of the special hardware requirements.

Network optimization techniques can reduce energy consumption while complying with the SLA. But it needs to be aware of the network topology and can applied only in a specific network topology, which limits its scalability and the flexibility.

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A Discrete Particle Swarm Optimization to Estimate Parameters in Vision Tasks

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Abstract—The majority of manufacturers demand increasingly powerful vision systems for quality control. To have good outcomes, the installation requires an effort in the vision system tuning, for both hardware and software. As time and accuracy are important, actors are oriented to automate parameter's adjustment optimization at least in image processing. This paper suggests an approach based on discrete particle swarm optimization (DPSO) that automates software setting and provides optimal parameters for industrial vision applications. A novel update functions for our DPSO definition are suggested.

The proposed method is applied on some real examples of quality control to validate its feasibility and efficiency, which shows that the new DPSO model furnishes promising results.

Keywords—component; component; industrial vision; image processing; optimization; DPSO; quality control

I. INTRODUCTION

Mostly, in a vision system, the problem of quality control requires a very high precision in object extraction often under imposed constraints. A variety of methods have been developed since decades towards image processing in Quality Control (QC), but the problem of operators' choice and their parameters' setting is still relevant, these settings are usually made by trial and error. In fact, algorithms may be adopted after a long series of tests.

The choice of operators and their parameters' adjustment require specific knowledge, and must be done experimentally due to the lack of automatic mechanisms. In spite of vision systems' diversity and the rich library of image processing approaches scientifically strong, the user intervention in most applications remains necessary.

Few systems have succeeded in automating vision applications without requiring user's intuition. B.Nikolay and B.Schneider and S.Jacob [1] proposed a method to optimize automatically parameters of vision systems for surface inspection. Their method is based on evolutionary algorithms. The major types that have been used are: Evolution strategies (ESs) and Evolutionary Programming (EP).

Several studies have been done and few authors proposed methods such as numerical optimization, which apply mathematical or statistical techniques to minimize, or maximize, an objective function defined over a parameter space. Few years later, Taylor proposed a method based on reinforcement learning to monitor parameters in vision applications [2]. However, different techniques have been developed on the basis of population approaches: Genetic Algorithms [3], Ant Colony Optimization (ACO) [4] and Particle Swarm Optimization [5]. In such approaches, each individual (agent) in a population start by building an approximate solution. With a mechanism of interaction and evolution, individuals converge towards the optimal solution.

Most of the proposed techniques have not been widely adopted for the parameter-tuning problem. This can be partly awarded to few application examples in the real image analysis. Visually, with different parameter types and the variety of their values, the problem is NP-complex.

The objective is to deploy the artificial intelligence (AI) techniques [6] for solving such problems. The most important property in distributed AI is the cooperation between agents to provide the best solution. In DPSO, particles are considered as agents that cooperate in some way to reach this goal.

The contribution of this paper is a novel model using DPSO adapted to parameters' adjustment. In most cases, the study of a vision operator concerns a discrete domain of parameters' values; so, the study is focalized on the standard PSO reformulation in a discrete way. Optimal parameter values for applications in vision systems are found. It is shown that using a discrete PSO is more adapted to parameter tuning and allows achieving good quality results. The approach is applied on real image examples in a quality control task and the outputs are compared to references.

The reminder of this paper is organized as follows, Section 2 gives an over view on the use of image processing in quality control. Section 3 describes the proposed approach. The experimental results are presented in Section 4. Finally conclusions are stated in Section 5.

II. STATE OF THE ART

In the majority of vision tasks, the user is required and sometimes obliged to combine several operators, each one has a multitude of parameters to be adjusted, but few systems have succeeded in automating vision applications without requiring user's intuition. Some authors searched to automate completely the process, S.Treuil, D. Driouchi and P.Ribereau [7] proposed a method to adjust parameters in an image processing chain based on an experimental 2^{k-p} factorial plan applied to a vision system designed for measuring the neck ratio of a sugar beet batch. Recently L.Franek and X.jiang [8] proposed to use orthogonal plans of experiments for parameter learning. They analyze means to estimate the optimal parameter setting. In addition, a combination of orthogonal arrays and genetic algorithm is used to further improve the performance.

A different technique was introduced in [9], using interactive visualization to develop novel histopathology image segmentation software, which illustrates its potential usefulness for parameter optimization purposes. In recent years, parameters' optimization in image processing is supported by artificial intelligence techniques [10], such as multi-agent architecture. I.Qaffo, M.Sadgal and A.Elfaziki [11] proposed an automatic method based on reinforcement learning for object recognition, using two types of agents: User Agent (UA) and Parameter Agent (PA). The UA gives necessary information to the system, as the combination of applicable operators, the set of adjustable parameters for each operator, and a values' range for each parameter. Then, it generates a PA for each combination of operators. The PA uses reinforcement learning to assign the optimal values for each parameter in order to extract the object of interest from an image. One of the most used methods in parameter optimization in image processing is population-based heuristics. Genetic algorithm [12] is widely used for this purpose. We proposed an optimization method based on the standard particle swarm optimization [13] to find the best values of free algorithm parameters used in image processing. The method is restricted to operators with numerical parameters and continuous fields.

On the other hand, particle swarm optimization (PSO) was first introduced by R. Eberhart and J. Kennedy [14] in 1995, as a novel nature inspired method from social behavior of birds in a flock. It is used on optimization problems that are partially irregular, noisy, change over time, etc.

Since 2002, researches applying PSO has grown rapidly. Popularity of PSO is due to its several strengths namely: very few parameters to adjust, its easiness of implementation, its robustness, and its convergence speed.

Many problems used this paradigm, like combinatorial optimization problems including vehicle routing problem [15], traveling salesman problem [16], and scheduling problems [17] [18].

Other application areas of PSO, the most potential includes fuzzy controller design for mobile robots [19], recognition of control chart patterns [20], real time robot path planning [21], image segmentation [22], speaker verification [23] and gesture recognition [24], to name a few.

All these researches have proved the PSO efficiency as a new tool to obtain satisfying optimization results. This study adopts a discrete PSO to present a new model optimizing vision systems operators' and parameters' values. This method proves its feasibility and efficiency.

III. FORMULATION OF THE OPTIMIZATION PROBLEM

A. Vision systems and tasks

Vision systems are designed to produce applications in image processing, object recognition, etc. Such applications come with several tasks and operators that transform a product stream into an information flow. A vision system is built on a succession of tasks (some ones can be executed in parallel), which can be a set of other tasks or an elementary task composed of different phases of treatment "Fig. 1".

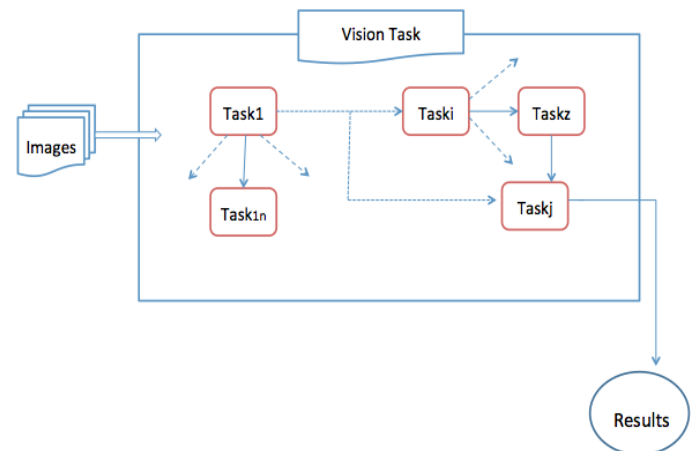


Fig. 1. General process to implement vision systems

Image processing used in the quality control domain, consists of supervising the basic parameters describing an image quality, in order to extract necessary measurements for control. The main problem here remains the segmentation quality, which affects hardly controls.

Image processing helps to increase flexibility and productivity in production factories. Further, it takes a hand in maintenance and enhances knowledge about the quality of products. Furthermore, it offers the advantage of being able to interfere at several levels, for example: the real-time quality inspection of gelatin capsules in pharmaceutical applications [25]. The designed image processing system is based on some tasks illustrated in "Fig. 2". Edge-based image segmentation technique for quality inspection is used to detect accepted and rejected capsules.

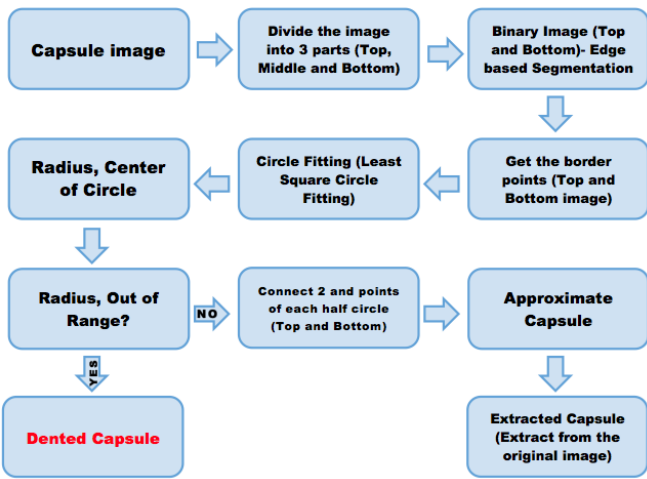


Fig. 2. Flow chart of the proposed capsule extraction method (source [25])

IV. OPERATORS AND PARAMETERS

To accomplish a vision task, it is necessary to go through multiple tasks; each one is a succession of several phases. Each phase has a set of possible operators.

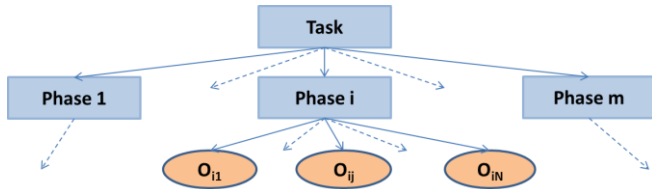


Fig. 3. Possible operators for each phase in a vision task

We may have several combinations of these operators, each combination can achieve the considered task, but the output is qualitatively different. Let's consider a task made of several phases. Since each phase concocts a set of feasible operators, n different operator's combinations C_k are built.

$$C_k = (O_1, O_2, \dots, O_N) \quad k=1 \dots n \quad (1)$$

An operator (O_i) requires fixing some parameters; a range of possible values for each parameter is given out "Fig. 4".

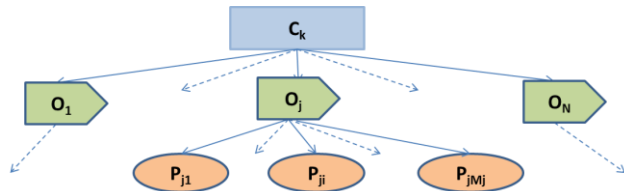


Fig. 4. Operator's combination and their parameters

The vision task performance is based on these parameter settings, and their variation strongly influences results. Below we mention some adjustable parameters of different algorithms such as edge detection.

When applying an edge operator, which identify points in a digital image at which the image brightness has discontinuities, a lot of filters can be applied here such as sobel, prewit and log, each one of these filters has a free parameter: the threshold, to remove edges with poor contrast

and then contours are formed by pixels higher than a given threshold. "Fig. 5" illustrates a segmentation done with different threshold values for a chosen filter (canny).

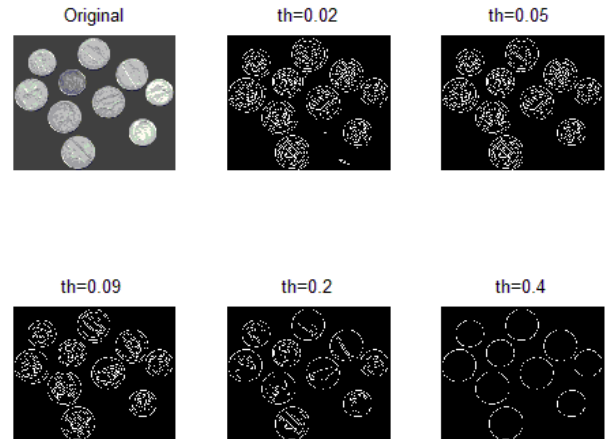


Fig. 5. Applying a canny filter with different thresholds values

A. Objective function definition

Mathematically the problem can be defined as follows: A model $P = (D, \omega, Err)$ where:

- D is the solution space, defined over a set of variables, and ω a set of constraints among the variables.
- An objective function $Err: D \rightarrow \mathbb{R}^+$ to be minimized.

A solution $d \in D$ is a complete assignment in which each variable has an assigned value; d must satisfy all the constraints. A feasible solution d^* is a global optimum if:

$$Err(d^*) < Err(d) \quad \forall d \in D \quad (2)$$

Back to our problem, a combination of operators C_k and an input image I_{input} are considered. The application of C_k provides an output result R_k (image or features):

$$R_k = C_k(I_{input}) \quad (3)$$

To evaluate the result, an objective function is necessary to compare qualitatively the outputs (R_k), a reference led by an expert is used as a ground truth R_r . The rating calculated by this objective function represents the error between the two images.

$$Err_k(C_k) = fitness(R_k, R_r) \quad (4)$$

The fitness function definition depends on the vision task. For sure optimal parameters' values give the best output. Consequently, this corresponds to the smallest error (Err), and then the operator's combination to adopt is the one corresponding to the minimal error:

$$Err = Min(Err_k(C_k)) \quad k=1 \dots n \quad (5)$$

B. State definition

Let's consider an operator's combination applied to the input image I_{input} , based on "Fig. 4", (1) and (2) we note:

$$I_{output} = (O_1, O_2, \dots, O_N)(I_{input}) \quad (6)$$

I_{output} is the result of applying operators' combination on the input image I_{input} .

To simplify, we consider a combination as a sequence of N operators where each operator is applied over the output of its predecessor:

$$I_j = O_j(I_{j-1}) \text{ for } j=1 \dots N, I_{input} = I_0, I_{output} = I_N = R_N \quad (7)$$

Considering that each operator O_j has M_j parameters and $M = \sum_{j=1}^N M_j$ is the number of all parameters. The error is simply function of parameters denoted by:

$$Err(p_1, p_2, \dots, p_M) = fitness(R_N, R_f) \quad (8)$$

Where p_i is a parameter taking values in a domain D_i . Then, we consider the cartesian product $D = D_1 * D_2 * \dots * D_M$, and we call a state of parameters' values each M-uplet $(u_1, u_2, \dots, u_M) \in D$ where u_i is a value of parameter p_i .

Using these notations, the problem is to find a state (an M-uplet) that minimizes the error function (Err) over the domain D .

So, the solution is the state $(u_1^*, u_2^*, \dots, u_M^*) \in D$, such as $Err(u_1^*, u_2^*, \dots, u_M^*)$ is minimal or:

$$(u_1^*, u_2^*, \dots, u_M^*) = ArgMin(Err(u_1, u_2, \dots, u_M)) \quad (9)$$

The objective function is not expressed directly with parameters, but it is established on the basis of results. In fact, direct numerical methods could not be applied to solve this problem. In contrast, the proposed method belongs to relaxation methods, which are preferred to solve this sort of problems. Our approach consists of searching to converge toward minimal error in an iterative way, relying on an optimization model.

V. THE PROPOSED APPROACH

As mentioned above, there is no model to establish directly a relationship between objective function and parameters, so direct numerical methods could not be applied to solve this problem. Following, we describe the procedure we employed to solve this problem applying a discrete PSO algorithm.

A. The PSO Model

The particle swarm optimization is based on the social behavior reproduction developed by R.Eberhart and J. Kennedy [14].

Particle swarm optimization is a population-based optimization algorithm modeled after the simulation of social behavior of birds in a flock. It is based on a set of individuals randomly arranged, called particles moving in the search space, each one is a potential solution and the aim is to get closer to the best solution.

A particle swarm is characterized by:

- The number of particles in the swarm.
- The topology and the neighborhood size of a particle that define his social network.
- The inertia weight of a particle, denoted w .

- The confidence factors, denoted by r_1 and r_2 , which weigh the tendency to return towards the best solution visited and the tendency to follow the neighborhood.

The performance of each particle, i.e. how close the particle is from the global optimum, is measured using an error function called also fitness function, which depends on the optimization problem.

Each particle has a memory about his best visited solution, as well as the ability to communicate with the particles forming his entourage. From this information, each particle keeps informed of its location and ability (the optimized function value), as well as the best place, and its corresponding ability, she has met so far in its flight.

Each particle i fly through an n -dimensional search space, and maintain the following information:

- x_i , the current position.
- $pbest_i$, the personal best position.
- v_i , the current velocity.

Building an m-uplet from predefined domains D_i forms the position of each particle. Then, the objective function evaluates the applied operators result with the selected parameters' values.

A general model of a particle swarm optimization algorithm is presented as:

```
Procedure PSO
Initialize a population of particles
do
  for each particle i with position  $x_i$  do
    if ( $x_i$  is better than  $pbest_i$ ) then
       $pbest_i \leftarrow x_i$ 
    end_if
  end_for
  Define  $gbest_i$  as the best position found so far
  by any of particles' neighbors
  for each particle do
     $v_i \leftarrow update\_velocity(x_i, pbest_i, gbest_i)$ 
     $x_i \leftarrow update\_position(x_i, v_i)$ 
  end_for
While (a stop criterion is not satisfied)
```

In Standard PSO, domains are continues and the velocity updates are calculated as a linear combination of position and velocity vectors. Thus, a particle velocity is updated using (10) and the position of this particle is updated using (11).

$$v_i(t+1) = w * v_i(t) + r_1 * (p_i(t) - x_i(t)) + r_2 * (gbest - x_i(t)) \quad (10)$$

$$x_i(t+1) = x_i(t) + v_i(t+1) \quad (11)$$

In the formula, w represents the inertia weight [14], $gbest$ is the best position among the best previous positions of particle informants. r_1, r_2 are numbers from a random distribution, and v_i must be in the range $[-V_{max}, V_{max}]$, where V_{max} is the maximum velocity.

This process is iterative and parallel, where in each iteration, all particles update their positions and must stop if there is a convergence, or after a fixed number of iterations.

B. Description of the proposed discrete PSO (DPSO)

In this case, parameter domains are discrete and some are non-digital. Obviously we cannot use directly the standard version of PSO designed for continuous domains. Several studies on particular applications, have discrete formulations of PSO (DSPO), beginning with BPSO (binary version) [26] to multivariate problems [27].

For discrete optimization problem, conventional PSO algorithm must address the following two issues:

- How to change the position of a particle?
- How to guarantee that positions are reasonable?

In DPSO, each particle represents its position in binary values, 0 or 1. Each particle's value can then be changed, or better say mutate, from one to zero or vice versa. In DPSO particle's velocity is defined as the probability that this particle changes its state to one [28].

The particles move in a state space restricted to 0 and 1, with a certain probability depending on individual and social factors. The probability of $x_i(t) = 1$, $Pr(x_i = 1)$, is a function of $x_i(t-1)$, $v_i(t-1)$, $pbest_i(t-1)$ and $gbest_i(t-1)$.

The probability of $x_i(t) = 0$ equals $1 - Pr(x_i = 1)$. Thus (10) is replaced by (12), where $rand3$ is a random number, $sig(v_i(t))$ is a logic transformation which can constrain $v_i(t)$ to the interval $[0,1]$ and can be considered as a probability:

$$sig(v_i(t)) = \frac{1}{1+e^{-v_i(t)}} \tag{12}$$

$$x_i(t) = \begin{cases} 1 & \text{if } rand3 < sig(v_i(t)) \\ 0 & \text{otherwise} \end{cases} \tag{13}$$

To extend the idea, another approach is proposed by M.Clerc [29] using the Traveling Salesman Problem (TSP) to illustrate the PSO concept for discrete optimization problems. M.Clerc defines a domain as a set of states with an order on objective functions values. He presents also some operations with position and velocity such as addition, subtraction and multiplication specific to TSP. A distance is defined to be utilized with physical neighborhoods. The idea is interesting, but the problem's definition and update functions depend on the application.

Inspired by M. Clerc idea regarding the states' set, we propose a discrete PSO model based on following definitions:

- $D=D_1*D_2*...*D_M$ is considered as a set of states (or a search space).

Each M-uplet $x = (u_1, u_2, \dots, u_M)$ is a state.

- Objective function Err is discrete and numerical with an order on states:

$$Err(x) \geq Err(x') \text{ or } Err(x') \geq Err(x)$$

- Position of a Particle is a state

- Domain representation: each value in the domain can be represented by its relative location. These locations are fixed in some conventions.

The model is illustrated in ‘‘Fig. 6’’.

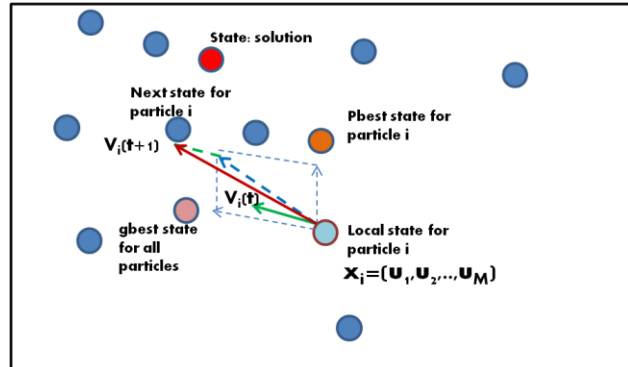


Fig. 6. Space State Presentation

A domain representation can be expressed in many ways to facilitate operator's definition on states. One of these representations is given here:

A domain $D_j = \{u_{1j}, \dots, u_{kj}, \dots, u_{nj}\}$ is ordered from 1 to n, $n=|D_j|$.

Each value has a location in the domain and then two reciprocal functions Rank and Value are defined:

$$Rank(u_{kj})=k, \text{ the rank of a value } u_{kj} \text{ in } D_j, (k=1, \dots, n)$$

$$u_{kj} = Value(k), \text{ the value in } D_j \text{ of rank } k.$$

Example:

$$D_j = \{a, b, c, d\}$$

$$Rank(a)=1, \dots, Rank(d)=4$$

$$Value(1)=a, \dots, Value(4)=d$$

The velocity can be expressed as a distance between ranks of two separated states.

C. The update functions

Since the DPSO problem claims to define positions discretely and updates them using velocity, we describe below the new proposition.

a) Add/subtract operators: To add two values at rank k and k' we simply move to the rank $k+k'$ and the result is the value corresponding to rank $k+k'$. Since D_j has a limited size $|D_j|$, $k+k'$ is calculated modulo $|D_j|$. Then the addition operator can be defined as:

$$u_{kj} \oplus u_{k'j} = u_{(k+k')j}$$

Idem for Subtraction, the result correspond to $k-k'$ modulo $|D_j|$:

$$u_{kj} \ominus u_{k'j} = u_{(k-k')j}$$

The position of a particle is a state $x = (u_1, u_2, \dots, u_M)$, were u_j represent values in D_j . To change the state we extend operations:

$$x \oplus x' = (u_1 \oplus u'_1, u_2 \oplus u'_2, \dots, u_M \oplus u'_M),$$

$$x \ominus x' = (u_1 \ominus u'_1, u_2 \ominus u'_2, \dots, u_M \ominus u'_M),$$

$$\text{Rank}(x) = (\text{Rank}(u_1), \dots, \text{Rank}(u_M)),$$

$x = (\text{Value}(k_1), \dots, \text{Value}(k_M))$, where k_j represent ranks of values in D_j .

b) *Velocity update*: We use the same expression in (10) with the new add/subtract operators and the velocity is expressed as a vector of moves (left: - and right: +)

$$v(t+1) = \text{round}(w * v(t) + r_1 * (\text{Rank}(p(t)) - \text{Rank}(x(t))) + r_2 * (\text{Rank}(g_{\text{best}}) - \text{Rank}(x(t)))) \text{ modulo } (|D_1|, \dots, |D_M|) \quad (14)$$

c) *Position update*: To update the value of a parameter, we update at first its rank:

$$\text{Rank}(x(t+1)) = \text{Rank}(x(t)) + v(t+1) \text{ modulo } (|D_1|, \dots, |D_M|) \quad (15)$$

Then, we obtain the correspondent value:

$$x(t+1) = \text{Value}(\text{Rank}(x(t+1))) \quad (16)$$

VI. APPLICATION

In the previous section the novel DPSO approach is described, the problem of choosing optimal operators and the optimal values of their parameters in a vision task is solved. The approach we presented in theory is applicable to any vision task that needs operators' selection or parameters' adjustment or both of them. In this section our approach is tested on two different tasks of image processing, first one is about contours detection of mechanical objects and second one is about text recognition and aspect inspection related to tickets label on industrial products.

A. Case study 1: Contour detection of mechanical objects.

a) Phases, operators and parameters determination

The task of contours detection allows identifying areas, of a digital image, corresponding to a brutal change in light intensity. It significantly reduces data quantity and eliminates the information judged less relevant, while preserving the important structural properties of the image, in order to extract measurements for example. This task is made of three phases of treatment "Fig. 7", firstly a pre-processing phase is necessary to remove any noise from the image, and then processing phase would take place to determine object contours, a post-processing phase will go after to eliminate insignificant contours. A comparison between a segmentation done by experts in image processing domain, and results obtained by DPSO approach will be done.

- Pre-processing phase

Preprocessing phase consists of improving image quality using filters; we propose a list of three filters predefined in Matlab: medfilt2, ordfilt2 and wiener2. Medfilt2 is a nonlinear operator called median filtering, used in image processing to reduce "salt and pepper" noise, ordfilt2 is also a nonlinear operator it is an order-statistics filtering, it replaces each element in the image by the orderth element in the sorted set of neighbors specified by the non zero elements in domain. Finally wiener2 is an adaptive noise-removal filtering; it uses a pixel wise adaptive Wiener method based on statistics estimated from a local neighborhood of each pixel.

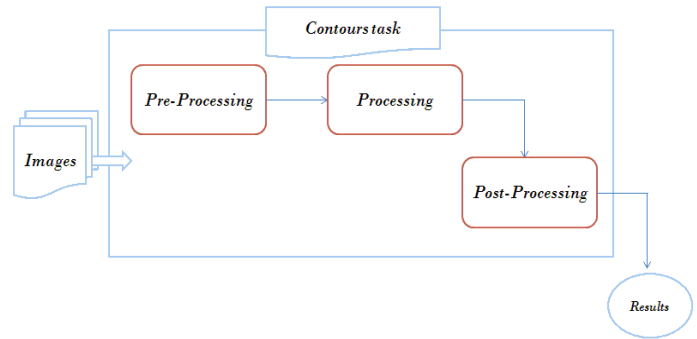


Fig. 7. Treatment phases of a contour task

To apply these filters a parameter must be specified: the filter size called t_w , a range of possible values would be provided "Fig. 8".

- Processing phase

Processing phase consists of detecting edges (contours), the operator applied is predefined in Matlab: edge¹ with two parameters to be adjusted, filter to use and the threshold. A range of possible values would be provided for each parameter "Fig. 8".

- Post-processing phase

Post-processing phase consists of refining the image by deleting small objects. The operator 'bwareaopen' would be applied as one operator of this phase; it is a morphological operator, which removes from a binary image all objects that have connectivity inferior than a predefined threshold. A range of possible thresholds and connectivity values would be provided "Fig. 8".

- Objective function

The objective function, called also error function, depends on the optimization problem. In this case, contours detection task, many adapted error functions are possible. The error of a particle should indicate how good the segmentation of the input image is, in comparison to the target segmentation.

The error function used here is based on the confusion matrix for a two-class classifier. Several standard terms have been defined for the two-class matrix; the one used in this work is the accuracy, which represents the proportion of the total number of predictions that were correct. It is determined using the equation:

$$Acc = \frac{t_p + t_n}{t_p + t_n + f_p + f_n} \quad (17)$$

Where t_p (true positive) represent white pixels well ranked (contours), t_n (true negative) represent dark pixels well ranked, f_p (false positive) represent contours misclassified and f_n (false negative) represent dark pixels misclassified.

¹ <http://www.mathworks.com/help/images/ref/edge.html>

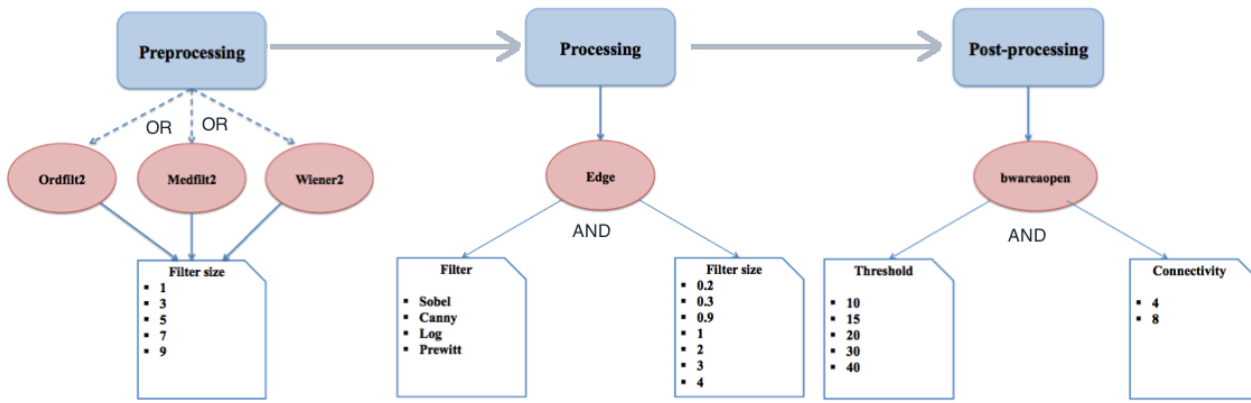


Fig. 8. Operators' and their parameters possible values for each phase

In reality, contour pixels found are compared to those in the ground truth image using DPSO approach. The error function considered is:

$$Err = 1 - Acc \tag{18}$$

Applying DPSO approach associates to each operator's combination an error value, in addition to its best parameter values.

$$C_1(O_1(p_2), \dots, O_n(p_k)) \text{ -----} \Rightarrow Err_1$$

$$C_2(O_1(p_3), \dots, O_n(p_j)) \text{ -----} \Rightarrow Err_2$$

⋮
⋮

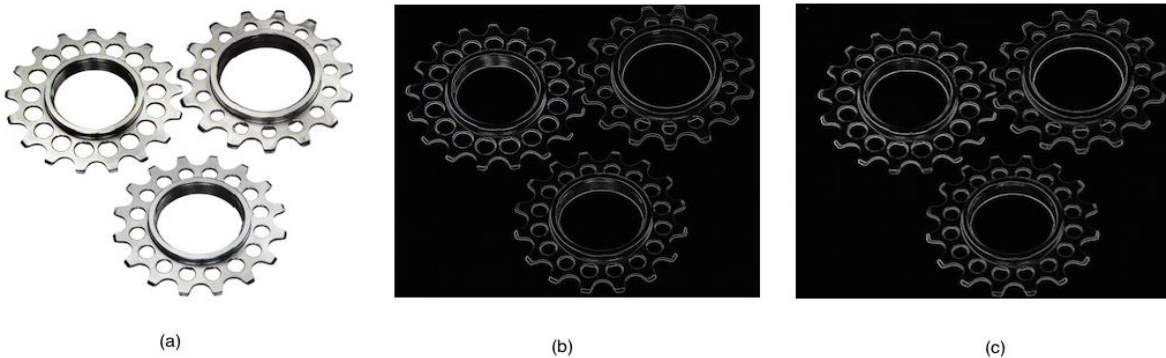
$$C_n(O_1(p_1), \dots, O_n(p_m)) \text{ -----} \Rightarrow Err_n$$

The best combination is as: $\min(Err_1, Err_2, \dots, Err_n)$.

b) Results and Discussion: The experiment was conducted on a dataset of mechanical objects images. DPSO model is applied and table 1 resumes best parameter's values and error rates of best operator's combination, and a simple of images' result is shown in "Fig.9".

TABLE I. BEST OPERATORS COMBINATION AND THEIR BEST PARAMETERS' VALUES FOR TEST IMAGES

		Pre-processing	Processing		Post-processing		Error rate
Image 1	Operator	Medfilt2	Edge		Bwareaopen		0.0023
	Parameters	1	Sobel	2	30	8	
Image 2	Operator	Medfilt2	Edge		Bwareaopen		0.0041
	Parameters	3	Prewitt	3	10	8	
Image 3	Operator	Ordfilt2	Edge		Bwareaopen		0.0012
	Parameters	2	Sobel	0.9	15	8	



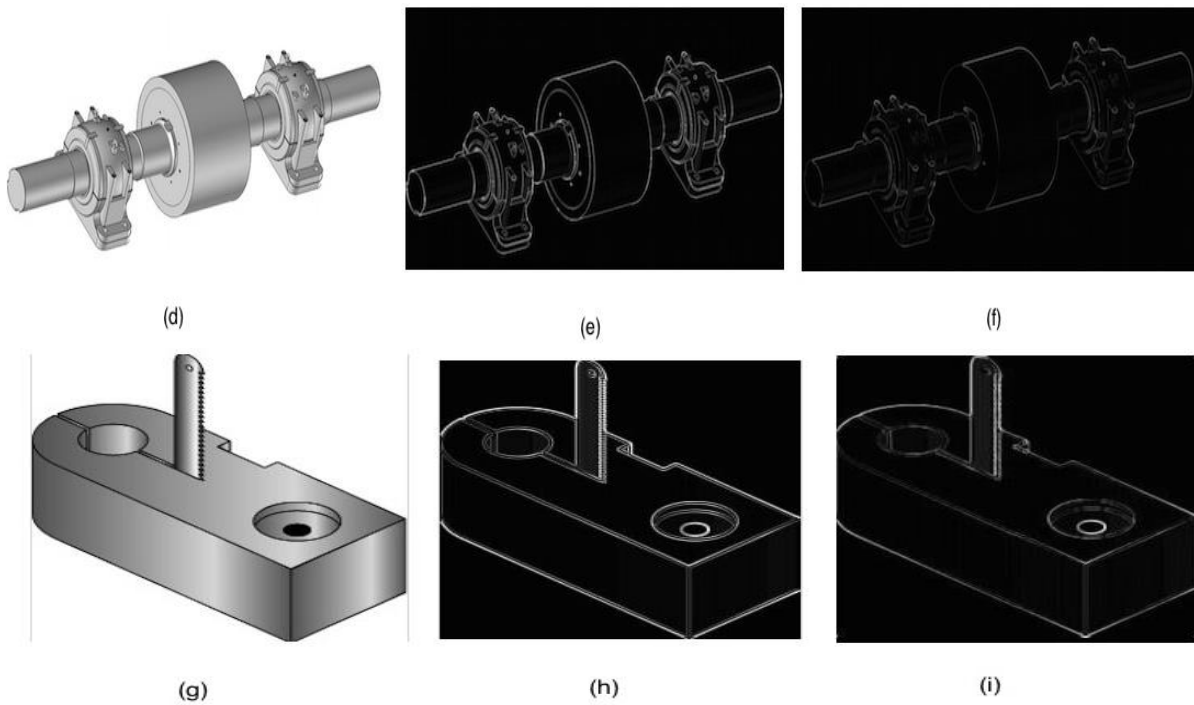


Fig. 9. From left to right: image of mechanical object, segmentation done by an expert and finally DPSO approach result

The proposed method achieves good results with a small error rate, but to improve even more the quality of results we will try in the following to adjust parameters of DPSO algorithm.

B. Case study 2: Text recognition

Let's consider a line producing bottles, a vision system will be configured, in order to control the stickers presence on the bottles, verify its position, the presence of the logo and the trade name.

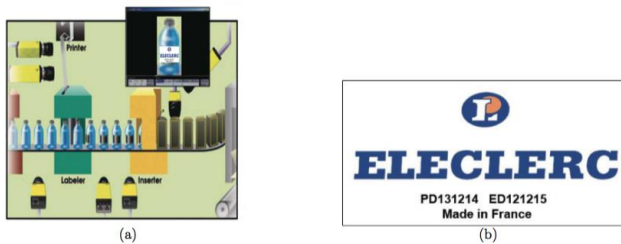


Fig. 10. Production line of bottles using a vision system for quality control. (b) The sticker image

The process of aspect inspection consists of detecting the ticket presence and making sure that it contains the correct logo and content. ‘Fig. 11’ shows different phases performed to achieve this vision task and illustrate operators used in each phase.

Phase 1: Determine regions	Phase 2: Intersection with edges	Phase 3: Filtering	Phase 4: Character recognition
•MSERFeatures	•Edge	•Vertcat •Stroke	• Regionprops •OCR

Fig. 11. Different phases of the vision task and operators candidates for each phase

Each one of these operators, except OCR, has some parameters to be fixed depending on the image we are working on. To find out the parameter values optimal combination, a range of values for each one of parameters is given up; a list of possible values of each parameter is provided in table 2.

TABLE II. VALUES CANDIDATES FOR EACH PARAMETER

Operators	Mserfeatures		Edge		Vertcat		Strock	RegionProps
	Max	Min	Threshold	Filter	Eccentricity	Solidity	Width variation	Areathreshold
Values candidates for each parameter	10	2000	0,001	Canny Log Prewitt Sobel	0,970	0,1	0,3	10
	20	3000	0,002		0,975	0,2	0,4	20
	30	4000	0,003		0,980	0,3	0,5	30
	40	5000	0,004		0,985	0,4	0,6	40
	50	6000	0,005		0,990	0,5	0,7	50
	60	7000	0,006		0,995	0,6	0,8	60
						0,7	0,9	70

Fixing parameters for an optimization algorithm is a very important step before proceeding to parameter optimization of a vision task. DPSO is applied using 30 particles; inertia weight is fixed to 0.5, Confidence factor r_1 to 0.3 and Confidence factor r_2 to 0.7. We run the algorithm for 30 iterations, for the only operator's combination of this task:

[MserFeatures, Edge, Vartcat, Stroock, Regionprops, OCR]

The objective function used to evaluate this task is composed of three steps, since we have to found three different regions: the first one must contain the logo, and it is compared to its reference using corr2, a predefined function in Matlab returning the correlation coefficient between two images. Second and third ones must contain a text, which we compare to references and penalize each unrecognized letter. The optimal parameter values found are summarized in table 3, and “Fig. 12” demonstrates experimental results of each

operator, using optimal parameters. Three different regions are detected. First one, which contains the logo, is compared to its reference. The second and third regions contains a text, which we recognize using OCR and compare to references. The final error rate represents the sum of the three error rates calculated over the three text regions found.

TABLE III. OPTIMAL PARAMETERS VALUES

Parameter	Optimal Value
Area-Range	[10, 6000]
Filter	Canny
Threshold	0,04
Eccentricity	0,995
Solidity	0,2
Width-variation	0,9
Area-threshold	10

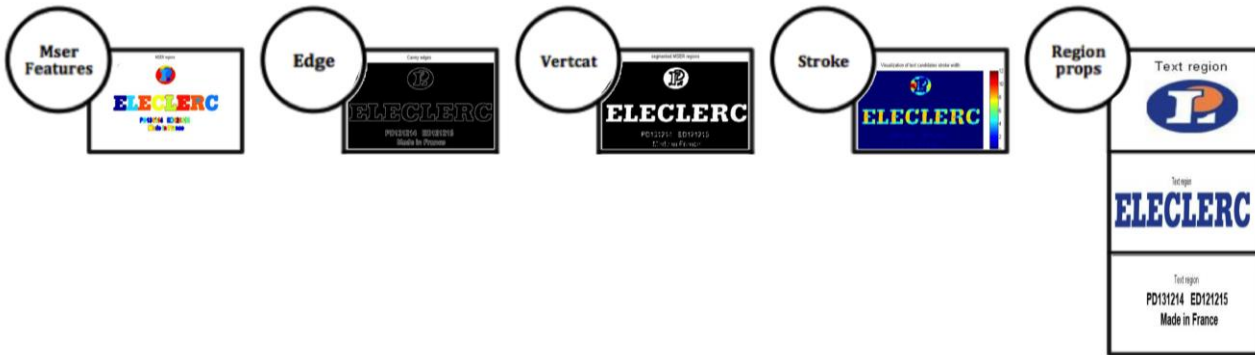


Fig. 12. Results of each operator application

The examples treated here validate practically our approach for images in quality control domain. This approach constitutes a new general and expressway of reasoning for any vision task that requires the right choice of operators and the right adjustment of their parameters, since it is based on easy mathematic operations.

C. Adjustment of DPSO parameters and discussion

Adjusting DPSO parameters is a very important step since they can have a large impact on optimization performance. In order to set proper parameters for the proposed DPSO approach some experiences will be carry on to adjust these parameters in order to see their influence on results quality [30]. Based on the examples bellow, the particle’s number and inertia weight influence on the error value has been studied, by varying them separately. The evolutions are shown in figures bellow.

- Number of particles

Selecting the proper number of particles is a critical step because it affects the algorithm performance. Number of particles needs to be sufficient to explore all possible states with the least possible iterations. In most cases, increasing the number of particles decreases the number of required algorithm iterations [31]. So the performance of the DPSO algorithm is tested with 5, 10... up to 100 particles. The number of particles effect is shown in “fig. 13”. It is observed that the DPSO model offers the best results when the

number of particles is bigger than 30 in case study 1, while in case study 2 DPSO model offers the best results when the number of particles is 30, 40, 50, 70 and 80.

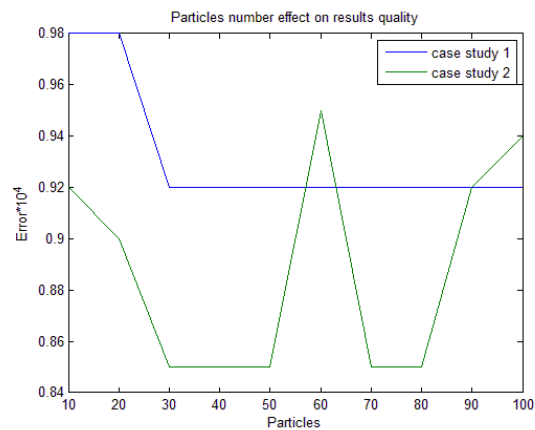


Fig. 13. Effect of particles number on performance of DPSO model

- Inertia weight

This term serves as a memory of previous velocities; it was first introduced by Shi and Eberhart [32]. To control the impact of the previous velocity: a large inertia weight favors exploration, while a small inertia weight favors exploitation [33]. The effect of inertia weight variation is shown in “Fig. 14”. It is observed that the DPSO model offers the best result

when the inertia weight is bigger than 0.5 in case study 1, while it starts from 2 in case study 2.

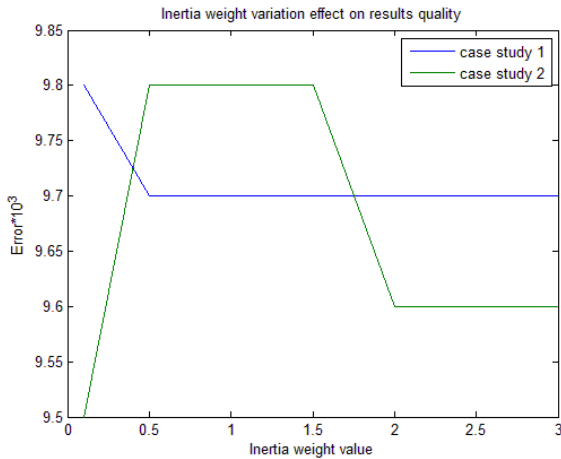


Fig. 14. Effect of inertia weight on performance of DPSO model

- Selected parameters

TABLE V. BEST OPERATORS COMBINATION AND THEIR PARAMETERS' VALUES FOR CASE STUDY 1 AFTER DPSO TUNING

Image	Operator	Pre-processing	Processing		Post-processing		Error rate
		Parameters	Medfilt2	Edge		Bewareaopen	
Image 1	Operator	Medfilt2	Edge		Bewareaopen		0.0022
	Parameters	1	Sobel	2	30	8	
Image 2	Operator	Medfilt2	Edge		Bewareaopen		0.0040
	Parameters	3	Prewitt	3	15	8	
Image 3	Operator	Ordfilt2	Edge		Bewareaopen		0.0012
	Parameters	2	Sobel	0.8	15	8	
			Width-variation			0.9	
			Area-threshold			10	

TABLE VI. BEST PARAMETERS' VALUES FOR CASE STUDY 2 AFTER DPSO TUNING

Parameter	Optimal Value
Area-Range	[10, 6000]
Filter	Canny
Threshold	0.06
Eccentricity	0.995
Solidity	0.1

In regards to the parameters of DPSO, Shi and Eberhart [33], Jordehi and Jasni [34] and Pedersen [35][36] studied parameters selection on particular problems. On the basis of the above parameter analysis results and literature, Table 2 provides the detailed setting for the proposed DPSO model. It may be noted here that, the error evolution is not stationary according to figures above. We can simply locate the minimal error interval and choose the corresponding values. Effectively, there is a slight error decrease regarding the values used in Sec. 5.1.2 and Sec 5.2 (first trial) and a slight modification of the optimal values of operators' parameters but not too significant.

TABLE IV. SETTING OF THE PROPOSED DPSO MODEL

	Values for case study 1	Values for case study 2
Particles number	30	30
Inertia weight	0.5	2.5

Tables 5 and 6 resumes new best operators' combinations and their optimal parameters after DPSO tuning.

To go further, we use a dataset of mechanical images and try to detect and recognize text on these images, using the same experimental setup described above. "Fig.15" shows the error rates obtained by the proposed DPSO approach with contrast to some other techniques presented in [37] [38] and [39].

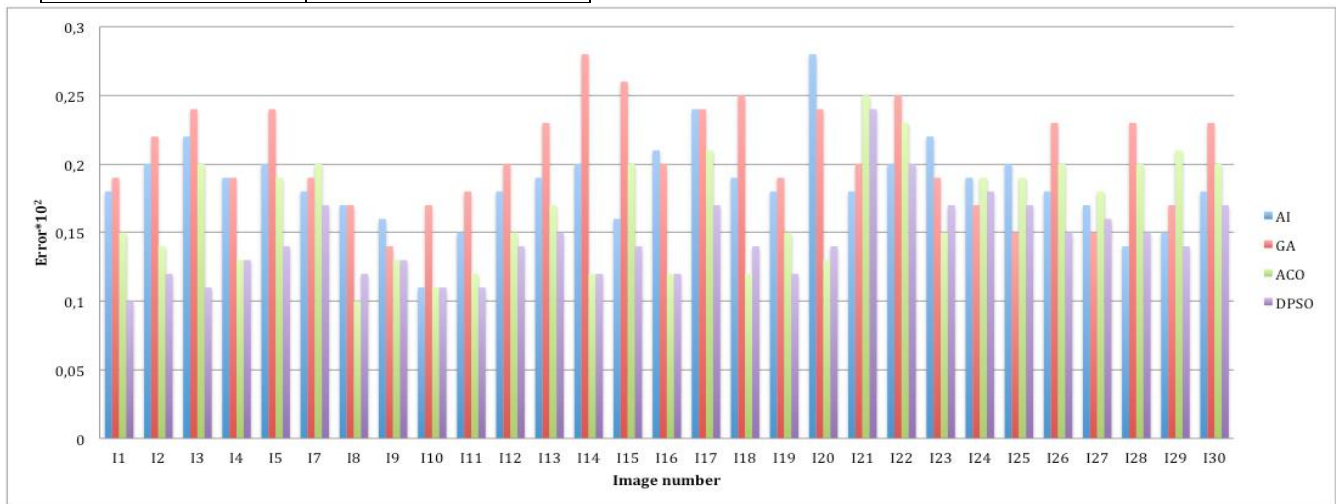


Fig. 15. Bar plot of test error with different methods. DPSO stands for Discrete Particle Swarm intelligence method proposed in this research, ACO denotes Ant Colony Optimization, GA is Genetic Algorithm, AI means artificial intelligence

Experimental results of optimal parameters specifically error rate, obtained by the proposed DPSO algorithm is compared with the ant colony optimization approach, genetic algorithm approach and artificial intelligence approach. The proposed DPSO approach obtains best results on the majority of test images, while ACO approach gives very close error values to first approach.

VII. CONCLUSION

Choosing the appropriate operators to apply and then adjusting their parameters values to accomplish a vision task is a very big challenge for users, in this work an automated method to optimize parameters values of image processing algorithms in quality control is presented, this system proceeds automatically to decide which operator is the most appropriate to use, and adjust automatically values of its free parameters. Novel update functions for the new DPSO definition are suggested. This new system is intended to have a better precision.

In practice a set of parameters is supplied, to which a range of values is provided, with the help of DPSO the approach is applied on a specimen of test, which demonstrates the performance of the proposed procedure. A comparison to others methods would be held to validate and support the satisfying results of DPSO.

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Improving Image Encryption Using 3D Cat Map and Turing Machine

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Abstract—Security of data is of prime importance. Security is a very complex and vast topic. One of the common ways to protect this digital data from unauthorized eavesdropping is encryption. This paper introduces an improved image encryption technique based on a chaotic 3D cat map and Turing machine in the form of dynamic random growth technique. The algorithm consists of two main sections: The first does a preprocessing operation to shuffle the image using 3D chaotic map in the form of dynamic random growth technique. The second uses Turing machine simultaneous with shuffling pixels' locations to diffuse pixels' values using a random key that is generated by chaotic 3D cat map. The hybrid compound of a 3D chaotic system and Turing machine strengthen the encryption performance and enlarge the key space required to resist the brute force attacks. The main advantages of such a secure technique are the simplicity and efficiency. These good cryptographic properties prove that it is secure enough to use in image transmission systems.

Keywords—chaotic 3D cat map; brute force attacks; Dynamic random growth technique; Turing machine; key space

I. INTRODUCTION

Up-to-date development and progress in the means of multimedia industry and the ways of communications have made studies focus on creating new schemes to enhance the security of transmission and storing multimedia data over open channels including the Internet and wireless networks. In mean times, multimedia data were transmitted over computer networks can be audio, image, and other multimedia types. During that, the images can be considered one of the most usable forms of information. These images often contain private or confidential information and sometimes they associated with financial interests [1, 2]. Security of digital image has become more and more important because of the advances in communication technology and multimedia technology. Three different ways to protect these digital images from unauthorized eavesdropping are cryptography, steganography and watermarking. Among these three techniques, cryptography has become one of the major tools to provide a high level of security. Image encryptions have applications in various fields including Internet communication, multimedia systems, medical imaging, telemedicine and military communication. As a result, the security for images, it needs to be hidden from unauthorized access to decode by transferring the multimedia data into an entirely different format, protected from unauthorized change,

and available to an authorized entity, when it is needed. Although the three previously mentioned requirements have not changed, they now have some new dimensions. Not only should images be confidential, when it is stored in the computer; there should also be a way to maintain its confidentiality when it is transmitted from one computer to another. "Fig. 1" presents a diagrammatic view of general encryption-decryption mechanism.

The primary essence of this paper would be the protection of images. The high efficiency of any cryptographic algorithm is the most important criterion by which the robustness of encryption is measured. Traditional image encryption algorithms, for instance, private key encryption standards (DES and AES) faces problems when used to encrypt large images and therefore, its efficiency becomes low and weak, public key standards such as Rivest Shamir Adleman (RSA), and the family of elliptic-curve-based Encryption (ECC), as well as the international data encryption algorithm (IDEA) requires a great computational time and super computers when used in encrypting real-time images, may not be the most desirable candidates for image encryption, especially for fast and real-time communication applications because of Cryptographic algorithms that use less time are much more preferable for encrypting such real-time images. Also, some encryption schemes may be run very slowly, and this increases the degree of security features, yet they would be of little use when dealing with real-time images [3].

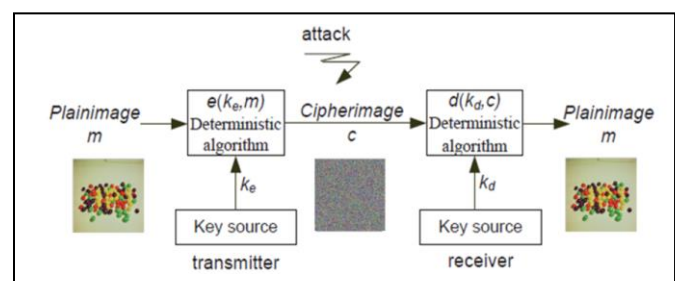


Fig. 1. General Encryption-Decryption Mechanism

In the last decade, chaos-based encryption techniques are considered suitable for practical applications since they have a perfect combination of speed, high security, complexity, reasonable computational overheads, and computational power. Moreover, chaos-based and other dynamical systems based algorithms have many important properties such as the

sensitive dependence on initial conditions and system parameters, pseudorandom properties, ergodicity, and non-periodicity [4].

Some chaos image encryption schemes have developed in recent years. In [5] proposed a new image encryption algorithm based on 3D chaotic map; this algorithm does not use the whole chaotic numbers, but only utilizes partial elements as running key. In chaotic sequence, those used elements keep certain interval, which is determined by the current ciphertext. Due to ciphertext closely related to plaintext, running-key is indirectly related with plaintext. In [6], shuffles the RGB layers by using 3D Cat map and standard map, and finally, the image is encrypted by performing XOR operation on the shuffled image and diffusion template. In [7], suggested a novel image encryption algorithm based on a three dimensional (3D) chaotic map; the design of the proposed algorithm based on three phases. In phase I, the image pixels are shuffled according to a search rule based on the 3D chaotic map. In phases II and III, 3D chaotic maps are used to scramble shuffled pixels through mixing and masking rules, respectively.

In this paper, an encryption algorithm for digital images which based on chaotic 3D cat map and the Turing machine is proposed. As cat map is weakness and has some common drawbacks that are: 1) periodic, and 2) can be easily cracked. By chosen plaintext attack, we use cat map in another more secure way, which has benefits of mixing and sensitivity to initial conditions and parameters, to overcome these former defects; through calculating an intermediate parameter based on the blue layer of the image. The intermediate parameter is used as the initial parameter of a chaotic map to generate a random key stream. Consequently, the generated key streams are dependent on the plaintext image, which can resist the chosen plaintext attack [8]. The 3D chaotic map with random choose of initial conditions and parameters are used to generate three discrete chaotic sequences with high sensitivity by iterations. Two of the sequences are then used to design a two-dimensional permutation by calculating new coordinates of each pixel of the image and shuffle it. Then elements of the third one affect the behavior of the other two sequences through effect the initial values of the chaotic system for the further encryption process, which increases the sensitivity of plaintext images of the scheme. Both theoretical and computer simulations show that the algorithm is secure enough to use in image transmission system.

Organize other sections of this paper as follows: Section 2 introduces a chaotic 3D cat map; Section 3 discusses Turing machine; Section 4 shows the proposed image encryption and decryption schemes; Sections 5 and 6 present the experiment results and security analysis; Section 7 presents time complexity; finally, Section 8 concludes this paper.

II. THE CHAOTIC 3D CAT MAP

Arnold's cat map is a discrete-time dynamical system. Specifically, it is a kind of cut-out transformation, is originally introduced into traverse theory by Arnold and so-called Arnold transformation [9]. Arnold cat transformation is a classical encryption algorithm. Mathematically, Arnold's cat map is given by "(1)":

$$\begin{pmatrix} x_{n+1} \\ y_{n+1} \end{pmatrix} = \begin{pmatrix} 1 & 1 \\ 1 & 2 \end{pmatrix} \begin{pmatrix} x_n \\ y_n \end{pmatrix} \text{mod } 1 \quad (1)$$

In the process of image encryption, the use of Arnolds Cat Map provides extra security, as it approaching higher randomness. The above 2D cat map is now generalized by producing two new parameters, p and q into "(1)", which increase key spaces to resist brute-force attacks and longer averaged period lengths improve map randomness [10] to resist statistical attacks, as follows "(2)":

$$\begin{pmatrix} x_{n+1} \\ y_{n+1} \end{pmatrix} = \begin{pmatrix} 1 & 1 \\ 1 & 2 \end{pmatrix} \begin{pmatrix} x_n \\ y_n \end{pmatrix} \text{mod } N \quad (2)$$

To increase the security of the Arnold's cat map, the generalized 2D cat map is extended to a 3D discrete chaotic map by introducing six control parameters; $a_x, a_y, a_z, b_x, b_y, b_z$, a three-dimensional cat map can be presented as the following formula "(3)":

$$\begin{pmatrix} x_{n+1} \\ y_{n+1} \\ z_{n+1} \end{pmatrix} = A \begin{pmatrix} x_n \\ y_n \\ z_n \end{pmatrix} \text{mod } 1 \quad (3)$$

where

$$A = \begin{pmatrix} 1+a_x a_z a_y & a_z & a_y + a_x a_z + a_x a_y a_z b_y \\ b_z + a_x b_y + a_x a_z b_y b_z & a_z b_z + 1 & a_y a_z + a_x a_y a_z b_y b_z + a_x a_z b_z + a_x a_y b_y + a_x \\ a_x b_x b_y + 1 & b_x & a_x a_y b_x b_y + a_x b_x + a_y b_y + 1 \end{pmatrix}$$

One can be easily verified that $\det A = 1$, which means the 3D discrete cat map is a 1-1 map and the sensitivity to initial conditions and parameters are kept unchanged. By simply setting $a_x = b_x = 1, a_y = b_y = 2, a_z = b_z = 3$, one gets

$$A = \begin{pmatrix} 7 & 3 & 17 \\ 23 & 10 & 56 \\ 4 & 1 & 10 \end{pmatrix}$$

Through numerical calculations, three eigenvalues of the above A are $\sigma_1 = 25.1314, \sigma_2 = 0.0215, \text{ and } \sigma_3 = 1.8470$. It verified that the leading Lyapunov exponent is strictly positive since and given by $\ln \sigma_1$. Consequently, the corresponding 3D cat map is higher chaotic with properties of mixing and sensitivity to initial conditions and parameters of it.

III. TURING MACHINE

A Turing machine deals with a sequential tape having only three symbols: 0, 1, *. In this example the machine will take a binary number delimited by two *, and increment it by 1.

For example, if it reads *1111 0110* it should give *1111 0111*. If it reads *0001 0011* it should give *0001 0100*. If it reads *1111 1111* it should give *0000 0000

*. And so on. “Fig. 2” presents the flow diagram of the encryption algorithm.

There are four ways to represent Turing Machine which are be from either right-to-left, or right-to-left inverse, or left-to-right, or left-to-right inverse.

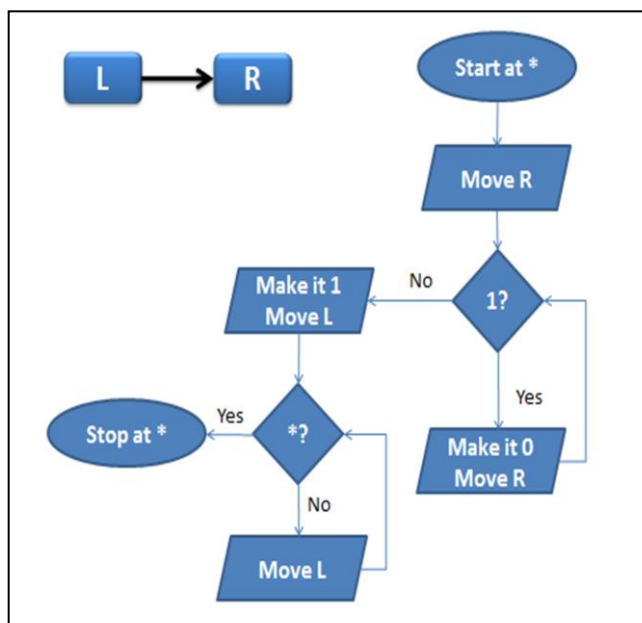


Fig. 2. Flow Diagram of Process of Left-to-Right Turing Machine

IV. DETAILS OF THE PROPOSED METHOD

A. The proposed encryption scheme

In the proposed method, three pseudorandom discrete variables of 3D chaotic cat map are adopted to permute color image $O(i, j)$ of size $I \times J$ using the formula of “(3)”. The presented method is a combination of two essential operations, permutation and substitution. “Fig. 3” gives the flow diagram of the encryption algorithm. The procedure of our encryption scheme is described as follow:

1) For 3D cat map “(3)”, as shown in [10], choose at random three initial value (x_0, y_0, z_0) and six control parameters $(a_x, a_y, a_z, b_x, b_y, b_z)$, which are served as secret keys.

2) Before, presenting these intrinsic processes, a preprocess operation is performed which use three initial values (x_0, y_0, z_0) to iterate the 3D chaotic map n_c times to avoid the harmful effect of the initial values, where n_c is a present integer and served as a secret key, too. Assign the last three outputs (x_0, y_0, z_0) from “(3)”, to be new initial values of a 3D discrete cat map.

3) Permutation Operation: In this procedure, a 3D cat map is still used. This process can be achieved through the following steps:

a) The plain image is divided into three basic layers (red, green and blue), each of them divided into four sub-blocks which are equal in size, then rotate the first three sub-images 90 degree where the 1st sub-image is rotated 90

degree twice times, the 2nd sub-image is rotated 90 degree triple times, and the 3rd sub-image is rotated 90 degree only once time.

b) We apply a 3D discrete chaotic cat map on red and green layers for each sub-image, while blue layer for all blocks is still unchanged.

c) While we iterate a chaotic 3D cat map for n_c times; where $n_c = 1, 2, \dots, 204$, we get three pseudorandom chaotic key stream values (x_1, y_1, z_1) , where x_1 and y_1 are used to calculate a new position (s, d) corresponding to the present pixel $O(i, j)$ using output of “(3)”.

d) The chaotic sequence S_z is produced using the pseudorandom chaotic key stream value z_1 which is generated by “(3)”. In order to avoidance fleeting effects, neglect the first 200 values resulting from iterating 3D cat map.

e) Let [8]:

$$low_j = \text{floor}(x_{200+j} \times 10^{14}) \bmod \frac{N}{8} + \frac{N}{2} + (j-1) \frac{N}{8}; j = 1, 2, 3, 4. \quad (4)$$

f) Calculate sum for the original blue layer of each block; since S_B^{ij} represents the pixels’ values of blue layer B at position (i, j) to affect the resulted values of a 3D separate chaotic cat map.

g) Perform the confusion procedure to the lower part of the size $low_i \times low_j$ of each of the four sub-images of the plain-image using 3D chaotic map respectively. “This is the so called [8] dynamic random growth technique, since the size of the lower part of each block of the plaintext image permuted by 3D chaotic map is random.”

h) Operate confusion operation to the whole each sub-block by 3D chaotic map.

4) Substitution Operation: In this procedure, the pixels’ values are altered by performing Turing machine on each block of the four sub-images. This process can be achieved through the following steps:

a) While confusing pixels’ values in permutation process, performing Turing machine (TM) either from left-to-right or vice versa.

b) We convert every pixel value to a binary number B of 8 bit lengths.

c) Turing machine works as if pixel value equals 1 converts it to 0 until reached to the 1st pixel value equals 0, then converts it to 1 and the rest of bits still un-changed, as introduced in “Fig. 2”.

$$O(m, n) = TM(O(s, d)) \quad (5)$$

$$O(s, d) = TM(O(m, n)) \quad (6)$$

d) After applying Turing machine for red and green layers of each sub-image, we get R' and G' .

e) To enhance the security of encryption process, apply a bit exclusive OR, using the following formula:

$$G' = R' \oplus G' \quad (7)$$

$$B' = G' \oplus B \quad (8)$$

f) After applying Permutation- Substitution Operation on all sub-blocks in the three layers we combine these sub-blocks into one block for each layer then combine this layer to construct cipher image.

B. The proposed decryption scheme

The decryption process is the inverse process of the corresponding encryption scheme. "Fig. 4" gives the flow diagram of the decryption algorithm.

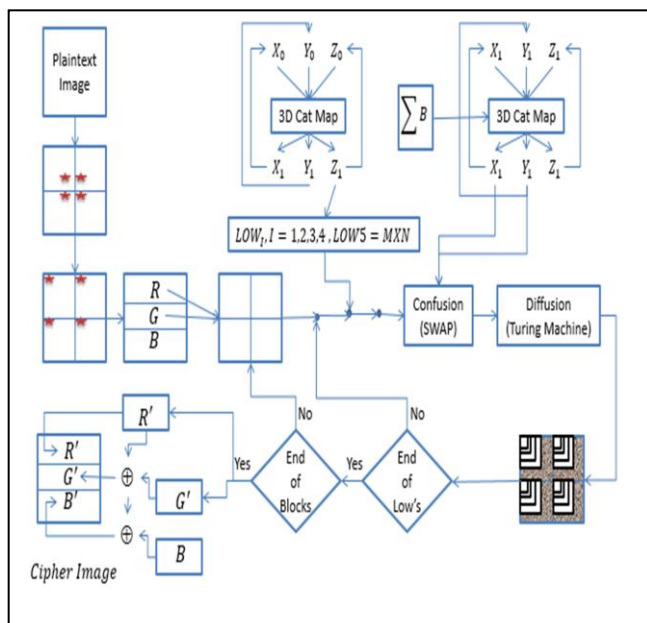


Fig. 3. Flow Diagram of Process of the Introduced Encryption Scheme

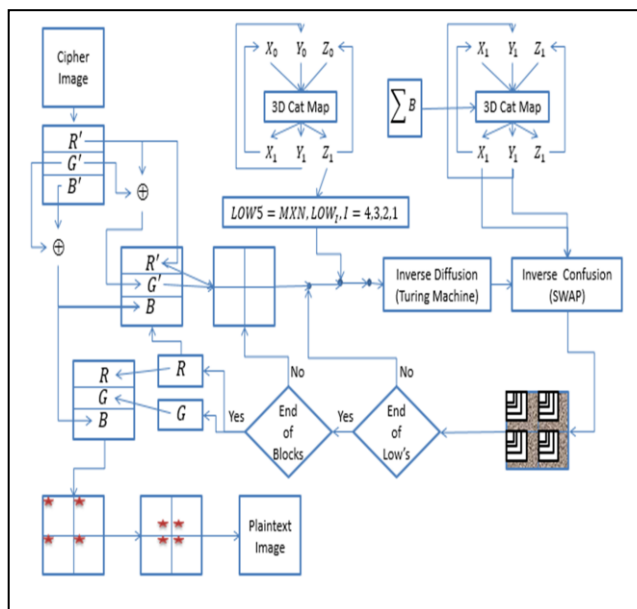


Fig. 4. Flow Diagram of Process of the Introduced Decryption Scheme

V. EXPERIMENTAL RESULTS

We use MATLABR2013a to simulate our proposed scheme on 2.30 GHz CPU, with 4.00 GB memory running on Windows 8.1. We select five standard color images as the plain images which are Lena.bmp, Splash.bmp, Jet.bmp, and Girl.bmp of size 512×512 and Lena.bmp of size 500×400 .

For a chaotic 3D cat map, the initial values $x_0 = 0.3, y_0 = 0.4, \text{ and } z_0 = 0.5$ and six control parameters $a_x = b_x = 1, a_y = b_y = 2, \text{ and } a_z = b_z = 3$, are served as keys, after iterating the map n_c times, where n_c is a present integer and served as secret key, too.

"Fig. 5" and "Fig. 6" show the results both of original and cipher images, Splash and Jet standard color test images, respectively. In the proposed encryption scheme, one round of the scheme indicates that the cipher reaches a high secure level.

VI. SECURITY ANALYSES

Security and performance are the crucial requirements for any encryption system. Different from the encryption of textual information, the encryption of visual information requires security against cryptographic attacks such as differential analysis, related-key attack, and statistical attack. Generally, an encryption scheme should be thoroughly analyzed before it can be used in practical applications. Indeed, several metrics can be used to measure the cipher's resistance to some typical attacks, for example, sensitivity analysis, statistical analysis, numeric analysis, etc.

Generally, if the encryption scheme is secure against most of the attacks, we can say that the encryption scheme is of high level of security. Otherwise, the encryption scheme is regarded as of low security. The detailed analyses are investigated in this section as follows.

A. Key space analysis

For an effective cryptosystem, the key space should be sufficiently large enough to prevent brute-force attack. "It is known that [11] the ideal key space should be larger than 2^{100} while considering the current computer computation speed. The total key space of our presented algorithm composed of three parts which are summarized as follow: ① the initial values (x_0, y_0, z_0) ; ② six parameters $(a_x, a_y, a_z, b_x, b_y, b_z)$; ③ preprocess parameter n_c of a chaotic 3D cat map, and then it is around $\approx 2^{392}$ which is much larger than in the algorithms of G. Gu, J. Ling [10] and X. Wang et al. [8]; so it is obvious that the presented technique has a large enough key space to resist common brute- force attacks. The key space comparison with our encryption scheme is shown in Table I.

TABLE I. KEY SPACE COMPARISON WITH THE EXISTING ALGORITHMS

Algorithm	Key Space
G. Gu, J. Ling [10]	10^{90}
X. Wang et al. [8]	$> 10^{96}$
Proposed algorithm	25×10^{150}

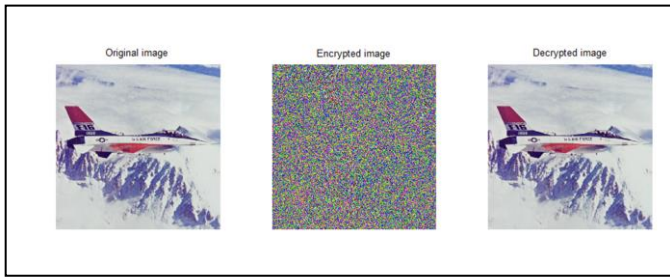


Fig. 5. Encryption and decryption results: Left side with original image, center with encrypted image, and right side with deciphered image of Splash image

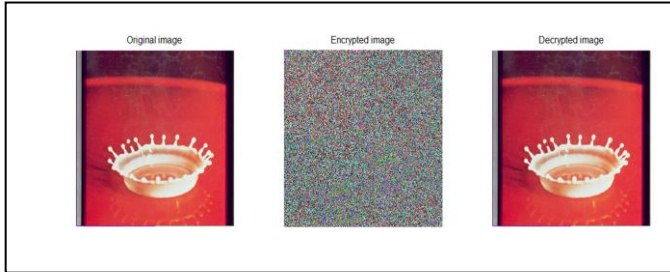


Fig. 6. Encryption and decryption results: Left side with original image, center with encrypted image, and right side with deciphered image of Jet image

B. Statistical analysis

Since most of the existing cipher systems [12] have been cryptanalyzed successfully; we need to prove the robustness and effectiveness of the proposed image encryption scheme against statistical attacks. So in this section, we provide and analyze three statistical tests which are histograms analysis DH and the correlations computation of the adjacent pixels in encrypted images r_{xy} , and information entropy of the ciphered image in order to evaluate the ability of the encryption algorithm to substitute the original image with uncorrelated encrypted image and two other metrics, NPCR and UACI, in order to which evaluate the diffusion characteristics of the encryption algorithm.

1) Histograms of corresponding images

To prevent the information leakage and aggressive attacks [13, 14], it must be ensured that the original and encrypted images do not have any statistical similarity. Histogram analysis is a visual test which reveals the distribution information of pixel values and statistical characteristics of images, by graphing the number of pixels at each color or grey scale intensity level.

An ideal histogram of the encrypted image should have a uniform and completely different histogram against the plain-image. The experimental results of histograms analysis on both original image and its corresponding cipher image using the introduced scheme are shown in “Fig. 7”; where 1st Frame is the original image and its histogram, 2nd Frame is the encrypted image and its corresponding histogram. The histogram of RGB layers are shown in “Fig. 8” where upper figure is the red layer of original image and its histogram, center is the green layer of original image and its histogram, and lower is the blue layer of original image and its histogram.

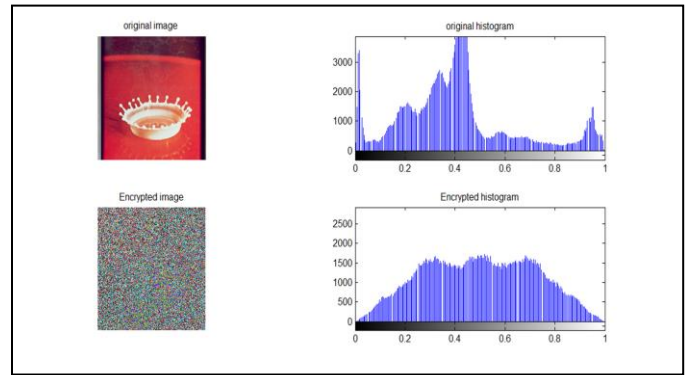


Fig. 7. Histogram analysis of image Splash

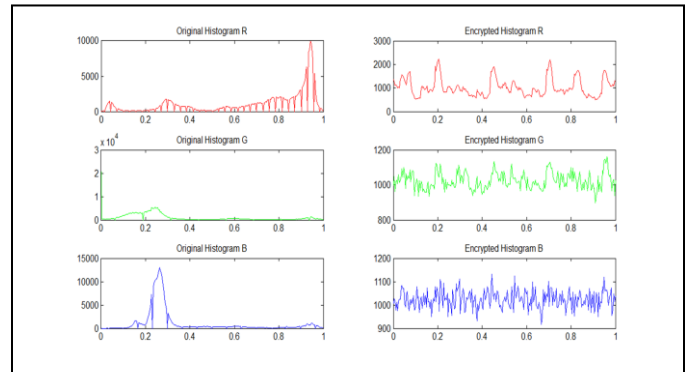


Fig. 8. Histogram Analysis of RGB Layers of Image Splash

2) Correlations Coefficient

In a digital image pixels are not independent and their relevance is great. This may indicate that a large area of the image has similar values. For example, in a common television digital image, the correlation coefficient of adjacent pixels may reach 0.9; that is, the relevance (which means information redundancy) is very big. The smaller the relevance, the better is the encryption effect and the higher is the security. We show the fact that the relevance [2] between plain images and cipher images is very small in two ways. It is important to calculate the correlation coefficients of two adjacent pixels of the encrypted image, so we analyze 2500 pairs of two-adjacent pixels of both original and encrypted images either in vertical, or horizontal, or diagonal directions. To do that, we use the following relations:

$$r_{xy} = \frac{\text{cov}(x,y)}{\sqrt{D(x)}\sqrt{D(y)}} \quad (7)$$

where

$$\text{cov}(x,y) = \frac{1}{N} \sum_{i=1}^N (x_i - E(x))(y_i - E(y)),$$

$$E(x) = \frac{1}{N} \sum_{i=1}^N x_i,$$

$$D(x) = \frac{1}{N} \sum_{i=1}^N (x_i - E(x))^2$$

where $E(x)$ is the mean of x , $D(x)$ denotes the variance at pixel value x , and $\text{cov}(x,y)$ is the covariance between two

color x and y ; where x and y are the gray-scale values of two pixels in the same place in the plain and cipher images, N is the total number of pixels selected from the image for the calculation. For the plain-image, the correlations between adjacent pixels in those directions are close to 1. This means that the adjacent pixels are highly correlated to each other. On the other hand, the correlation coefficients of the encrypted image are close to 0 and hence the adjacent pixels in the encrypted image are entirely uncorrelated to each other as shown in "Fig. 9". All the correlation coefficients are calculated and listed in Table II.

3) Entropy analysis

Entropy is the most significant feature of disorder, or more precisely unpredictability. Relying on Shannon's theory, "it measures [16] the randomness and quantifies the expected value of the information contained in a message". For example, a long sequence of repeating characters has entropy of 0, since every character is predictable and a truly random sequence has maximum entropy, since there is no way to predict the next character in the sequence. The information entropy $H(x)$ of a plain image of a random variable x can be calculated using the following formula:

$$H(x) = \sum_{i=1}^N p(x_i) \log_2 \frac{1}{p(x_i)} \quad (8)$$

where $p(x_i)$ represents the probability distribution of appearance of symbol x . The ideal information entropy value of the cipher image should be close to 8 after encryption, which leads to less possible for the cryptosystem to divulge the information. Table III shows the entropy values of the proposed scheme, as we can find that they are all close to the ideal value 8. As one can see, the information entropy in the proposed algorithm is close to G. Gu, J. Ling [10], and better than X. Wang et al. [8] algorithms as shown in Table IV.

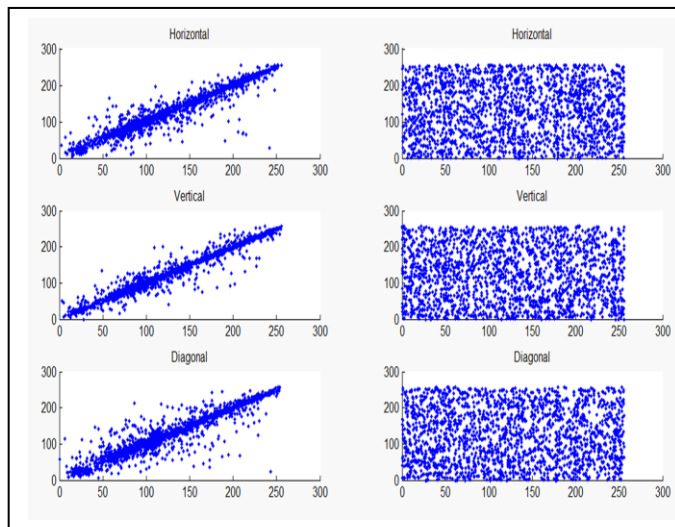


Fig. 9. Correlation chart. Upper frame with horizontal distribution, center with vertical distribution, and lower with diagonal distribution. Left side with Lena tested image, right side the proposed technique

TABLE II. CORRELATION COEFFICIENTS OF ADJACENT PIXELS IN SPLASH IMAGE

Directions	Plain image	Decrypted image
Horizontal	0.925665	0.014981
Vertical	0.998203	-0.008089
Diagonal	0.925010	0.007515

TABLE III. INFORMATION ENTROPIES VALUES

Tested Images	Proposed Scheme
Lena.bmp 512 × 512	7.9994
Splash.bmp 512 × 512	7.9991
Jet.bmp 512 × 512	7.9993
Girl.bmp 512 × 512	7.9993
Lena.bmp 500 × 400	7.9989

TABLE IV. THE RESULTS OF COMPARISON OF INFORMATION ENTROPY

Tested image	The result of information entropies		
	Proposed Scheme	G. Gu, J. Ling [10]	X. Wang et al. [8]
Lena.bmp (256 × 256)	7.99743	7.9999	7.99705

4) Differential attack

A desirable property for cipher image [17] is being sensitive to any tiny changes (e.g., modifying only one pixel) in plain-image to monitor the change in the result. Consequently, we can be able to figure a meaningful relationship between both original and encrypted images. An Assailant often tries to elicit this crucial relationship by making minor changes, usually only one pixel, in the plain-image while encrypting the plain-image with the same encryption keys, and then observes the changes of corresponding cipher image. Also, this action facilitates finding the encryption keys and breaking them.

To test the influence of one-pixel change on the plain image encrypted by the proposed scheme, two common measures may be used [18], number of pixels change rate (NPCR) and unified average changing intensity (UACI), which are calculated. The NPCR is used to measure the percentage of the number of pixels changed in cipher-text after making a slight change in plaintext. Therefore, the theoretical greatest upper-bound of the NPCR is 100%. Therefore, the NPCR is calculated by using the following formula "(9)" and "(10)":

$$NPCR_{R,G,B} = \frac{\sum D_{R,G,B}(m,n)}{M \times N} \times 100\% \quad (9)$$

where

$$D_{R,G,B}(m,n) = \begin{cases} 1, & \text{if } C^1(m,n) \neq C^2(m,n) \\ 0, & \text{if } C^1(m,n) = C^2(m,n) \end{cases} \quad (10)$$

The UACI is used to measure the averaged intensity change for pixels in cipher-text images after making a slight change in a plaintext image. It is demonstrable that the UACI of an ideal cipher for 8-bit gray images is about 0.3346. Therefore, the UACI is calculated by using the following formula "(11)":

$$UACI_{R,G,B} = \frac{1}{M \times N} \sum_{m,n} \frac{|c^1(m,n) - c^2(m,n)|}{255} \times 100\% \quad (11)$$

where c^1 and c^2 are cipher images before and after one pixel change in a plain-image, respectively. The pixel values at grid (m,n) in c^1 and c^2 are denoted as $c^2(m,n)$ and $c^2(m,n)$; a bipolar array D is defined as “(10)”. Table V shows the results of NPCR and UACI for one pixel change in the following standard images. It is clear that, the proposed cipher has good performances in both the NPCR and UACI analyses. Simulation results fit the expectations of the ideal cipher very well.

As one can see, the NPCR and UACI values in the proposed algorithm are close to G. Gu, J. Ling [10] and better than X. Wang et al. [8] algorithms as shown in Table VI.

TABLE V. THE NPCR AND UACI OF CIPHERED IMAGES WITH ONE BIT DIFFERENT BETWEEN THE PLAIN IMAGES

Test Images	NPCR	UACI
Lena.bmp 512 × 512	0.99651	0.34168
Splash.bmp 512 × 512	0.99699	0.35269
Jet.bmp 512 × 512	0.99666	0.32529
Girl.bmp 512 × 512	0.99648	0.32463
Lena.bmp 500 × 400	0.99629	0.34325

TABLE VI. THE NPCR AND UACI PERFORMANCE

Items	Proposed Scheme	G. Gu, J. Ling [10]	X. Wang et al. [8]
NPCR	0.99651	0.99748	0.99586
UACI	0.34168	0.33915	0.33253

VII. TIME COMPLEXITY

Time complexity has become more and more important issue when the scale of an application grows. So, [19] is the relation of computing time and the amount of input. Time complexity of an algorithm signifies the total time required by the program to run to completion. In "computational complexity theory", intuitively the "computational" part means problems that can be modeled and solved by a computer. "The main idea of algorithm's analysis [20] is to have a measure of efficiency of algorithms that doesn't depend on machine specific constants, and doesn't require algorithms to be implemented and time taken by programs to be compared".

Formally, the time complexity $O(f(n))$ of the proposed algorithm can be calculated using the following formula:

$$f(m,n) = \left[2 \times \left(4 \times (16m_1^2 + 16m_2^2 + 16m_3^2 + 16m_4^2 + 16m \times n) \right) \right] + c,$$

where $m_1, m_2, m_3, m_4 < m$

Consequently, the time complexity for the proposed scheme is $O(f(m,n)) = M \times N$.

VIII. CONCLUSIONS

This paper develops an improving permutation—substitution image encryption architecture, based on a chaotic 3D cat map and Turing machine in the form of dynamic random growth technique. A new block encryption scheme was proposed for secure digital images that included two primary processes, Turing machine for substitution and a chaotic 3D cat map for pixels permutation. Many statistical tests were operated to prove the suitability of the algorithm. Experimental tests and statistical analyses have shown effectiveness and robustness of the presented scheme against various attacks; prove that our introduced technique is pretty fast image encryption. According to these pretty characteristics, the proposed encryption scheme may be convenient for different applications, and demonstrate that it reaches a high-security level while transmitting of digital images over the open network. The results of the comparison show that the proposed scheme has more benefits compared to these algorithms of G. Gu, J. Ling [8] and X. Wang et al. [10] ciphers.

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Performance Analysis of CPU Scheduling Algorithms with Novel OMDRRS Algorithm

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Abstract—CPU scheduling is one of the most primary and essential part of any operating system. It prioritizes processes to efficiently execute the user requests and help in choosing the appropriate process for execution. Round Robin (RR) & Priority Scheduling(PS) are one of the most widely used and acceptable CPU scheduling algorithm. But, its performance degrades with respect to turnaround time, waiting time & context switching with each recurrence. A New scheduling algorithm OMDRRS is developed to improve the performance of RR and priority scheduling algorithms. The new algorithm performs better than the popular existing algorithm. Drastic improvement is seen in waiting time, turnaround time, response time and context switching. Comparative analysis of Turn around Time(TAT), Waiting Time(WT), Response Time (RT) is shown with the help of ANOVA and t-test.

Keywords—Round Robin; Turn-around time; Waiting Time; t-test; Anova test

I. INTRODUCTION

CPU scheduling is similar to other types of scheduling, which have been studied over the years. CPU scheduling refers to the decision of allocating a single resource among multiple clients. It also tracks the order of allocation and duration. The primary objective of scheduling is to optimize system performance. Optimization system is considered as most deemed criteria by the system designers [1]. There are numerous algorithms for CPU Scheduling with distinct benefits and shortcomings. To understand and comprehend them in detail, they need to be simulated and performance indices must be studied. This paper depicts the usability of different scheduling algorithms, compare them on the basis of different performance criteria and introduce a newly designed improvised RR scheduling algorithm(OMDRRS). OMDRRS “An Optimum Multilevel Dynamic Round Robin Scheduling Algorithm” is a simulator that has been successfully implemented using VB6.0. The simulator demonstrates the algorithm behavior in opposition to a simulated mix of process loads. The results of all algorithms are compared for the scheduling criteria like turnaround time, waiting time, response time and context switch etc.

A. Scheduling Criteria

Schedulers use different scheduling criteria to enhance the performance of CPU.

Utilization/Efficiency: CPU should be best utilized by allocating the significant tasks; so that it should not be ideal.

Throughput: To increase the number of processed jobs per hour.

Turnaround time: Total time taken from submission of the process till the completion. Turnaround time should minimize the time of users who wait for the output.

Waiting time: Should be minimized as it is the total time spent in ready queue

Response Time: Is the duration after submission till the response. It should be minimal in case of interactive users.

Fairness: CPU should be unbiased and every process should get its fair time to execute.

B. Organization of the Paper

This paper is divided into four sections. Section I gives a brief introduction on the various aspects of the scheduling algorithms, the approach of the current paper and the motivational factors leading to this improvement. Section II presents the hypothesis of study, research methodology, data collection, methods used and the pseudo code of Dynamic Round Robin Algorithm (OMDRRS). In Section III, an experimental analysis and Result of our algorithm OMDRRS and its comparison with the RR algorithm, Priority scheduling and FCFS algorithm is presented. Conclusion and future scope is presented in Section IV.

II. OBJECTIVE

The objective of the study is to compare the performances of different CPU scheduling Algorithms with OMDRRS.

A. Hypothesis of Study

The main or the principal instrument in the research is hypothesis. Its main factors are to suggest new experiments and observations. A number of experiments have been conducted with the deliberate objective of testing the hypothesis.

- Hypothesis 1.1

Ho1.1: There is no significant difference in Turnaround time of various CPU Scheduling Algorithms

H11.1: There is significant difference in Turnaround time of various CPU Scheduling Algorithms.

- Hypothesis 1.2

Ho1.2: There is no significant difference in Waiting time of various CPU Scheduling Algorithms

H1 1.2: There is significant difference in Waiting time of various CPU Scheduling Algorithms

- Hypothesis 1.3

Ho 1.3: There is no significant difference in Response time of various CPU Scheduling Algorithms.

H1 1.3: There is significant difference in Response time of various CPU Scheduling Algorithms.

B. Research Methodology

- Sample Size

The sample size is taken as 50.

C. Data Collection Method

Primary data is entered through the designed simulator by 50 respondents and performance of various CPU Scheduling algorithms is calculated on each of them.

D. Choice of Respondents

The study focused upon the performance analysis of various CPU Scheduling algorithms. The respondents were filtered on the basis of their knowledge on operating systems and CPU scheduling algorithms.

E. Simulator

The purpose of designing the simulator was to provide exactly the same processes to all CPU Scheduling Algorithms for performance comparison without any variations. The simulator was implemented to simulate the operations of First Come First Serve, Shortest Job First (Non Preemptive & Preemptive), Highest Priority, Round Robin and Improved Round Robin scheduling algorithms. These algorithms were implemented in order to establish a valid premise for effective comparison. Simulator has two operating modes: Manual Process Entry and Automatic Process Generator. In Automatic Process Generator system, it fetches all the active processes with burst time and assumes that all the processes were arrived at same arrival time and have no priority. Whereas in the manual entered process, user enters the burst time as per their requirement, as well as arrival time and priority of the each process. In this study respondents entered primary data in manual mode. Based on the selected scheduling algorithm, the average turnaround time, average waiting time, response time, context switch and Gantt chart were calculated. The simulation was run several times to ensure fairness to all datasets each algorithm is evaluated using Average Turn-around Time, Average Waiting Time, Response Time and Gantt chart as the performance evaluation indices.

F. Proposed Algorithm

The proposed algorithm^[8] combines the working principle of fundamental scheduling algorithms. Dynamically Time Slice (DTS) is calculated which allocates different time quantum for each process based on priority, shortest CPU burst time and context switch avoidance time.

Step 1:

Compute the factor analysis $F = \text{Burst time} * 0.2 + \text{Arrival time} * 0.3 + \text{Priority of the process} * 0.5$

Step 2:

Shuffle the processes in ascending order according to the factor of each process in the ready queue (RQ) such that the head of the ready queue contains the lowest factor process based on the burst time, arrival time & priority of the process.

Step 3:

low= RQ(burst value of the first process),
high=RQ(burst value of the last process)
 $TQ = (\text{low} + \text{high}) / 2$

Step 4:

Assign the time quantum and apply for each process say $k = TQ$.

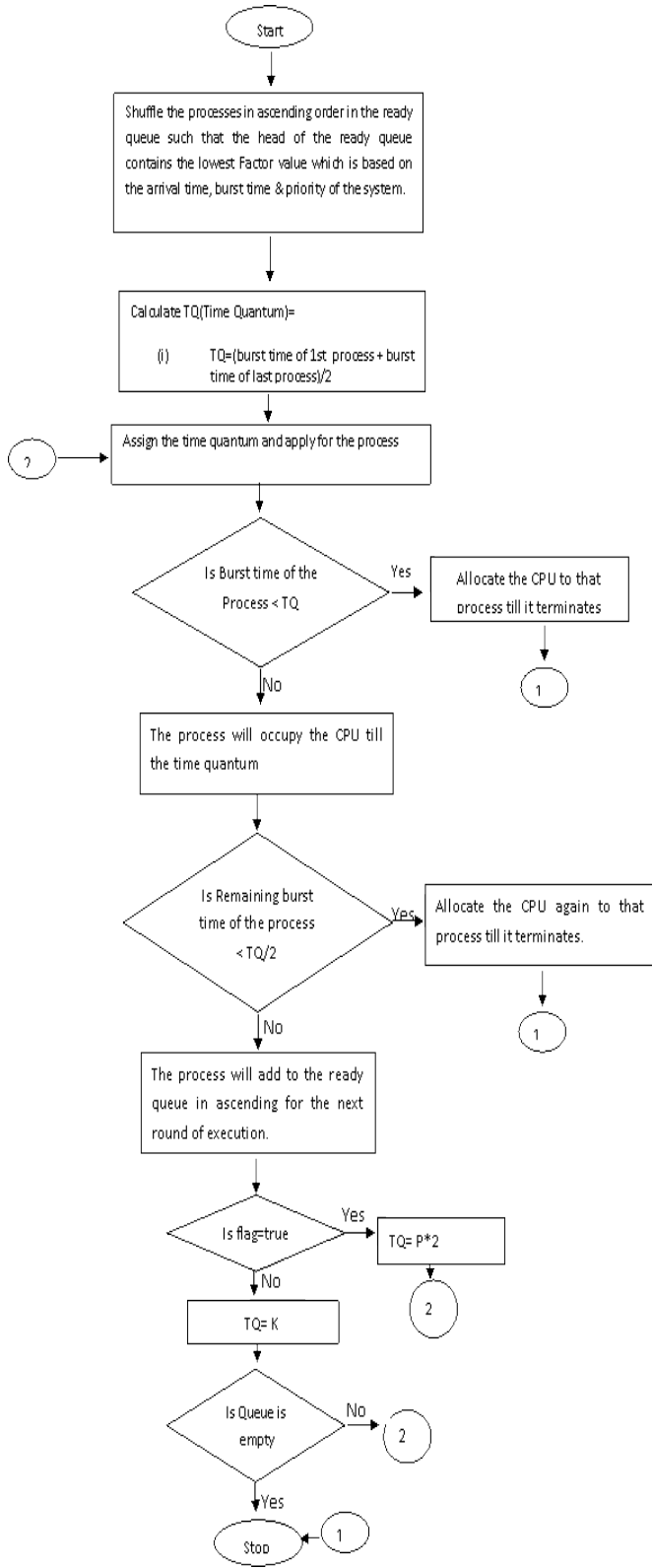
Step 5:

```
IF (burst time of the process < k)
{
    Allocate the CPU to that process till it
    terminates.
}
ELSE IF (Remaining burst time of the process <
k/2)
{
    Allocate the CPU again to that process till it
    terminates.
}
ELSE
{
    The process will occupy the CPU till the time
    quantum and it is added to the ready queue in
    ascending order according to the remaining
    burst time for the next round of execution.
     $TQ = TQ * 2$  or  $TQ = TQ / 2$ 
     $K = TQ$ 
    Goto Step 4
}
```

G. Time Complexity of OMDRRS Algorithm

The OMDRRS algorithm would be maintaining all jobs based on the factor analysis that is ready for execution in a queue. Insertion of each job will be achieved in $O(1)$, but deletion would require $O(n)$ time, where n is the number of processes in the queue. Whenever a process arrives, a record for it can be inserted into the queue based on its burst time, arrival time and priority in $O(n)$ time, where n is the number of processes in the queue. Thus, the time complexity of OMDRRS is equal to that of a typical linear sorting algorithm which is $O(n)$. If a new task arrives it is then sorted with the remaining processes and then executed in the same way. To find a task with the lowest burst time the scheduler needs to search in the ready queue, then the order of searching would be $O(n)$.

H. Logic Diagram of OMDRRS Algorithm



III. DATA ANALYSIS METHOD

Burst time, Arrival time and Priority of the processes were filled by 50 respondents in developed simulator. The simulator generated results of various scheduling algorithms and novel OMDRRS which were further analyzed and compared with anova and t-test.

• Ethical Considerations

The author didn't modified the existing CPU scheduling algorithm concepts and implemented them as it is in the simulator with the new version of round robin algorithm i.e. OMDRR.

A. Experimental Analysis

Fifty processes have been defined by CPU burst time, arrival time and priority of the processes. These fifty processes are scheduled in first come first serve, shortest job first, priority scheduling, round robin scheduling and also in the proposed algorithm. The turnaround time and waiting time have been calculated through simulator and the results were compared.

TABLE I. CALCULATED TURNAROUND TIME OF FCFS, SJF(NP), SJF, PS, RR AND OMDRRS ALGORITHM THROUGH SIMULATOR

PID	Burst Time	Arrival Time	PRIORITY	FCFS	SJF(NP)	SJF(P)	PS	RR	OMDRRS
1	23	0	3	23	23	428	231	637	175
2	34	5	1	523	766	766	72	827	789
3	34	3	3	321	698	698	269	825	803
4	12	6	4	660	169	147	393	571	211
5	8	8	2	744	83	62	209	305	75
6	10	4	5	358	111	90	493	477	140
7	31	1	1	54	631	631	32	765	841
8	23	2	4	120	495	495	355	668	377
9	9	3	5	272	92	71	483	465	84
10	16	6	1	593	301	279	88	551	130
11	1	1	2	547	24	9	164	177	16
12	12	8	3	756	181	159	296	583	237
13	15	9	9	778	269	247	822	622	613
14	6	6	1	666	47	26	94	265	26
15	7	2	5	127	54	33	460	67	36
16	9	3	4	348	101	80	364	464	45
17	11	5	8	369	133	123	712	527	418
18	7	8	9	763	75	54	807	344	461
19	4	9	6	822	35	14	606	356	241
20	15	5	2	455	254	232	179	492	114
21	20	6	4	613	428	406	413	739	454
22	14	3	5	263	209	187	474	471	225
23	7	2	6	97	61	40	579	74	67
24	24	1	2	78	542	542	118	653	199
25	22	5	5	407	450	450	515	702	483
26	16	8	3	736	333	311	312	591	332
27	33	5	8	440	664	664	745	830	830
28	12	3	7	339	157	135	618	475	296
29	22	6	2	688	472	472	201	735	354
30	19	9	9	841	368	346	841	758	680
31	34	5	5	489	732	732	549	829	817
32	40	9	4	818	841	841	453	841	760
33	12	1	8	90	145	101	662	400	273
34	15	3	3	235	239	217	284	449	99
35	7	5	6	574	68	47	586	227	161
36	14	3	4	249	195	173	378	463	154
37	16	5	6	385	285	263	602	524	434
38	28	2	8	155	570	570	690	775	641
39	35	6	9	648	801	801	800	833	775
40	16	8	7	704	317	295	634	599	542
41	20	2	4	175	388	366	332	672	261
42	15	3	2	287	224	202	163	442	60
43	4	2	3	179	31	6	235	44	20
44	6	3	1	327	41	20	38	96	15
45	23	5	5	546	518	518	572	717	506
46	16	8	7	720	349	327	650	607	558
47	11	2	8	190	122	112	701	427	284
48	20	5	9	567	408	386	765	729	598
49	30	2	2	220	600	600	148	771	407
50	3	5	4	577	27	10	381	188	29

TABLE II. CALCULATED WAITING TIME OF FCFS, SJF(NP), SJF, PS, RR AND OMDRRS ALGORITHM THROUGH SIMULATOR

PID	Burst Time	Arrival Time	PRIORITY	FCFS	SJF(NP)	SJF(P)	PS	RR	OMDRRS
1	23	0	3	0	0	405	208	614	152
2	34	5	1	484	727	727	33	788	750
3	34	3	3	284	661	661	232	788	766
4	12	6	4	642	151	129	375	553	193
5	8	8	2	728	67	46	193	289	59
6	10	4	5	344	97	76	479	463	126
7	31	1	1	22	599	599	0	733	809
8	23	2	4	95	470	470	330	643	352
9	9	3	5	260	80	59	471	453	72
10	16	6	1	571	279	257	66	529	108
11	1	5	2	541	18	3	158	171	10
12	12	8	3	736	161	139	276	563	217
13	15	9	9	754	245	223	798	598	589
14	6	6	1	654	35	14	82	253	14
15	7	2	5	118	45	24	451	58	27
16	9	3	4	336	89	68	352	452	33
17	11	5	8	353	117	107	696	511	402
18	7	8	9	748	60	39	792	329	446
19	4	9	6	809	22	1	593	343	228
20	15	5	2	435	234	212	159	472	94
21	20	6	4	587	402	380	387	713	428
22	14	3	5	246	192	170	457	454	208
23	7	2	6	88	52	31	570	65	58
24	24	1	2	53	517	517	93	628	174
25	22	5	5	380	423	423	488	675	456
26	16	8	3	712	309	287	288	567	308
27	33	5	8	402	626	626	707	792	792
28	12	3	7	324	142	120	603	460	281
29	22	6	2	660	444	444	173	707	326
30	19	9	9	813	340	318	813	730	652
31	34	5	5	450	693	693	510	790	778
32	40	9	4	769	792	792	404	792	711
33	12	1	8	77	132	88	649	387	260
34	15	3	3	217	221	199	266	431	81
35	7	5	6	562	56	35	574	215	149
36	14	3	4	232	178	156	361	446	137
37	16	5	6	364	264	242	581	503	413
38	28	2	8	125	540	540	660	745	611
39	35	6	9	607	760	760	759	792	734
40	16	8	7	680	293	271	610	575	518
41	20	2	4	153	366	344	310	650	239
42	15	3	2	269	206	184	145	424	42
43	4	2	3	173	25	0	229	38	14
44	6	3	1	318	32	11	29	87	6
45	23	5	5	518	490	490	544	689	478
46	16	8	7	696	325	303	626	583	534
47	11	2	8	177	109	99	688	414	271
48	20	5	9	542	383	361	740	704	573
49	30	2	2	188	568	568	116	739	375
50	3	5	4	569	19	2	373	180	21

B. Results of Analysis

In order to analyze the difference in performance of the various CPU scheduling algorithms, ANOVA and t-test are used and the following results are obtained as follows:

TABLE III. COMPARISON OF TURNAROUND TIME IN FCFS, SJF(NP), SJF, PS, RR AND OMDRRS ALGORITHM

ANOVA						
Source of Variation	SS	Df	MS	F	P-value	F crit
Between Groups	2164333	5	432866.7	7.398212	1.49E-06	2.244703
Within Groups	17201834	294	58509.64			
Total	19366167	299				

In the above table of ANOVA turnaround time is significantly differ for different scheduling algorithms, because

the computed value (F) is greater than tabulated value (F crit) at 5% level of significance. Hence H0 1.1 is rejected. Rejecting null hypothesis proves that there is significant difference in Turnaround time of various CPU Scheduling Algorithms.

TABLE IV. COMPARISON OF WAITING TIME IN FCFS, SJF(NP), SJF, PS, RR AND OMDRRS ALGORITHM

ANOVA						
Source of Variation	SS	Df	MS	F	P-value	F crit
Between Groups	2164333	5	432866.7	7.808022	6.44E-07	2.244703
Within Groups	16298981	294	55438.71			
Total	18463314	299				

In the above table of ANOVA shows that overall waiting time is significant. Since the computed value (F) is greater than tabulated value (F crit) at 5% level of significance. Hence H01.2 is rejected. Rejecting null hypothesis proves that there is significant difference in waiting time of various CPU Scheduling Algorithms.

TABLE V. COMPARISON OF RESPONSE TIME IN FCFS, SJF(NP), SJF, PS, RR AND OMDRRS ALGORITHM

ANOVA						
Source of Variation	SS	df	MS	F	P-value	F crit
Between Groups	2064121	5	412824.3	8.606909	1.26E-07	2.244703
Within Groups	14101500	294	47964.29			
Total	16165622	299				

In the above table of ANOVA depicts that overall response time is significant. Since the computed value (F) is greater than tabulated value (F crit) at 5% level of significance. Hence H01.3 is rejected. Rejecting null hypothesis proves that there is significant difference in response time of various CPU Scheduling Algorithms.

TABLE VI. COMPARISON OF TURN AROUND TIME BETWEEN PRIORITY SCHEDULING AND DYNAMIC ROUND ROBIN ALGORITHM WITH THE HELP OF T-TEST

	PS	OMDRRS
Mean	431.36	342.92
Variance	55263.05	69012.8098
Observations	50	50
Pearson Correlation	0.397149	
Hypothesized Mean Difference	0	
Df	49	
t Stat	2.280126	
P(T<=t) one-tail	0.013495	
t Critical one-tail	1.676551	
P(T<=t) two-tail	0.026989	
t Critical two-tail	2.009575	

At 5% level of significance h01.1 is rejected hence there is a significant difference between turnaround time of priority scheduling and proposed dynamic round robin algorithm(OMDRRS).

TABLE VII. COMPARISON OF TURN AROUND TIME BETWEEN ROUND ROBIN SCHEDULING AND DYNAMIC ROUND ROBIN ALGORITHM WITH THE HELP OF T-TEST

	RR	OMDRRS
Mean	532.98	342.92
Variance	50676.26	69012.81
Observations	50	50
Pearson Correlation	0.810487	
Hypothesized Mean Difference	0	
Df	49	
t Stat	8.706293	
P(T<=t) one-tail	8.11E-12	
t Critical one-tail	1.676551	
P(T<=t) two-tail	1.62E-11	
t Critical two-tail	2.009575	

At 5% level of significance $h_{01.1}$ is rejected hence there is a significant difference between turnaround time of round robin algorithm and dynamic round robin algorithm(OMDRRS)

TABLE VIII. COMPARISON OF WAITING TIME BETWEEN PRIORITY SCHEDULING AND DYNAMIC ROUND ROBIN ALGORITHM WITH THE HELP OF T-TEST

	PS	OMDRRS
Mean	409.94	321.5
Variance	54987.81	64623.32
Observations	50	50
Pearson Correlation	0.372315	
Hypothesized Mean Difference	0	
Df	49	
t Stat	2.280126	
P(T<=t) one-tail	0.013495	
t Critical one-tail	1.676551	
P(T<=t) two-tail	0.026989	
t Critical two-tail	2.009575	

At 5% level of significance $h_{11.2}$ is rejected hence there is a significant difference between waiting time of Priority Scheduling and Dynamic Round Robin Algorithm. Above table proves that Proposed algorithm OMDRRS is better than the Priority scheduling algorithm in terms of waiting time.

TABLE IX. COMPARISON OF WAITING TIME BETWEEN ROUND ROBIN SCHEDULING AND DYNAMIC ROUND ROBIN ALGORITHM WITH THE HELP OF T-TEST

	RR	OMDRRS
Mean	511.56	321.5
Variance	46738.5	64623.32
Observations	50	50
Pearson Correlation	0.796369	
Hypothesized Mean Difference	0	
Df	49	
t Stat	8.706293	
P(T<=t) one-tail	8.11E-12	
t Critical one-tail	1.676551	
P(T<=t) two-tail	1.62E-11	
t Critical two-tail	2.009575	

At 5% level of significance $h_{11.2}$ is rejected hence there is a significant difference between waiting time of Round Robin Scheduling and Dynamic Round Robin Algorithm. Hence OMDRRS is also better than the round robin scheduling algorithm.

TABLE X. COMPARISON OF TURNAROUND TIME BETWEEN PRIORITY SCHEDULING AND DYNAMIC ROUND ROBIN ALGORITHM WITH THE HELP OF T-TEST

	PS	OMDRRS
Mean	431.36	342.92
Variance	55263.05	69012.8098
Observations	50	50
Pearson Correlation	0.397149	
Hypothesized Mean Difference	0	
Df	49	
t Stat	2.280126	
P(T<=t) one-tail	0.013495	
t Critical one-tail	1.676551	
P(T<=t) two-tail	0.026989	
t Critical two-tail	2.009575	

At 5% level of significance $h_{11.1}$ is rejected hence there is a significant difference between turnaround time of Priority Scheduling and Dynamic Round Robin Algorithm. Above table proves that our proposed algorithm OMDRRS is better than the Priority scheduling algorithm in terms of turnaround time.

TABLE XI. COMPARISON OF TURNAROUND TIME BETWEEN ROUND ROBIN SCHEDULING AND DYNAMIC ROUND ROBIN ALGORITHM WITH THE HELP OF T-TEST

	RR	OMDRRS
Mean	532.98	342.92
Variance	50676.26	69012.81
Observations	50	50
Pearson Correlation	0.810487	
Hypothesized Mean Difference	0	
Df	49	
t Stat	8.706293	
P(T<=t) one-tail	8.11E-12	
t Critical one-tail	1.676551	
P(T<=t) two-tail	1.62E-11	
t Critical two-tail	2.009575	

At 5% level of significance $t_{11,1}$ is rejected hence there is a significant difference between turnaround time of Round Robin Scheduling and Dynamic Round Robin Algorithm. Above table proves that our proposed algorithm OMDRRS is better than the Round Robin scheduling algorithm in terms of turnaround time.

TABLE XII. DETERMINISTIC STATISTICS OF VARIOUS SCHEDULING ALGORITHM

	FCFS	PS	RR	OMDRRS
Mean	438.72	431.36	532.98	342.92
Standard Error	34.32485908	33.24546629	31.83591208	37.15179936
Median	423.5	433	561	278.5
Standard Deviation	242.7134062	235.0809465	225.1138932	262.7028926
Confidence Level(95.0%)	68.97838553	66.80926454	63.97665936	74.65933461
95% confidence interval for execution of TAT	(335.75, 541.69)	(331.62, 531.09)	(437.47, 628.49)	(231.46, 454.37)

The confidence interval for execution time of turnaround time of various scheduling algorithms at 95% shows that the time is varying between 231.46 nanoseconds to 454.37 nanoseconds that is 95% of the jobs will complete its execution within this time frame.

IV. CONCLUSION AND FUTURE SCOPE

This paper statistically analyzes and compares various scheduling algorithms with proposed algorithm. It analysis the performance of various CPU scheduling algorithms with respect to dynamic time slice concept in Round Robin CPU scheduling. The suggested algorithm and other scheduling algorithm are executed on the simulator and evaluated on Anova and t-test. The Results clearly depicts that OMDRRS

performs better than existing algorithms on the basis of Anova and t-test results comparative analysis of Turnaround time, waiting time & context switch clearly shows that OMDRRS gives much better turnaround time with very less waiting time. Deterministic statistics shows that the confidence index is improved in the case of OMDRRS algorithm. Results concludes that the proposed algorithm is superior then commonly used algorithm with less waiting response time, less turnaround time and context switching; thereby reducing the overhead and results in saving lots of memory space. Taking the base of proposed algorithm more improvement can be made the future.

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Assessment and Comparison of Fuzzy Based Test Suite Prioritization Method for GUI Based Software

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Abstract—The testing of event driven software has significant role to improve overall quality of software. Due to event driven nature of GUI based software many test cases are generated and it is difficult to identify test cases whose fault revealing capability is high. To identify those test cases test suite prioritization is done. Various test suite prioritization methods exists for GUI based software in literature. Prioritization methods improve the rate of fault detection. In our previous work we have proposed a fuzzy model for test suite prioritization of GUI based software. In this method priority is assigned on the basis of multiple factors using fuzzy model. These factors are: The type of event, Event Interaction, and Parameter-value interaction coverage-based criteria. Using this method a test oracle will be organized in the descending order of its effectiveness. In this paper we are evaluating the proposed fuzzy model and comparing results with other prioritization methods.

Keywords—Test suite prioritization; Fuzzy Model; Comparison of prioritization methods

I. INTRODUCTION

Testing is a quality assurance activity [10]. Testing of GUI impose many challenges due to its event driven nature and interaction with the user [9]. Event driven software takes sequence of events as input after change of state generate new sequence of input as output [1, 8]. Many test cases are generated to cover combination of all events. To improve the effectiveness these test cases generated for event driven software a prioritization method are developed. The aim of this prioritization is to reorder the test cases according to fault revealing capability of test cases.

This paper is organized as follows: Section II describes the research background for the proposed work. Section III describes Fuzzy model. Section IV discuss about evaluation of fuzzy based test case prioritization method. Section V covers assessment of effectiveness of test suite. Conclusion is presented in section VII.

II. RESEARCH BACKGROUND

The problem of test suite prioritization is widely explored by many researchers. Few researchers have significant contribution in this area. One imperative work is proposed by Memon Atif M, Bryce Renee C. (2007). In this work test cases are prioritized using event interaction coverage. Rate of fault detection improves for the test suits which have the larger number of 2 way and 3 way interaction. Huang Chin-Yu et al. (2010) in their work proposed three weight based method. These methods are Equal weight, Fault prone weight and Random weight. They have classified events and assign weight value based on their importance. Results shows that high to low Adjusted weight method find more fault. Another work by Bryce Renee C. et al. (2011) proposed cost-based combinatorial interaction coverage metric as 2way interaction coverage and cost-based 2way interaction coverage. According to experimental results the difference in APFDC between 2way and cost-based 2way for CPM was less than 3%. Wei Sun et al. proposed a multi-objective test case prioritization strategy is proposed to combine two objectives: statement coverage and event coverage for GUI applications. Experiment shows that multi-objective has better performance than both single-objective strategies.

III. FUZZY MODEL

This model was proposed in our previous work [5]. In this model multiple factors are considered for prioritization. Major components of GUI are identified as prioritization factors. These factors are: The type of event, Event Interaction, and Parameter-value interaction coverage-based criteria. A fuzzy model is developed for test suite prioritization which considers these factors as input and assigns weight value to each test case. These factors are categorized as very low, low, medium, high and very high. Impact of these factors are categorize in five categories as very low, low, medium, high and very high. According to this impact priority is assigned. Random tiebreaking is used when weight value comes same for multiple test cases. Total 125 rules are generated for fuzzy model. Centre of gravity (COG) is used for defuzzification.

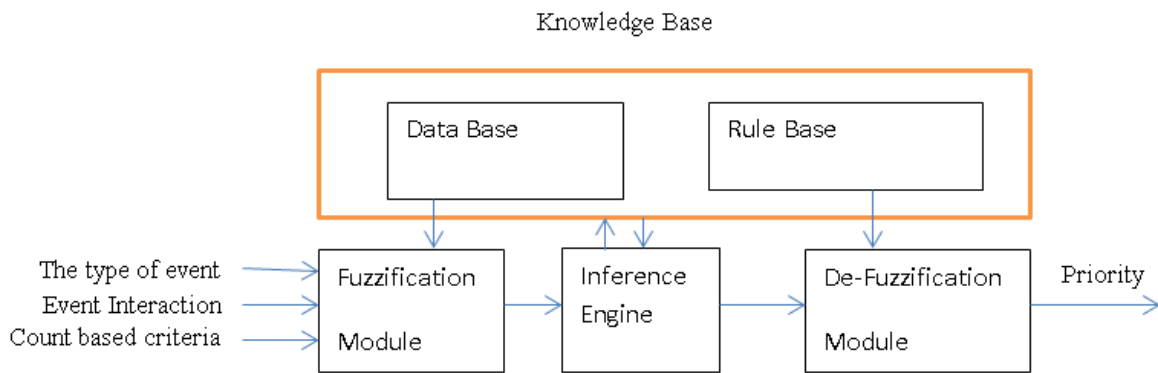


Fig. 1. Fuzzy Model for Prioritization [5]

A. Evaluation of fuzzy based test case prioritization method

For the evaluation of fuzzy based test case prioritization method application Terppaint 3.0 and some of its L1 and L2 test cases are considered. For each test case its parameter count, event sum and number of event Interaction covered are counted for the input of fuzzy model. These values are shown in table 1.

TABLE I. TERPPAINT 3.0 TEST CASES

Test Case	Parameter count	Event Sum	Event Interaction
T1	57	6	1
T2	95	14	1
T3	94	13	1
T4	112	16	1
T5	74	12	1
T6	58	10	1
T7	76	12	1
T8	75	12	1
T9	36	8	2
T10	77	5	2

A parser is developed to calculate the Parameter Count Event Sum and Event Interaction. These values are given as input to the fuzzy model and output value is generated. Output of fuzzy model is shown in table II as FIS output.

On the basis of output of fuzzy model priority is assigned to each test case. Maximum priority is assigned to those test cases

whose FIS output is highest and lowest priority is assigned to those test cases whose FIS output is lowest.

TABLE II. APFD FOR TERPPAINT 3.0

Test Case	Parameter count	Event Sum	Event Interaction	FIS Output	Priority
T1	57	6	1	0.22	10
T2	95	14	1	0.82	1
T3	94	13	1	0.811	2
T4	112	16	1	0.487	6
T5	74	12	1	0.565	4
T6	58	10	1	0.366	8
T7	76	12	1	0.383	7
T8	75	12	1	0.577	3
T9	36	8	2	0.232	9
T10	77	5	2	0.535	5

Once priority is assigned to test cases evaluation of prioritized sequence is done. In order to evaluate prioritized sequence faults identified by each test case in previous run is considered.

With the count of number of faults identified by the test case and its priority assigned by the fuzzy model its rate of fault detection has been calculated as shown in table 3.

TABLE III. PRIORITY TABLE FOR TEST CASES OF TERPPAINT 3.0

Test Case	Parameter count	Event Sum	Event Interaction	FIS Output	Priority	APFD Calculation
T1	57	6	1	0.22	10	10
T2	95	14	1	0.82	1	2
T3	94	13	1	0.811	2	6
T4	112	16	1	0.487	6	18
T5	74	12	1	0.565	4	12
T6	58	10	1	0.366	8	24

T7	76	12	1	0.383	7	42
T8	75	12	1	0.577	3	12
T9	36	8	2	0.232	9	45
T10	77	5	2	0.535	5	40

B. Assessment of effectiveness of test suite:

Average Percentage of Fault Detection (APFD) metric is considered for the assessment of effectiveness of test suite. This metric calculates rate of fault detection. Higher value of this metric represents effective prioritized sequence. APFD is defined as (Rothermel et al., 2001):

$$APFD = 1 - \frac{TF_1 + TF_2 + \dots + TF_m}{nm} + \frac{1}{2n} \quad (1)$$

Where TF_1 : number of faults identified by first test case

n : total number of test cases under consideration

m : total number of faults identified by test cases

Final APFD value for TerpPaint 3.0 is 0.49. These results are further compared with APFD of Parameter count method and event weight method and results are shown in table 5.

TABLE IV. COMPARISON WITH OTHER METHODS

Test Case	APFD Calculation for Parameter Count		% Event Coverage	Event Weight	APFD Calculation for Event Weight		APFD for Fuzzy based Prioritization Method	
	Priority	APFD			Priority	APFD	Priority	APFD
T1	9	9.00	0.80	1.61	9	9.00	10	10
T2	2	4.00	2.01	10.04	1	2.00	1	2
T3	3	9.00	2.01	10.04	2	6.00	2	6
T4	1	3.00	1.20	3.61	6	18.00	6	18
T5	7	21.00	1.61	6.43	3	9.00	4	12
T6	8	24.00	1.20	3.61	7	21.00	8	24
T7	5	30.00	1.20	3.61	8	48.00	7	42
T8	6	24.00	1.61	6.43	4	16.00	3	12
T9	10	50.00	0.80	1.61	10	50.00	9	45
T10	4	32.00	1.61	6.43	5	40.00	5	40
	Final APFD	0.51				0.47		0.49

For the comparison of results prioritization is done for the same test suite using various methods as shown in table 5 and their APFD value is calculated for the comparison.

TABLE V. COMPARISON OF VARIOUS PRIORITIZATION METHODS

Prioritization Method	APFD %
Fuzzy Method	0.49
Best order of test cases	0.66
Worst order of test cases	0.31
Event Weight Method	0.47
Random order of test cases	0.36

It is demonstrated in figure 2 that Fuzzy based prioritization method outperform most of the prioritization methods. APFD of Fuzzy based test case prioritization method is very close to best method of parameter count based method. But these results will very when number of parameters will be different in other applications.

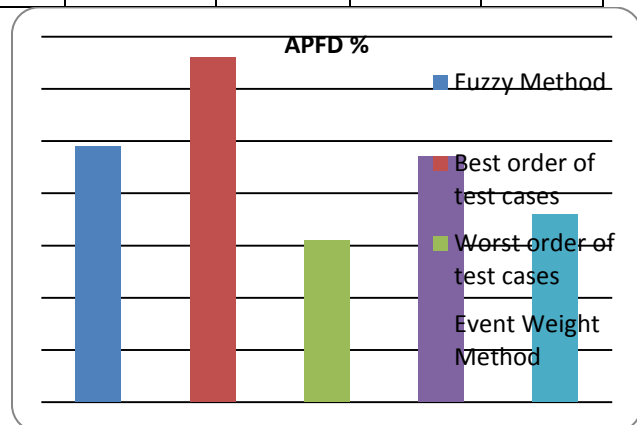


Fig. 2. Comparison of various test case prioritization methods

IV. CONCLUSION

A fuzzy based technique which was proposed in previous research to assign priority of test cases is evaluated in this paper. It is further compared with existing prioritization methods. Experimental results shows that fuzzy based

prioritization method outperform most of the prioritization methods. APFD of Fuzzy based test case prioritization method is very close to best order of test cases. Experimental results shows that the proposed fuzzy model is proved to be an effective approach for test case prioritization for GUI based software.

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Bag of Features Model Using the New Approaches: A Comprehensive Study

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Abstract—The major challenge in content based image retrieval is the semantic gap. Images are described mainly on the basis of their numerical information, while users are more interested in their semantic content and it is really difficult to find a correspondence between these two sides. The bag of features (BoF) model is an efficient image representation technique for image classification. However, it has some limitations for instance the information loss during the encoding process, an important step of BoF. This is because the encoding is usually done by hard assignment i.e. in vector quantization each feature is encoded by being assigned to a single visual word. Another notorious disadvantage of BoF is that it ignores the spatial relationships among the patches, which are very important in image representation. To address those limitations and enhance the results, novel approaches were proposed at each level of the BoF pipeline. In instance the combination of local and global descriptors for a better description, a soft-assignment encoding manner with a spatial pyramid partitioning for a more informative image representation and a maximum pooling to get the final descriptors. Our work aims to give a detailed version of the BoF, including all the levels of the pipeline, as a support leading to a better comprehension of the approach. We also compare and evaluate the state-of-the-art approaches and find out how these changes at each level of the pipeline could affect the performance and the overall classification results.

Keywords—Bag of features; Image classification; Local and global descriptors; Locality-constrained Linear Coding; Spatial pyramid; Pooling

I. INTRODUCTION

The Content-Based Image Retrieval (CBIR) systems are supposed to provide us with a simple accessible way for researching and retrieving images from a large scale image collection based on semantic and visual content of the images. The main problem in content-based image retrieval (CBIR) and visual recognition arises from the “semantic gap” which is the difference between the information that one can extract from the visual data and the interpretation that this same data has for a user in a given context. In order to fill this gap many challenges need to be overcome, for instance, there is the large scale nature of the problem where the datasets are large, the image descriptor dimension is large and the annotated training data is restricted. [1, 2, 3, 4]. Also, we need to deal with other factors, such as the viewpoint, scale, rotation and illumination changes [5, 6].

In addition to this, occlusion [7] and clutter [8] make the visual recognition task even harder since an object can be

occluded and only part of it is visible. Finally, there is the elusive notion of similarity which includes: the intra-class variation, which is the diversity that exists between instances of the same object and the inter-class similarity, or the similitude between instances of different object categories.

Our work is based on the bag of features approach. The main idea is that a set of local image patches is sampled using some method (e.g. keypoints detector) then the local descriptors are extracted from the patches using (e.g. SIFT descriptor [5]). The resulting descriptors are then quantified to form a codebook (e.g. using K-means) then the image descriptors are encoded by being projected to the linear subspace of the closest visual words (e.g. using LLC). Now, we need to go from these obtained code vectors to a visual representation, for this we can use the spatial pyramid scheme in order to capture the spatial information of these descriptors. The local code vectors are aggregated using a pooling method to obtain the final image descriptor. The resulting global descriptor vector is used as a representation of the image for learning and classification purposes (e.g. it is used to learn an image classification rule based on an SVM classifier).

The main goal of this paper is to first give a helpful basis for the new researchers interested in image classification by providing them with a detailed version of the BoF method that includes all the levels of the pipeline. Normally, researchers focus on a specific level of it considering the other levels as known and understood. Then, at each level of the pipeline, we compare the standard methods used in bag of features versus the state-of-the-art methods to see when and how we can combine them to get the best performance accuracy. Finally, we'll clarify the added value of each one of these novel approaches, if it is for capturing more information or for reducing the computational cost.

In section 2, we briefly review the related work. We present the bag of feature pipeline in section 3 with all its levels. We define the purpose of each level and give the most relevant used approaches to achieve it. In section 4 the comparative system is described along with the experiments and results. Section 5 concludes the paper.

II. RELATED WORK

The bag of features (BoF) [9, 1, 10] approach has established itself as the state-of-the-art for image classification. It consists of five steps, the feature extraction phase, the codebook creation and encoding of the features phase, the

feature pooling and finally the learning and classification phase. Researchers tried to improve the standard bag of features [1] by optimizing each and all the steps in the pipeline.

The feature extraction phase is done in two steps, first we sample the patches using either interest points [1, 5, 9, 11, 12, 13, 14, 15, 16, 17, 18, 19, 20] or densely with a regular grid [21, 22, 23, 24]. Until recently in [25], when they combined the two ways and suggested to find interest points within a dense grid. The second step would be to extract the features and get the descriptors, here again we can describe the patches using a global spatial layout like GIST [26], a local descriptor like SIFT [1, 16, 5, 9], a filter based [21, 22] or a raw patch based [14, 20, 15, 17, 19] representations.

To quantize local descriptors into visual words, we must first generate the visual vocabulary. For this purpose, algorithms like mixture of Gaussian [27], mean-shift [28] agglomerative clustering [14, 17] and the most popular of all K-means [1, 20, 19, 22] are used. For the encoding part the simplest way assigns a local descriptor to the closest visual word, giving one and only one nonzero coefficient. Also known as “hard”-assignment, this does not consider codeword ambiguity which causes a significant information loss [29, 30, 31]. In [32, 31], a “soft”-assignment coding is proposed to deal with these problems by assigning a local descriptor to all visual words in a weighted manner. Recently, sparse [33, 34] and local [34, 35, 36, 37] coding proved to be very efficient. They optimize a linear combination of few visual words to approximate a local descriptor and code it with the optimized coefficients.

Once we get our code vectors we need to aggregate them in order to get the final descriptor, sum-pooling (average-pooling) which simply sums the coefficients, it has been commonly used to obtain the image-level. Recent work indicates that max-pooling which chooses the largest coefficient for a visual word can lead to a better classification performance [35, 38, 34, 39]. In order to compare and quantify the influence of some of these methods; in the different levels of the pipeline; and to see how these changes affect the classification performance, we need an image classifier that could predict image labels, once the image descriptors are extracted. For the BoF classification, usually the discriminative methods result in a higher classification accuracy, this is why for this paper we chose to use an SVM classifier [40] which has been the most popular classifier in the past decade.

III. BAG OF FEATURES

Inspired from the original text representation model, bag of words [41] used for document classification, BoF was introduced first by [9] in video retrieval then with [1] for image categorization. An image is represented as an unordered collection of visual words. BoF gives an extremely compact description of images as they are represented as histograms of local descriptors. The main idea is to obtain visual words (features) by quantizing the local descriptors of images in the dataset based on a visual vocabulary. The vocabulary is constructed by clustering a large set of local descriptors using algorithms like K-means [42, 43]. The algorithm takes as an input the training data description and gives as an output a set

of clusters. Each cluster would be represented by one visual word. The image is now represented as a bag of visual words and a histogram can be built with a dimension equal to the visual vocabulary size, each bin will contain the visual word’s frequency with respect to the image.

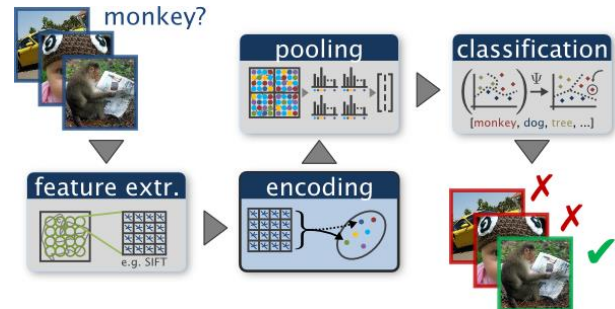


Fig. 1. The BoF pipeline by [4]

A. Feature sampling and extraction

The goal is to obtain a representative set of image patches covering the most relevant information in a given image. After detecting the most interesting regions in each image, the feature extraction pipeline starts in order to compute the vectors that will describe these regions. A local descriptor is used in image categorization and object recognition tasks and also to match similar object instances. Many methods for feature description can be employed; in our work we adopted the commonly most used feature descriptors that achieved state-of-the-art results on several benchmarks over the years.

SIFT descriptor [5] is a sparse feature representation that consists of both feature extraction and detection. To detect scale-invariant characteristic points, the SIFT approach uses cascaded filters, where the difference of Gaussians (DoG), is calculated on rescaled images progressively. Then interest points are described by gradient orientation histograms to get a 128-dimensional vector as the SIFT representation for a pixel. Given an image, SIFT finds all the keypoints in the image with respect to the gradient feature of each pixel. Every keypoint contains the information of its location, local scale and orientation. Then, based on each keypoint, SIFT computes a local image descriptor. Combining all the local descriptors, we get the complete features from the image.

A fast alternative is **dense SIFT descriptor (DSIFT)** which is provided by the VLFeat [44] open source library. DSIFT is an extension of the SIFT algorithm. It makes some new assumptions: (a) the location of each keypoint is not from the gradient feature of the pixel, but from a predesigned location; (b) the scale of each keypoint is all the same which is also predesigned; (c) the orientation of each keypoint is always zero. With this assumptions, DSIFT can acquire more features in less time than SIFT does. Figure 2 below shows the features extracted by both SIFT and DSIFT descriptors from the same images. **SURF** [6] based on the same principles and steps of SIFT. It has a Hessian-based detector and a distribution-based descriptor generator. It uses a blob detector based on the Hessian to find points of interest. In contrast to the Hessian-Laplacian detector by [18], SURF also uses the determinant of the Hessian for selecting the scale, as it is done by [45].

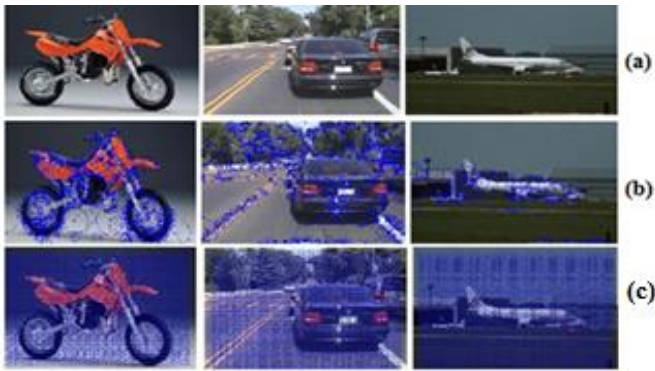


Fig. 2. (a) original images (b) Visualization of SIFT features (c) Visualization of dense SIFT features

An orientation is first assigned to the keypoint. Then a square region is constructed around the keypoint and rotated according to the orientation. The region is split up regularly into smaller 4x4 square sub-regions. For each sub-region, we compute a few simple features at 5x5 regularly spaced sample points. The horizontal and vertical Haar-wavelet responses dx and dy are calculated and summed up over each sub-region and form a first set of entries to the feature vector. The absolute values of the responses $|dx|$ and $|dy|$ are also calculated, together with the sum of vector to form a four-dimensional descriptor. And for all 4x4 sub-regions, it results in a vector of length 64.

GIST [26] captures the spatial characteristics of the scene categories. The main idea is to split an image using a regular grid and compute average response magnitudes of a number of Gabor filters in each spatial cell. It uses Gabor filter to extract a holistic description of a lot properties in the image. These properties are highly related to the underlying scene where the image was taken. It will help determine what type of object the image is showing. The resulting descriptor encodes the existence of edge-like local structures at various orientations and scales.

B. Creation of the codebook and feature quantization

The encoding phase transforms the local descriptor obtained to a new form, using the visual vocabulary (codebook or dictionary).

1) Creating the codebook:

K-means clustering k-means clustering is a vector quantization method and probably the most common way of constructing the visual vocabulary. It aims to partition N descriptors into k clusters in which each descriptor belongs to the cluster with the nearest mean, serving as a prototype of the cluster. This results in a partitioning of the data space into Voronoi cells [46]. To get an example of a visual word from the pipeline we save the paths to images for which we used the descriptors to create the visual words, so that we can visualize our visual vocabulary like shown in figure 3.

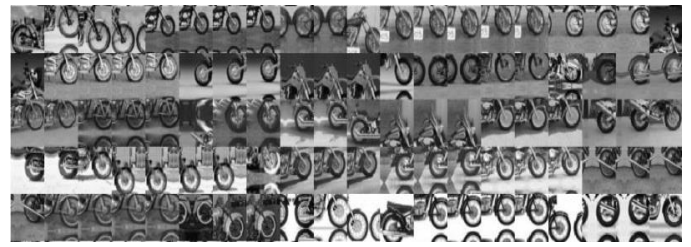


Fig. 3. The image patches corresponding to a resulting visual word from our pipeline, after the codebook creation

2) **Encoding**: The standard BoF method for encoding transforms the local descriptor into a more adapted form using the codebook [1, 42, 9], but it has some shortcomings for instance the information loss caused by vector quantization where a descriptor can be similar to many visual words or be different from all of them. Recently, the hard quantization of features was replaced by alternative methods that would keep more information about the original image features (soft-assignment). A patch is assigned to multiple visual words in a weighted manner according to its proximity to vocabulary centers in the local descriptor space [47, 30, 31]. This has been done in two ways: (1) by expressing features as combinations of visual words (e.g., soft quantization [32], local linear encoding [35]). (2) by recording the difference between the features and the visual words (e.g., Fisher encoding [48], super-vector encoding [49]).

In the encoding step we chose to use three different approaches, first a hard assignment encoding as the standard BoF vector quantization method. Then, in order to better apprehend the new soft-assignment technique, we used a distance-based soft quantization method and a reconstruction-based soft assignment method renowned for its good performance.

Nearest Neighbors requires two parameters the number of neighbors (k) and the distance measure (e.g. Euclidean). The algorithm repeats iteratively the calculation for each descriptor. The code vector (representation) created will be of the size of the dictionary and will contain the frequency count of the assignment of each descriptor to a visual word. To do the assignment we use an L2 distance and pick the nearest neighbor visual word to each descriptor.

The 1-NN algorithm is as follow: Given the data $D = \{x_i, y_i\}$, a distance function d and input x we find: $j = \arg \min_i d(x, x_i)$ and return y_j . The problem is that the strict memorization aspect of 1-NN leads to information loss. One way to deal with this is local averaging; instead of just one neighbor, we find K nearest neighbors and use them in a weighted manner. All neighbors are used, but with different weights. Closer neighbors receive higher weights. The weighting function (kernel function) is the Gaussian: $K(x, x_i) = \exp\left\{\frac{-d^2(x, x_i)}{\sigma^2}\right\}$

The *K*-Nearest Neighbors with Gaussian kernel similarity metric is as follow: Given the data $D = \{x_i, y_i\}$, the Kernel function K and input x . If $y \in \pm 1$ the return weighted majority is $\text{sign}(\sum_{i=1}^n K(x, x_i) y_i)$

Locality-constrained Linear (LLC) Coding hard-assignment selects the first closest visual word, it is fast but it gives a large quantization error. Recently [35] proposed a reconstruction based approach where features are “reconstructed” by a linear combination of visual words.

Let x be a set of D -dimensional local descriptors extracted from an image, i.e. $x = [x_1, x_2, \dots, x_N] \in R^{D \times N}$ where D is the descriptor’s dimension (e.g. SIFT=128) and N is the number of descriptors ($D \times N$ matrix).

Given a codebook with M entries, $B = [b_1, b_2, \dots, b_M] \in R^{D \times M}$. The coding scheme converts each descriptor into an M -dimensional code to generate the final image representation. For a descriptor x and a codebook B , coefficients are chosen to solve:

$$\arg \min_{\gamma} \sum_{i=1}^N \|x_i - B\gamma_i\|^2 + \lambda \|d_i \odot \gamma_i\|^2 \text{ s.t. } 1^T \gamma_i = 1, \forall i$$

Which means x_i could approximately be reconstructed by $B\gamma_i$. With $d_i = \exp\left(\frac{\text{dist}(x_i, B)}{\sigma}\right)$ is the locality adaptor which gives lower weights to the basis vectors that are different from the input descriptor x_i and vice versa, this leads to the locality constraint.

$\text{dist}(x_i, B) = [\text{dist}(x_i, b_1), \dots, \text{dist}(x_i, b_M)]^T$, where $\text{dist}(x_i, b_j)$ is the Euclidian distance between x_i and b_j . σ is a parameter controlling the weight decay speed for the locality adaptor. \odot denotes the element-wise multiplication and $1^T \gamma_i = 1$ insures sparseness.

The distance regularization of LLC effectively performs feature selection: it selects local bases for each descriptor and form a local coordinate system, and in practice only those bases close to x_i in feature space have non-zero coefficients.

This suggests, a fast approximation of LLC can be developed by removing the regularization completely and instead using the K ($K < D < M$) nearest neighbors of x_i as a set of local bases B_i and solve a smaller linear system to get the code vectors: $\arg \min_{\tilde{\gamma}} \sum_{i=1}^N \|x_i - B_i \tilde{\gamma}_i\|^2 \text{ s.t. } 1^T \tilde{\gamma}_i = 1, \forall i$ which reduces the computation complexity from $O(M^2)$ to $O(M + K^2)$ where $K \ll M$.

C. Aggregation of the encoded local descriptors to obtain the final image descriptor

Now that we have our encoded descriptors, an image is still represented by too many code vectors (encoded descriptors), so we’ll need to pool these vectors in order to describe the image with one final descriptor. In the standard BoF pipeline we perform hard assignment for coding and then directly start aggregating the encoded local descriptors with sum pooling. In our work, we extend the old version of BoF by first adding the spatial pyramid scheme to capture the spatial information, we then proceed with the aggregation using either sum pooling or max pooling; the latter is a newly used approach that appears to be giving promising results.

1) *The spatial pyramid matching (SPM)* [50]: aims to incorporate the global spatial layout into the image representation. Inspired from the original work of [16], the spatial pyramid creates a pyramid of regular grids with increasingly finer cells. Each spatial cell will give us a histogram of visual words and the concatenation of all the histograms in a weighted manner gives us the final descriptor.

In the original work they construct a three-level pyramid. In a level $i \in \{0, \dots, n\}$ of the pyramid, the image is divided into 2^i spatial cells. Each cell has a histogram of visual words, and the image is represented with $\sum_{i=0}^n 2^i \times M$ vector, where M is the length of the codebook. This is a form of spatial pooling when we add the spatial position by assigning the histogram of each level with a weight, then we aggregate all the weighted histograms to get the final descriptor.

2) *Pooling*: Given the coding coefficient γ of each local descriptor in an image, a pooling operation is often used to obtain an image level representation p where $p \in R^M$ with M the total number of visual words.

Sum or average pooling [50] With sum-pooling, the i^{th} component of p is $p_j = \sum_{i=1}^l \gamma_{ij}$, where l is the total number of local features in an image. Dividing p_j by l gives us sum or average pooling. The histogram of number of occurrences of visual words in an image is essentially obtained by applying sum pooling to hard-assignment coding results.

Max-pooling [34] The i^{th} component of p is defined as $p_j = \max_i \gamma_{ij}$, where $i=1, \dots, l$. For each codeword we select the feature with the maximum coefficient γ . Max-pooling often gives better classification than average pooling. When used with hard-assignment coding scheme, max-pooling gives a binary histogram, indicating the presence or absence of each visual word in an image.

D. The classification phase

In image categorization, the goal is to automatically annotate images with predefined categories. Once the image descriptors are extracted, image labels are predicted using a set of classifiers.

SVM classification: In our work we rely on Support Vector Machine (SVM) classifiers to carry out the classification task. SVM is a discriminative classifier formally defined by a separating hyper plane. In other words, given labeled training data (supervised learning), the algorithm outputs an optimal hyper plane which categorizes new examples. Given a labeled set $L = \{x_i, y_i\}_{i=1}^l$, where $x \in R^d$ and $y \in \{-1, +1\}$. A linear maximal margin classifier $f(x) = w^T x + b$ can be found by solving:

$$\min_{w, b, \xi} \sum_{i=1}^l \xi_i + \lambda \|w\|^2 \text{ s.t. } y_i (w^T x + b) \geq 1 - \xi_i, \xi_i \geq 0, \forall i, i = 1, \dots, l. \text{ For a hyper plane } w \in R^d, \text{ and its offset } b \in R.$$

The extent of the SVM classifier lies in its easy extension to the nonlinear case [51]. Highly nonlinear nature of data can be taken into account by using the kernel trick such that the hyper plane is found in a feature space induced by an adapted kernel function $K(x_i, x_j) = \langle \Phi(x_i), \Phi(x_j) \rangle$. SVM handles nonlinearly separable problems using kernel functions by

projecting the data points into a higher-dimensional feature space $x \rightarrow \Phi(x)$. The Multi-class problem can be discriminated by performing, several “one-versus-all” classifications.

It was empirically found that, to achieve good performance, traditional SPM has to use classifiers with nonlinear Mercer kernels, e.g., Chi-square kernel. However, the nonlinear classifier has to afford additional computational complexity, bearing $O(n^3)$ in training and $O(n)$ for testing in SVM, where n is the number of support vectors. This implies a poor scalability of the SPM approach for real applications [34, 35].

To improve the scalability, researchers opted for a nonlinear feature representations that work better with linear classifiers, e.g. the works of [52, 53, 34, 35] where the final representation generated by using these encodings, achieved state-of-the-art performances on several benchmarks with a linear SVM classifier. In our experiments, we use a one-versus-all linear SVM classifier, implemented by LibSVM toolbox [55] and the optimized values of the parameters of SVM models are given by cross validation.

IV. EXPERIMENTS AND RESULTS

In this section, we first give the detailed implementation of our experiment then we evaluate the different methods introduced in Section 3 in each level of the BoF pipeline.

A. Implementation

1) *Fetching the data:* We follow the experimental setup of [16, 34], where the training set contains 30 images and the testing set contains 50 images per class. We create our ground truth matrix that will contain the labels of the testing images.

2) Training phase:

a) *Detection and extraction of features:* the local descriptors used are DSIFT or SURF that will be combined sometimes with the global descriptor GIST for experimental purposes.

- **DSIFT:** For the patch sampling process we set the spacing of the dense grid to 6 pixels, and the sampling window is set to 16*16 pixels.
- **SURF:** We detect SURF keypoints using a threshold set to 55 as it gives the best results for our datasets, and we select the strongest features. We then extract the SURF descriptors.
- **GIST:** We convolve the image with 32 Gabor filters at 4 scales, 8 orientations, producing 32 feature maps of the same size of the input image. Then, we divide each feature map into 16 regions (by a 4x4 grid), and average the feature values within each region. Finally, we concatenate the 16 averaged values of all 32 feature maps, resulting in a 512-dimension GIST descriptor.

b) *Codebook computation:* we use K-means algorithm on the descriptors extracted in the previous step in order to create our vocabulary; we chose a maximum number of iterations for K-means equal to 50 to generate 1024 visual words as suggested in the work of [55].

c) *Coding:* In this phase the descriptors are represented by a code vector using the codebook generated in the previous step. Here, we used 3 encoding algorithms, the standard hard assignment NN, KNN with a Gaussian kernel measure and LLC known for its good performance.

- **NN:** To represent our training and testing images as a histogram of visual words, for each image we will densely sample the descriptors and simply count how many fall into each cluster according to our visual word vocabulary. This is done by finding the nearest neighbor k-means centroid for every feature.
- **KNN:** K=5 Neighbors are used, the kernel Gaussian function $K(x, x_i) = \exp\{-d^2(x, x_i)/\sigma^2\}$ is then calculated for each visual word using the Euclidian distance d . We used a Gaussian kernel and chose $\sigma=500$ since it gave the best results according to our experiments.
- **LLC:** Using the Approximated Locality-constraint Linear Coding requires setting the number of neighbor visual words M considered for each encoded descriptor. This is set to K=5 as suggested in the original work [35]. As for The regularization constant λ in the computation of the projections is set to 10^{-4} .

d) *Aggregation to get the final descriptor:* Once we get all our code vectors, we need to aggregate them in order to get the final image descriptor.

- **Spatial pyramid:** The spatial pyramid is set to 3 levels so the image is divided increasingly into 2^i spatial cells with $i = \{0, 1, 2\}$.
- **Pooling:** we compared the two known methods average and max pooling. For sum-pooling, we'll average all the code vectors to create the final descriptor and for max-pooling we select the feature with the maximum coefficient.

e) *Learning:* once the code vectors are ready, we get our training matrix with the training labels that will be used by the SVM classifier in order to create the SVM model and generate the primal variable w of linear SVM.

3) *Testing phase:* in this phase we follow the same steps (from a to d) explained earlier in the training phase but using the test images instead of the training images.

- **Classification:** we classify the resulting code vectors using SVM, which contains the code vectors of the training set, their labels, the SVM model and the code vectors of the testing set. The classifier will predict the label of each image in the testing set and put it in a matrix.

4) *Performance and accuracy:* Multi-class classification is done with the trained SVM that learned to separate each class from the rest; a test image is assigned the label of the classifier with the highest response.

We Use the leave-one-out cross-validation framework to tune the parameters. We perform a 5-fold cross-validation where we train on 4 folds and we test on the remaining fold,

one at-a-time, and then the average accuracy is reported. Based on the accuracy we select the best parameters values.

To compute the classification accuracy ((number of well classified images/ number of images) *100) we compare our ground truth labels with the predicted ones and build the confusion matrix for our system figure 4.

B. Experiments and results

The results obtained were run on 30 training and 50 testing examples and a 1024 sized codeword dictionary. We tested the setup on the standard workstation (Xeon X5650 2.67 GHz, 6 core (12 CPU), 32 GB ram) using Matlab. We added the computing time column (in seconds) to evaluate the computational cost of the methods when running the offline process part of the image classification.

The results section is composed of three main sets of experiments, we start with the first phase of the BoF which is the feature extraction, so we compare the results obtained using different descriptors with a standard version of the pipeline which uses NN for coding, no spatial pyramid and simple sum pooling.

We then compare the different encoding methods one can use in the encoding phase of the BoF pipeline; we combined these methods with the different descriptors from the last setup.

At last we compare the pooling phase of the BoF pipeline, here we added the spatial pyramid scheme to the last setup and combined it with the different pooling methods.

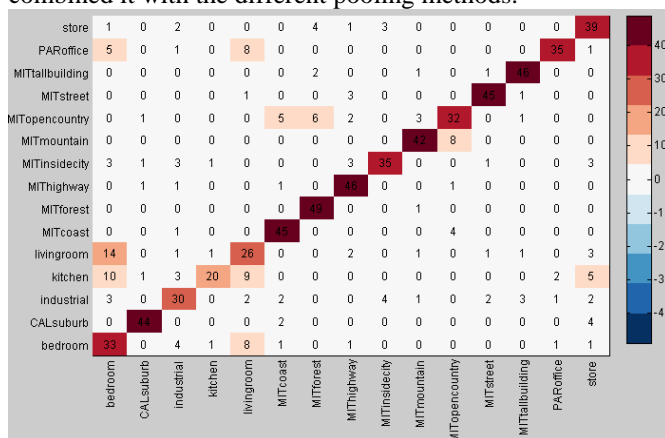


Fig. 4. Confusion matrix for 15 scene categories dataset resulting from our BOF pipeline, using SIFT+GIST as descriptors, LLC for encoding, SPM and max pooling. Along the diagonal we find the number of well-classified test images out of 50. The entry located in the i^{th} row, j^{th} column represents the number of images of class i being misclassified to class j

1) 15 scene categories:

Our first dataset is composed of fifteen scene categories which were gradually built. The initial 8 classes were collected by Oliva and Torralba [26], and then 5 categories were added by Fei-Fei and Perona [56]; finally, 2 additional

categories were introduced by Lazebnik et al. [50]. Each category has 200 to 400 images all of them in jpeg format, and the average image size is 300×250 pixels. This dataset contains a wide range of outdoor and indoor scene environments. A lot of works were tested on this dataset; most of them focus on dictionary learning, quantization and classification methods and use spatial pyramid matching [50]. It is widely recognized that whole spatial layout information is effective on this dataset. The experimental results are shown in the tables forthcoming.

2) Indoor dataset:

Our second dataset of experiments is the Indoor dataset [57]. It contains 67 Indoor categories, and a total of 15620 images in different resolutions. The number of images varies across categories, but there are at least 100 images per category. All images are in jpeg format. This database is known for being one of the most challenging open problems in high level vision. Most scene recognition models that give good results for outdoor scenes perform poorly in the indoor domain. The main difficulty is that while some indoor scenes can be well characterized by global spatial properties, others are better described by the objects they contain. Thus, to address the indoor scenes recognition problem we need a system that can exploit local and global discriminative information.

3) Descriptor comparison:

We know that in BoF model, keypoints are extracted from the grey level images and the local descriptors don't contain color information. Furthermore, BoF model captures only the local information, losing the overall distribution of visual information. The performance could be enhanced if we can combine global features and local descriptors together.

Both DSIFT/SURF and GIST feature representations describe different features in their target images. So we combine global GIST features with local DSIFT or SURF features in order to see if these complementary features could be used together to increase the image classification accuracy. Table I shows the accuracy results for the different image descriptors. We used the simple vector quantization scheme with NN for coding, no spatial pyramid and simple sum pooling in order to compare the descriptors.

TABLE I. ACCURACY ACHIEVED WITH THE DIFFERENT FEATURE DESCRIPTORS

	Image descriptor	Accuracy	Time
15 Scene Categories	DSIFT	63.34 %	429.98s
	SURF	48.54 %	291.53s
	DSIFT+GIST	66.94 %	831.07s
	SURF+GIST	65.07 %	672.64s
Indoor	DSIFT	32.18 %	14810.57s
	SURF	26.39 %	7455.70s
	DSIFT+GIST	30.51 %	15495.33s
	SURF+GIST	29.89 %	10725.74s

TABLE II. ACCURACY ACHIEVED WITH THE DIFFERENT ENCODING METHODS

15 scene categories	Image descriptor	Encoding	Accuracy	Time	Indoor	Image descriptor	Encoding	Accuracy	Time
	DSIFT	NN	63.34 %	429.98s		DSIFT	NN	32.18 %	14810.57s
DSIFT	KNN	64.67 %	475.21s	DSIFT	KNN	32.27%	15884.10s		
DSIFT	LLC	64.67 %	557.40s	DSIFT	LLC	32.03 %	18903.22s		
DSIFT+GIST	NN	66.94 %	831.07s	DSIFT+GIST	NN	30.51 %	15495.33s		
DSIFT+GIST	KNN	65.60 %	870.64s	DSIFT+GIST	KNN	30.36 %	17293.81s		
DSIFT+GIST	LLC	66.14 %	959.64s	DSIFT+GIST	LLC	30.30 %	20701.02s		
SURF	NN	48.54 %	291.53s	SURF	NN	26.39 %	7455.70s		
SURF	KNN	49.34 %	319.36 s	SURF	KNN	22.12 %	7688.52s		
SURF	LLC	50.40 %	411.95s	SURF	LLC	26.18 %	9699.77s		
SURF+GIST	NN	65.07%	672.64s	SURF+GIST	NN	29.89 %	10725.74s		
SURF+GIST	KNN	64.27%	699.48s	SURF+GIST	KNN	29.98 %	11553.35s		
SURF+GIST	LLC	64.67 %	791.55s	SURF+GIST	LLC	29.89 %	12429.21s		

TABLE III. ACCURACY ACHIEVED USING SPATIAL PYRAMID AND DIFFERENT POOLING METHODS

15 scene categories	Image descriptor	Encoding	SPM	Pooling	Accuracy	Time	Indoor	Image descriptor	Encoding	SPM	Pooling	Accuracy	Time
	DSIFT	NN	Yes	Sum	64.94 %	467.10s		DSIFT	NN	Yes	Sum	32.45 %	17923.76s
DSIFT	NN	Yes	Max	67.47 %	469.76s	DSIFT	NN	Yes	Max	37.20 %	17784.88s		
DSIFT	KNN	Yes	Sum	67.87%	611.88s	DSIFT	KNN	Yes	Sum	30.60 %	22740.40s		
DSIFT	KNN	Yes	Max	69.87 %	618.42s	DSIFT	KNN	Yes	Max	36.45 %	22297.42s		
DSIFT	LLC	Yes	Sum	65.07 %	694.75s	DSIFT	LLC	Yes	Sum	33.32 %	26032.58s		
DSIFT	LLC	Yes	Max	70.67 %	661.22s	DSIFT	LLC	Yes	Max	37.35 %	25602.24s		
DSIFT+GIST	NN	Yes	Sum	64.80 %	855.31s	DSIFT+GIST	NN	Yes	Sum	29.71 %	18118.85s		
DSIFT+GIST	NN	Yes	Max	73.74 %	878.74s	DSIFT+GIST	NN	Yes	Max	38.99 %	18930.11s		
DSIFT+GIST	KNN	Yes	Sum	64.67 %	944.66s	DSIFT+GIST	KNN	Yes	Sum	29.47 %	22971.09s		
DSIFT+GIST	KNN	Yes	Max	74.14 %	1004.65s	DSIFT+GIST	KNN	Yes	Max	38.42 %	27083.96s		
DSIFT+GIST	LLC	Yes	Sum	64.67 %	1031.66s	DSIFT+GIST	LLC	Yes	Sum	29.47 %	25801.05s		
DSIFT+GIST	LLC	Yes	Max	75.60 %	1078.20s	DSIFT+GIST	LLC	Yes	Max	39.23 %	26664.71s		
SURF	NN	Yes	Sum	46.27 %	383.22s	SURF	NN	Yes	Sum	23.44 %	12306.90s		
SURF	NN	Yes	Max	56.40 %	382.35s	SURF	NN	Yes	Max	31.41 %	12498.50s		
SURF	KNN	Yes	Sum	44.80 %	645.51s	SURF	KNN	Yes	Sum	15.05 %	20238.74s		
SURF	KNN	Yes	Max	60.94 %	632.27s	SURF	KNN	Yes	Max	29.98 %	20160.84s		
SURF	LLC	Yes	Sum	43.34 %	735.12s	SURF	LLC	Yes	Sum	17.50 %	22281.54s		
SURF	LLC	Yes	Max	58.14 %	690.94s	SURF	LLC	Yes	Max	31.41 %	20763.74s		
SURF+GIST	NN	Yes	Sum	64.14 %	728.63s	SURF+GIST	NN	Yes	Sum	29.50 %	14263.72s		
SURF+GIST	NN	Yes	Max	69.34 %	799.80s	SURF+GIST	NN	Yes	Max	36.81 %	15183.18s		
SURF+GIST	KNN	Yes	Sum	64.54 %	894.66s	SURF+GIST	KNN	Yes	Sum	29.50 %	21631.45s		
SURF+GIST	KNN	Yes	Max	70.40 %	1015.97s	SURF+GIST	KNN	Yes	Max	36.93 %	22830.13s		
SURF+GIST	LLC	Yes	Sum	64.40 %	981.35s	SURF+GIST	LLC	Yes	Sum	29.50 %	24736.23s		
SURF+GIST	LLC	Yes	Max	69.74 %	1067.70s	SURF+GIST	LLC	Yes	Max	36.99 %	24167.81s		

As we can see from table1 DSIFT descriptor gives better results than SURF descriptor, although it is much more time consuming than the latter.

We then combined the GIST feature vectors with the encoding of DSIFT/SURF descriptors by appending the feature vectors together. The best accuracy rate was achieved by combining DSIFT and GIST for the fifteen dataset with 66.94% and 32.18% using DSIFT solely for the Indoor dataset. DSIFT is slower than SURF but gives better results. When we combined the local descriptors with GIST the time of calculations increased but so did the accuracy. GIST and DSIFT are slow but captures more information.

4) Encoding comparison:

To compare the encoding phase of the pipeline we kept the same codebook with 1024 entries and used the different

descriptors with our encoding methods (NN, KNN, and LLC). Table 2 shows the accuracy results for using different coding schemes without spatial pyramid and with sum pooling.

We see that the results are more less the same for the different encoding methods, where there is a minor difference between the accuracies when using the same image descriptor. However, there is an important difference in numbers when the encoding methods are combined with different descriptors. We can see that the best accuracy was achieved for the fifteen dataset by DSIFT+GIST descriptor combined with NN for encoding, and with DSIFT descriptor combined with KNN for the Indoor dataset. As for the computational cost, it is distinct from the results that the most expensive encoding method is LLC followed by KNN then lastly the standard NN for vector quantization.

5) Spatial pyramid with pooling comparison:

For the last phase we compare the different pooling methods. Table 3 shows the accuracy results for combining the different descriptors and coding methods with spatial pyramid then aggregating with either max or sum pooling. We notice that the accuracy increased immediately when we added the spatial pyramid scheme and max pooling but so did the computational time. These two recent approaches enabled our setup to achieve the best performance with 75.60% for the fifteen dataset and 39.23% for the Indoor dataset, both using DSIFT+GIST as a descriptor and LLC for encoding.

For the performance we can surely assert that max pooling do improve the overall classification over the widely used sum pooling method, but we can't say much about the difference in computational time between the two approaches.

C. Discussion

In order to evaluate the new approaches in the bag of features model, we compared them to the previous standard methods with respect to the same codebook and many other constraints as mentioned in the implementation details. The paper gives the results using two popular datasets and a linear SVM classification.

The local descriptor DSIFT is the better choice for a good feature extraction and when combined with the global descriptor GIST it captures more information, resulting in an even better performance.

The different encoding methods NN, KNN, LLC gave more less the same results when combined with average pooling and no spatial pyramid, but when combined with spatial pyramid and max pooling LLC gave the best results. The important fact is that spatial pyramid matching and max pooling did significantly improve the performance of all the encoding methods. The improvement of the encoding methods accuracies is about 5 to 10%. This shows that spatial information is important for image classification.

Including both local and global descriptors for the first step of the BoF model, enhanced the performance. Local context and global quantized information were combined to make conventional features more discriminative. Using those resulting robust features we built our vocabulary with K-means algorithm which represents the online process part of the BoF model.

For the encoding step, we used 3 different approaches the hard assignment NN algorithm, a soft assignment distance-based KNN algorithm and a soft assignment reconstruction method LLC. Vector quantization with NN is fast but gives poor results; LLC is much more efficient since the assignment is optimized as to minimize the reconstruction error unlike the purely distance-based assignment with KNN.

The computational efficiency of spatial pyramid matching combined with max pooling for the last step of the BoF model yielded the highest classification rates on challenging data for instance the Indoor dataset.

The importance of this work lies in the details about each level of the pipeline that includes the methods used and their

efficiency. This results in an apprehension of each step, which leads us to a better understanding of the BoF model as a whole.

Our setup enabled us to achieve the best performance using DSIFT and GIST as descriptors since they capture more information, then LLC for encoding which was combined with spatial pyramid matching and max pooling. The classification was then measured on the basis of the BoF model by using a linear SVM classifier. These changes in each level of the pipeline resulted in an enhancement of the results over the basic bag of features model. The new approaches maybe a bit costly but the gained efficiency is worth the added computing time.

V. CONCLUSION

This paper presents an overall insight on the bag of features model. We proposed an image classification pipeline which we used to compare and evaluate the standard bag features methods with the new ones and it appears that these changes in each step of the pipeline did indeed affect the performance.

In fact, we came to the conclusion that the right combination of the methods used in each level of the pipeline as we did in our setup leads to better results. The first amelioration took place in the first step of the pipeline where we combined some descriptors used for features extraction that captures local and global context information. Actually this improvement is due to the global GIST descriptor which adds a spatial envelope to the local features and generates a more complete description of the image.

In the next step we compared different encoding methods, first NN representing the hard assignment vector quantization, then KNN for a distance-based soft assignment and lastly LLC as a reconstruction-based soft assignment. The resulting code vectors were finally used in the last step of the pipeline where another improvement took place, when we added a spatial pyramid layer for when we want to aggregate our code vectors combined with sum pooling as a basic aggregation method or max pooling as a new approach. The SPM max-pooling combination clearly boosted the performance results.

At last, we can say that each step of the pipeline holds part of the final results which means that selecting the methods to use is the most crucial part of the BoF model. In fact, the good performance we achieved is mainly due to the recent approaches we introduced in each step of the pipeline which increased the classification accuracy over the standard bag of features baseline.

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Maximally Distant Codes Allocation Using Chemical Reaction Optimization with Enhanced Exploration

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Abstract—Error correcting codes, also known as error controlling codes, are sets of codes with redundancy that provides for error detection and correction, for fault tolerant operations like data transmission over noisy channels or data retention using storage media with possible physical defects. The challenge is to find a set of m codes out of 2^n available n -bit combinations, such that the aggregate hamming distance among those codewords and/or the minimum distance is maximized. Due to the prohibitively large solution spaces of practically sized problems, greedy algorithms are used to generate quick and dirty solutions. However, modern evolutionary search techniques like genetic algorithms, swarm particles, gravitational search, and others, offer more feasible solutions, yielding near optimal solutions in exchange for some computational time. The Chemical Reaction Optimization (CRO), which is inspired by the molecular reactions towards a minimal energy state, emerged recently as an efficient optimization technique. However, like the other techniques, its internal dynamics are hard to control towards convergence, yielding poor performance in many situations. In this research, we proposed an enhanced exploration strategy to overcome this problem, and compared it with the standard threshold based exploration strategy in solving the maximally distant codes allocation problem. Test results showed that the enhancement provided better performance on most metrics.

Keywords—Evolutionary Algorithms; Chemical Reaction Optimization; Maximally Distant Codes; Binary Knapsack Problem; Fault Tolerance

I. INTRODUCTION

The goal of this research is to allocate sets of codes that maximize mutual distances for use as error control codes. This is of great significance for many applications, like data retrieval and communication. Finding optimal or even near optimal solutions for practically sized problems using brute force search is a challenge due to the prohibitively large solution spaces. The problem $A(n, m, d)$ is about locating a set of m codewords with n bits that are at least d bit apart. The solution space of the small instance (7, 16, 3) is at least 1020, while a slightly larger instance like (8, 16, 3) has more than 1024 solutions to explore. The solution space of a relatively small sized problem, like (10, 64, 3), exceeds 480 Googol (4.8×10^{102}), which is large enough to rule out any exact search algorithm even using supercomputing power.

According to the packing sphere theorem, aka Hamming bound, if S is a code of strings of n bits with $d(S) = 2k+1$, where k is the radius (maximum number of bits that can be

corrected) and d is the distance, then the cardinality $|S|$, or size set, is defined as:

$$|S| \leq \frac{2^n}{\sum_{i=0}^k C(n, i)} \quad (1)$$

Using (1), loose bounds on the sets cardinality are shown in Table I for selected values of d and k ; d errors detection and k errors correction. If the inequality in (1) does not hold then the code S does not exist while if it holds, there is no assurance of existence of such a code, and this is why those bounds are not quite useful and better ones are needed, later sections will provide tighter ones as research outcomes. Later, tighter bounds will be presented as reported by researchers.

TABLE I. CODEWORDS VERSUS MINIMAL DISTANCE AND RADIUS

n	Error Correction / Detection		
	$k = 1, d = 3$	$k = 2, d = 5$	$k = 3, d = 7$
8	28	6	2
10	93	18	5
12	315	51	13
14	1092	154	34
16	3855	478	94
18	13797	1524	265

Evolutionary optimization algorithms offer optimal or near optimal solutions with affordable computational effort; time and resources. Such algorithms have been used in solving complex engineering problems with varying quality and time tradeoffs. The CRO algorithm has emerged recently as an adaptive method to explore such large spaces with ordinary computational resources. In a previous work, we adapted the CRO algorithm to the maximally distant codes allocation problem by mapping it to the well-known binary Knapsack problem. The results were acceptable but with poor exploitation and exploration balance in some situations, as the algorithm spent more time exploring the space even with tight exploration reactions conditions, yielding less fruitful computations. We extended the research to adjust the exploration reactions through better control; randomized and quality dependent rather than threshold only dependent.

II. LITERATURE REVIEW

Due to significance, error correcting code allocation problem has been addressed for a long time using various methods. Evolutionary paradigms were applied with varying degrees of success; for example, the work in [1] described some initial experiments of Genetic Algorithms (GAs) to discover maximal distance codes, and discussed the potential advantage of genetic algorithms in this problem domain, others like [2], compared the performance of many evolutionary algorithms with local search and greedy methods in solving the error correcting codes discovery problem, and concluded that the GAs were the best of all other algorithms in general, with even more performance advantage as the cases got harder, while [3] tackled the error correcting codes allocation with two related techniques, Memetic Algorithm (MA) and Scatter Search (SS), by investigating the instantiation of those techniques for error correction codes design, the local improvement strategy and the combination method in specific, and reported that those techniques could outperform previous approaches. In [4], both GA and Genetic Programming (GP) were examined on three different binary error correcting codes problems to generate optimal sets of codes, and devised a new chromosome representation, claiming benefits in certain conditions.

Power efficient design was addressed in many places. For example, a GA with integrated symbiotic mechanism was proposed in [5], to locate codes that provide single error correction and double error detection. The work formulated the selection of the parity check matrix into a collection of individual and specialized optimization problems. Another approach in [6] targeted the power consumption reduction in single error correcting, double error detecting checker circuits in memory, using the degrees of freedom in selecting the parity check matrix of the error correcting code, by using Simulated Annealing (SA) and GA to solve non-linear power optimization problems. Tests on actual memory traces of benchmarks indicated that considering power along with area and delay when selecting the parity check matrix could result in significant power reductions. Closely related work was reported in [7], which investigated the use of different evolutionary algorithms to improve the lower bounds for given parameters by relating this problem to the well-known Maximum Clique Problem. As in most of the evolutionary methods, local search mechanisms were integrated, like the one in [8], which presented a new local search algorithm for the error correcting codes problem called the Repulsion Algorithm (RA), and used it with a parallel GA to solve the problem, and compared it against a pure parallel GA. They achieved important improvement with the inclusion of the RA. In a related context, the authors of [9] resolved the question of the utility of the crossover operator in earlier studies on optimizing DNA error correcting codes, where the crossover operator in question was found to be substantially counterproductive and the majority of the crossover events produced results that violated the minimum distance constraints required for error correction.

Recently, few new evolutionary paradigms emerged, most importantly the one proposed in [10], called Chemical Reaction Optimization (CRO), to solve optimization problems by

mimicking the interactions of molecules in a chemical reaction to reach a low energy stable state. The performance of the proposed algorithm was tested using three nondeterministic polynomial time hard combinatorial optimization problems; two traditional benchmark problems and a real-world problem, reporting competitive results compared with the existing successful metaheuristics. The authors employed this technique in [11] to the population transition problem, to maximize the probability of universal streaming by manipulating population transition probability matrix, and reported better performance than many commonly used methods for controlling population transition in many practical live streaming systems. Researchers have applied this technique to many intractable problems in various fields, like solving the grid scheduling problem in [12], which compared it with four generally acknowledged methods, and reported superior performance, and scheduling Directed Acyclic Graph (DAG) jobs in heterogeneous computing systems, and a Double Molecular Structure-based CRO (DMSCRO) method as in [13], to encode the execution order of the tasks in a DAG job, and the task-to-computing-node mapping, along with four elementary operations, and a fitness function suitable for DAG scheduling, and verified the effectiveness over a large set of randomly generated and real-world problems graphs, and testing the performance of CRO on three nondeterministic polynomial time hard combinatorial optimization problems reported in [10], claiming that it was very competitive with the existing metaheuristic, and outperformed them in some cases, like the real-world problem.

Among the other applications, [14] used it in developing an allocation algorithm to study three utility functions for utilization and fairness with hardware constraints, and showed that it outperformed the others by a good margin, and [15] used the CRO in allocating sets of maximally distant codes for a certain set of parameters, and reported good results in a relatively short time for small instances, and some difficulties in larger instances due to exploration reactions inefficiency, while [16] proposed an Adaptive CRO (ACRO) to alleviate the effort in tuning parameters, reducing the number of optimization parameters in canonical CRO along with an adaptive scheme to evolve them. They performed simulations on a widely-used benchmark of continuous problems claiming superior performance over canonical CRO. A multiobjective variant of CRO was reported in [17], called nondominated sorting CRO, and meant to exploit chemical reaction optimization features in tackling problems involving multiple criteria, making the multiobjective algorithm efficient from a computational cost viewpoint. Benchmarks test against reported good convergence.

Some novel approaches built on the CRO, for example the one in [18] proposed for training higher order neural networks, with two modifications; fixing the population size and using greedy reversible action after the regular actions. Compared to the basic CRO algorithm and two variants, using well known neural networks benchmarks, results reported significant statistical improvements. Another trend was to use hybrids. For example, [19] proposed a CRO with greedy strategy algorithm (CROG) to solve the binary Knapsack problem, by integrating a greedy strategy and random selection to repair the infeasible

solutions. The experimental results have shown superior performance compared to the standard GA, the Ant Colony Optimization (ACO), and the Quantum Inspired Evolution (QIE). The Orthogonal CRO (OCRO) is yet another example of hybridization reported in [2], adding quantization orthogonal crossover for global search. Tests on a set of several benchmark functions reported faster convergence speed to close to optimal solutions, especially for high-dimensional functions. Another hybrid was built on a local search to solve the Travelling Salesman problem (TSP) as in [21], by integrating the Lin-Kernighan (LK) local search, claiming better tradeoff between the exploration abilities of CRO and the exploitation abilities of LK local searcher, resulting in more efficient algorithm. The hybrid reported in [22], called Hybrid CRO (HCRO), was developed to solve task scheduling problems, by integrating a selection strategy with standard CRO. Both simulation and real-life experiments proved that the HCRO algorithm task scheduling was much better than the existing algorithms in terms of makespan and speed of convergence. Another hybrid was proposed in [23], by combining the two local search operators in the CRO with global search ability for global optimum, incorporating concepts from the CRO and the Particle Swarm Optimization (PSO). Tests on a set of twenty three benchmark functions have shown that this CRO and PSO hybrid could outperform the CRO in most of the experiments. Moreover, a novel computational method was reported in [24] as more robust and with less parameters than that used in the literature, called the Artificial Chemical Reaction Optimization Algorithm (ACROA), and applied it to multiple-sequence alignment and data mining.

Like any other evolutionary algorithm, parameters setting is one of major issues in the CRO implementation, and a study of their dynamics was reported in [25], covering various parameters. Numerical experiments for two test functions in the category of non-linear constrained optimization problems reported in the literature are carried out, indicating at par performance compared with other optimization methods.

Clearly, the CRO has a potential as an optimization method with universal applicability, but like all other algorithms, its parameters selection may lead to improper convergence sometimes. According to [26], when averaged over all the involved objective functions, all search algorithms behave exactly the same in terms of performance. All these algorithms are successful in solving different kinds of optimization problems. Hence an algorithm, compared to others may show equal performance on the average, but could outperform many others when matched to the right problem type.

III. MAXIMALLY DISTANT CODES

Data transmission and retention reliability requires the use of codes with error tolerance capability. Finding such sets of codes using exact search strategies requires hefty computational power even for a moderately sized instance. The objective of the search may vary; from finding a set of n -bit maximally distant codewords with predefined set size, to finding the largest set of n -bit codewords with some predefined minimum hamming distance. There can be many ways to map this problem to evolutionary algorithms, like a matrix

representing a set of codewords as a candidate solution, or a binary vector representing m codes n bits each. Both of those mappings require a good deal of time to validate the solutions after reactions. A good mapping must provide a balanced memory and processing cost during the search to achieve acceptable utilization of resources. The maximally distant discovery problem maps well to the binary Knapsack problem with minimal validity test cost, and will be used in the two implementations; the standard and proposed version with enhanced exploration.

Also critical to the processing cost, and hence the search quality, is the ability of the mapping and the processing to offer diversification while producing valid solutions to avoid wasting computational resources. The maximally distant codes allocation problem lends itself easily to the binary Knapsack problem, where the ones and zeros in a vector representing a candidate solution indicate inclusion and exclusion of codewords of the corresponding positions. The problem is well suited to evolutionary search paradigms in general.

The theoretical bounds of $A(n, d)$ stated in Table I are practically loose, and over a thousand papers have been written describing methods to improve those bounds, and the results till recent dates are partially shown in Table II as stated in [27].

TABLE II. CODEWORDS VERSUS MINIMAL DISTANCE LOWER BOUNDS

n	Hamming Distance		
	$d = 3$	$d = 5$	$d = 7$
8	20	4	2
10	72	12	2
12	256	32	4
14	1024	128	16
16	2720 – 3276	256 – 340	36 – 37
18	10496 – 13104	1024 – 1280	128 – 142
20	36864 – 43688	2560 – 4096	512

IV. CHEMICAL REACTION OPTIMIZATION

The CRO algorithm starts with an initial set of randomly selected molecules, and iteratively applies one of four reactions until some stopping criteria is met. The exploitation reactions have an equal number of inputs and outputs and hence the population size remains fixed regardless of how often they are applied. On the contrary, the exploration reactions have no balance; they either decrease or increase the population size as they generate one out of two or two out of one, and unless they are equal in frequency, the population size grows to an unwanted limit or diminishes to one. Extremely large population size is a computational burden, while extremely small population size is inefficient in space exploration. Balancing the population size is important for a fruitful search.

To allocate sets of maximally distant codes, we need to maximize the minimum distance between any two codes in the set or the mean distance among all codes. Since the CRO is a

minimization strategy, codewords similarities will be used as a cost function for guiding the search. For n -bit codewords, the similarity is in the range 0 to $n-1$, and for m codewords, there exist $m(m-1)/2$ values of similarity related to all pairs. Using a composite cost function with a balancing factor $0 \leq \mu \leq 1$, we used the following cost function:

$$C = \frac{2\mu}{m(m-1)} \sum_{i=1}^m \sum_{j=i+1}^m S_{ij} + (1-\mu) \text{Max}(\text{Max}(S_{ij})) \quad (2)$$

Where:

S is the Similarity matrix of m^2 entries.

m is the number of Codewords.

The CRO design has two major components, which are molecules that represent solutions, and elementary reactions, necessary to traverse the solution space looking for an optimal or near optimal solution.

A. Molecules

Molecules represent possible solutions to the problem and their characteristics make one molecule distinguishable from another. Chemically, bonds form and break acquiring and releasing energy, respectively. This energy exchange with the surroundings abides by the first law of thermodynamics, which states that energy can neither be created nor destroyed. The major attributes that describe a molecule are:

- Structure, which represents the solution currently held by a molecule. In our case, the molecule has the form of a binary vector whose size is equal to 2^n for n -bit codewords.
- Potential Energy, which represents the energy corresponding to the structure, and it depicts the value of the objective function of the current molecular structure.
- Kinetic Energy, which represents the level of tolerance of a molecule to change to a less favorable structure, i.e., with higher potential energy.

B. Elementary Reactions

Elementary reactions are the means to explore various parts of the solution space and to exploit the most feasible neighborhoods. Reactions allow molecules to collide with each other or with the walls causing structural changes, making possible a product formation. There are two types.

- Exploitation

Exploitation is a process that tends to search for better solutions in the neighborhood of one. This is achieved through molecule collisions, either with the wall or with another molecule. Typically, molecule(s) go through subtle change hoping for better ones. Exploitation is carried out through two major reactions:

* Wall Collision

This is a unimolecular reaction in the form of a collision with a wall to bounce with some energy loss and a subtle structural deformation. The lost energy is stored in a central energy buffer and its amount is proportional to the sum of its

kinetic energy and its potential energy gain. The factor used is called kinetic energy loss rate. This reaction performs local search with ability of escaping local minima.

* Molecular Collision

This is a bimolecular reaction, and it occurs when two molecules collide and bounce back, it is similar to that of wall collision in that these reactions are also not vigorous. The result of this reaction is a subtle structural change of the reactants.

- Exploration

Exploration is a process that tends to search for better subspaces for later exploitation, and hence escaping local optima. This is achieved through major structural change to molecules which are not capable of getting better anymore. Exploration is carried out through two major reactions, decomposition and synthesis, when molecules representing solutions exhaust chances of exploitation. Description and conditions of those reactions are detailed below.

* Decomposition

This is a unimolecular reaction, but it differs from the wall collision in that it produces two molecules instead of one, with relatively significant structural change. When a molecule hits a wall, it breaks up into two molecules with new structures. Typically, the new structures reflect major transformation, and hence it is important for exploration when the local search in a certain neighborhood is not feasible anymore. Energy is transferred from the buffer to sustain its change to form the new molecules.

* Synthesis

This is a bimolecular reaction, and it also occurs when two molecules collide and combine to form a single molecule. Synthesis reactants involve a vigorous change in their molecular structures, which is also necessary to reach out for new subspaces.

The following sections detail the four elementary reactions using a simple example to find 8 maximally distant 5-bit codewords, The reactants are shown in red color while the products are in blue color.

1) Unimolecular Reactions

There are two unimolecular reactions; wall collision and decomposition. In wall collision, a molecule hits the wall and bounces back with some deformation; a minor structural change.

One way to carry out this is to pick an entry at random and scan from there up, with wrap around, looking for an opposite one, to flip both. Another way to do it is to select two random numbers in the range 0 to 31 to index the entries to flip under the condition that the selected entries are opposite, i.e. one of them is 1 and the other is 0. Fig. 1 shows an example of wall collision reaction using two random cuts.

```
0011000100001000100010000100001010010010
0011000100001000100110000010000000100100
```

Fig. 1. Wall Collision Reaction

The second unimolecular reaction is decomposition. In this case, a molecule breaks up to make two new molecules with major structural differences from the original ones. Typically, two copies of the molecule are modified by selecting a random number in the range 0 to 31 to make a cut in the two structures, then the upper part of one and the lower part of the other are shuffled, by circulating those strings a random number of bits. Fig. 2 shows an example, the gray cells are the fixed parts, and the rest of each is the result of circulation.

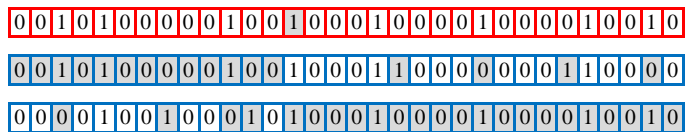


Fig. 2. Decomposition Reaction

2) *Bimolecular Reactions*

There are two reactions that involve two molecules as reactants. The first is synthesis, in which the two selected molecules are merged into one. A molecule is formed as a bitwise logical *xor* of the two molecules, then 1's or 0's are inverted at random to keep the numbers of 1's right; *m* entries. Fig. 3 shows an example. The gray cells are 1's, inverted 0's, to make the number of 1's equal 8 for a valid solution.

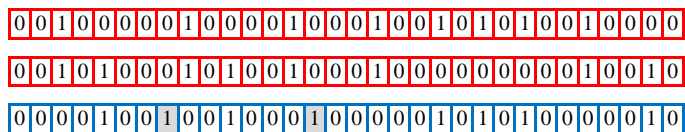


Fig. 3. Synthesis Reaction

The second bimolecular reaction is the molecular collision, where two molecules collide to bounce with subtle structural difference. In this case, a good deal of the properties of both are handed over to the resulting molecules. A random number in the range 0 to 31, is used to make cuts in the two reactants, then the upper parts are swapped, and if the solutions are invalid, having more or less 1's than *m*, the molecules are scanned to invert 1's or 0's properly. Fig. 4 shows an example, the gray cell represents a random cut for both reactants, the upper parts are then swapped and the molecules are scanned to invert 1's or 0's at random to keep the numbers of 1's right; *m* entries. The gray cells in the products refer to inverted entries.

Iterative application of reactions to molecules representing solutions generates better ones converging to an optimal or near optimal solution. It is quite important for convergence to carry out transformations that provide balanced exploration and exploitation.

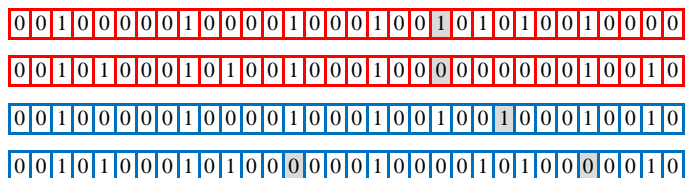


Fig. 4. Molecular Collision Reaction

Table III shows the significance of those reactions to the exploration and exploitation aspects.

The algorithm starts by setting initial parameters; population size as an initial number of molecules in the pool, molecular collision rate, kinetic energy loss rate, initial kinetic energy, buffer size, and decomposition and synthesis thresholds. Then, a set of initial molecules are generated at random, or through some vision for a more feasible initial set. A series of reactions, typically one at a time, are applied iteratively until the stopping criterion is satisfied, concluding with best possible solutions.

In each iteration, either one unimolecular or one bimolecular reaction is triggered, based on a preset rate called MoleColl.

TABLE III. ACTIONS AND SIGNIFICANCE

Reaction	Reactions: Types and Contributions		
	Type	Exploration	Exploitation
Wall Collision	Unimolecular	Minor	Major
Decomposition	Unimolecular	Major	Minor
Synthesis	Bimolecular	Major	Minor
Molecular Collision	Bimolecular	Minor	Major

V. BALANCED EXPLOITATION AND EXPLORATION

The exploitation and exploration balance of the standard implementation suffers most of the time, especially when used to solve large instances as reported by many researchers, resulting in low improvement rates after few thousands computations. It is quite hard to figure out how the search traverses the solution landscape, but improving the exploration reactions performance is necessary for convergence.

New search operators were found useful as stated in the literature. In this work, the performance was enhanced by relaxing the exploration reactions.

In the unimolecular path, decomposition reaction in the standard implementation takes place if the selected molecule satisfies the decomposition criterion, i.e., (number of hits – minimum hit number) > α , where α can be interpreted as the tolerance of duration for the molecule without obtaining any new local minimum solution. If so, the molecule will experience decomposition, otherwise it will go through a wall collision.

In the bimolecular path, synthesis reaction in the standard implementation takes place if the selected molecules, two or more, satisfy the synthesis criterion; the total kinetic energy is less than some preset minimum β . If it is satisfied by all the selected molecules, they combine through synthesis. Otherwise, they experience an molecular collision.

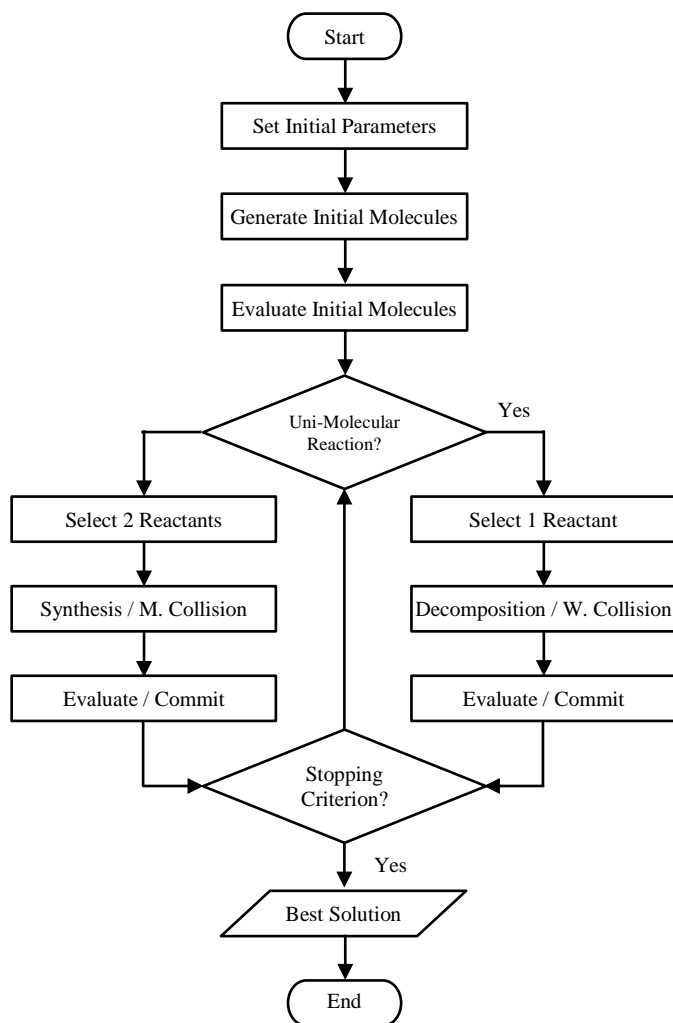


Fig. 5. Enhanced CRO Algorithm

The solutions generation function performs one of the four reactions, based on a preset rate *MoleColl*, shown in Fig. 5 as unimolecular or bimolecular decision. Otherwise, it performs unimolecular reactions. In the standard implementation, one of the two conditions DEC and SYN, standing for decomposition and synthesis discussed earlier, decide whether to carry out decomposition or synthesis, respectively, or one of the other reactions, wall collision or molecular collision, respectively, as shown in Fig. 6 and Fig. 7. The solution evolution is achieved by repetitively provoking one of the four elementary reactions. Convergence requires proper exploration, and triggering the decomposition and synthesis processes based on some static threshold may not be efficient.

The proposed enhancement has a different approach to trigger those processes. In the unimolecular path, the selected molecule is allowed to perform both decomposition and wall collision, but only one of them is committed and the other is abandoned. In the bimolecular path, the two selected molecules are allowed to perform both synthesis and molecular collision, and again only one of them is committed and the other is abandoned. Fig. 8 and Fig. 9 shows pseudocode for the enhanced reaction triggers.

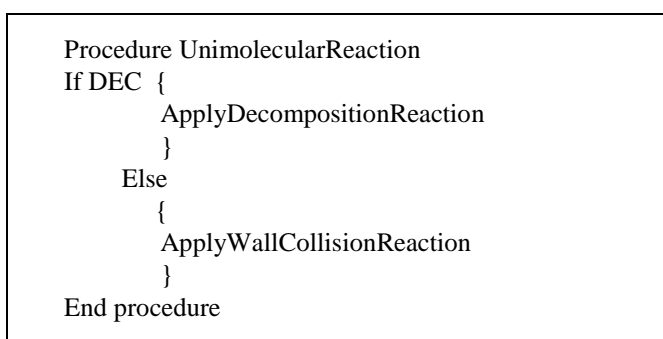


Fig. 6. Standard Unimolecular Reactions

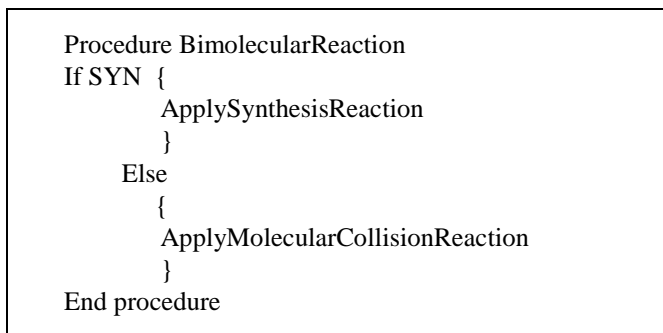


Fig. 7. Standard Bimolecular Reactions

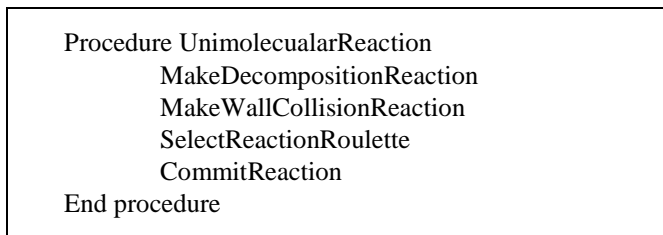


Fig. 8. Enhanced Unimolecular Reactions

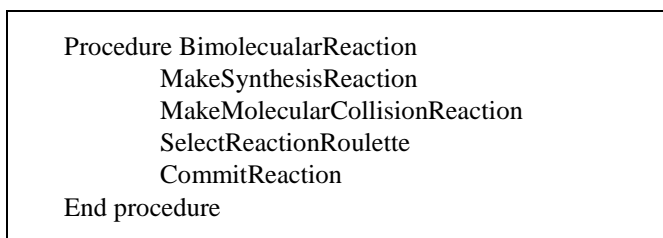


Fig. 9. Enhanced Bimolecular Reactions

In this reaction triggering scheme, the exploration reactions compete with exploitation reactions on two to one basis in the unimolecular types and on one to two bases in the bimolecular types. This means favoring the exploitation in the bimolecular reactions and favoring the exploration in the unimolecular reactions. The process is randomized but the fitness of the products plays a role in the triggering process. This scheme may lead to undesired increase in the population size, due to favoring the decomposition over synthesis, which can be resolved by a check against lower and upper bounds, to reinsert and drop molecules based on some criterion like the fitness value.

VI. EXPERIMENTS

In the initialization phase, PopSize, KELossRate, MoleColl, Buffer, InitialKE, α , and β were configured based on the literature recommendations, and the initial set of molecules is generated at random. Their initial potential energies are determined by their corresponding objective function values while their initial kinetic energies were set to InitialKE.

Table IV shows the standard algorithm performance reported earlier. The target was to locate sets of maximally distant codes of 8-bit strings, for three cost calculation scenarios: maximum similarity (corresponding to minimum distance), mean similarity (corresponding to mean distance) distance, and the two together with equal significance.

Table V shows the enhanced exploration method impact, using the same instances and initial population reported earlier. Although marginal, test results demonstrate better performance especially using the minimum distance metric.

TABLE IV. IMPACT OF THE WEIGHTING PARAMETER ON SEARCH

Set Size	Weighting Parameter					
	$\mu = 0.0$		$\mu = 0.5$		$\mu = 1.0$	
	Mean	Min	Mean	Min	Mean	Min
8	4.43	4	4.46	4	4.57	3
16	4.06	2	4.27	2	4.27	2
32	3.77	1	4.13	1	4.13	1
64	3.72	1	3.86	1	3.86	1

TABLE V. IMPACT OF THE WEIGHTING PARAMETER ON SEARCH

Set Size	Weighting Parameter					
	$\mu = 0.0$		$\mu = 0.5$		$\mu = 1.0$	
	Mean	Min	Mean	Min	Mean	Min
8	4.52	4	4.57	4	4.68	3
16	4.29	3	4.36	2	4.39	2
32	3.95	2	4.21	1	4.22	1
64	3.88	1	3.93	1	3.96	1

Table VI shows the performance of the algorithm in locating sets of 10-bit codewords with various set sizes. The proposed enhancement outperformed the standard, in both minimum distance and mean distance metrics. Many of those runs were repeated many times with the same initialization for fair comparison.

Table VII shows comparative performance of the two implementations in locating sets of 12-bit codewords with various set sizes. Table VIII shows comparative performance of the algorithms in finding three sets of fixed size and codeword length. Clearly the enhanced exploration is better in at least the mean distance metric if not in both.

TABLE VI. PERFORMANCE AGAINST SET SIZE (10-BIT CODEWORDS)

Set Size	Standard		Enhanced	
	Mean	Min	Mean	Min
8	5.34	4	5.68	5
16	5.13	3	5.26	4
32	5.09	3	5.12	4
64	5.08	2	5.07	3
128	5.08	2	5.08	2

TABLE VII. PERFORMANCE AGAINST SET SIZE (12-BIT CODEWORDS)

Set Size	Standard		Enhanced	
	Mean	Min	Mean	Min
32	6.42	4	6.68	4
64	6.26	3	6.42	4
128	6.15	3	6.28	3
256	6.12	3	6.14	3
512	6.10	2	6.11	2

TABLE VIII. PERFORMANCE AGAINST CODEWORD LENGTH / SET SIZE

Code Length / Set Size	Standard		Enhanced	
	Mean	Min	Mean	Min
10-bit / 16	5.11	4	5.36	5
12-bit / 64	6.13	3	6.41	4
14-bit / 256	7.24	3	7.58	3

TABLE IX. SUCCESS RATE COMPARISON

Code	Bound	Computations	Success Rate	
			Standard	Enhanced
8, 16, 3	28	16,000	12	14
8, 4, 5	6	4,000	11	13
10, 64, 3	93	64,000	11	14
10, 16, 5	18	16,000	10	11
12, 256, 3	315	256,000	11	13
12, 32, 5	51	32,000	9	14
14, 1024, 3	1092	1,024,000	8	10
14, 128, 3	154	128,000	8	9

The power of finding solutions is expressed as success rate as shown in Table IX, where the two implementations were run 20 times each with the objective of finding sets of 4 code lengths: 8, 10, 12 and 14, and two variations of each.

The number of computations was based on the space size. The proposed enhanced exploration implementation demonstrated a marginal improvement over the standard, in terms of the number of times it could locate a solution.

The advantage of the enhanced exploration is shown in Fig. 10 as progress plot, mean distance and minimum distance of the best solution over time. The standard implementation stabilized within few thousands of iterations, while the enhanced kept improving for longer time, indicating better exploration power. The enhanced strategy resulted in larger fluctuations of the mean distance over time due to the continuous generation of solutions in new subspaces, and hence better exploration with increased time budgets.

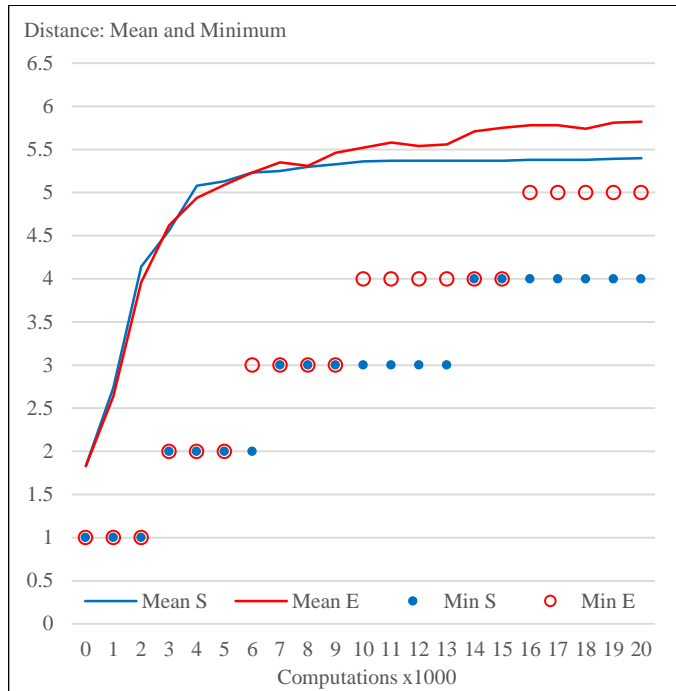


Fig. 10. Search Progress; Locating 12-bit Codeword Sets

Tests were run on i7 Intel quad core processor based desktops with 16 GB DRAM. Typical runs took from few minutes for small instances to few tens of minutes for the largest, carrying out tens of thousands computations, until improvement rate reaches 0.1%.

VII. CONCLUSION

Solving the Mapping the Maximally Distant Codes Allocation Problem by the Chemical Reaction Optimization algorithm has been reported in literature with some problems especially with larger instances. The issue is that the exploration reactions consume a good deal of fruitless computations. In this enhancement, we proposed a different approach to manage the exploration reactions. Instead of using preset threshold based triggering, causing unpredicted reactions rate, we involved the fitness of the selected molecules and randomization process to trigger those reactions. Using various metrics, progress or fitness change over time and success rate, the proposed triggering methods outperformed the standard implementation.

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Enhancement Bag-of-Words Model for Solving the Challenges of Sentiment Analysis

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Abstract—Sentiment analysis is a branch of natural language processing, or machine learning methods. It becomes one of the most important sources in decision making. It can extract, identify, evaluate or otherwise characterizes from the online sentiments reviews. Although Bag-Of-Words model is the most widely used technique for sentiment analysis, it has two major weaknesses: using a manual evaluation for a lexicon in determining the evaluation of words and analyzing sentiments with low accuracy because of neglecting the language grammar effects of the words and ignore semantics of the words. In this paper, we propose a new technique to evaluate online sentiments in one topic domain and produce a solution for some significant sentiment analysis challenges that improves the accuracy of sentiment analysis performed. The proposed technique relies on the enhancement bag-of-words model for evaluating sentiment polarity and score automatically by using the words weight instead of term frequency. This technique also can classify the reviews based on features and keywords of the scientific topic domain. This paper introduces solutions for essential sentiment analysis challenges that are suitable for the review structure. It also examines the effects by the proposed enhancement model to reach higher accuracy.

Keywords—Sentiment analysis; Bag-Of-Words; sentiment analysis challenges; text analysis; Reviews

I. INTRODUCTION

Sentiment Analysis (SA) [1] is a Natural Language Processing and Information Extraction task that aims to obtain researcher's feelings expressed in positive or negative reviews or opinion by analyzing a big numbers of documents and papers [2]. The important issue [3] in sentiment analysis is to recognize how sentiments are expressed in reviews and whether the expressions refer (acceptable) or negative (unacceptable) reviews toward the subject. Sentiment analysis is a laborious [4] task for performing with computers and algorithms. Identifying some patterns is hard for machines or even impossible while it is easy for human beings. There are some intractable situations for computers such as: Can't deal with pronouns and what they refer to. It also can't understand the different meaning of words such as ironies or sarcasm. The difficulty in putting a standard in some punctuation marks such as (!!!). It's very hard to evaluate the world knowledge information. Sentiment analysis requires to generate a big lexicon which takes a big time to search for required words. The Hardness in evaluating words within two polarities: positive and negative. Sometimes we need to understand some features or keywords for each topic or deal with multi-

language. Other problems face sentiment analysis in understanding reviews meaning and grammar for one language or multi-languages. Especially, there are several words that have different meaning and polarities. Thus, sentiment analysis involves identification of sentiment meaning, expressions [5], Polarity and expressions strength, and their relationship to the subject. The volume of linguistics resources is enormous. The bag-of-words (BOW) [6,7] models evaluation uses many techniques such as Naive Bayes (NB), Support Vector Machine (SVM) and Maximum Entropy (ME) classifiers [8] that have been exhibited to go well in the binary positive negative sentiment classification tasks on document-level datasets like movie reviews. There are three essential classification levels in sentiment analysis. The differences between document-level, sentence-level, aspect level and word-level SA declares in the level of evaluation the sentiments and classification polarity based on document, sentence, aspect/entity or word/term. The sequence of sentiment analysis process of each one of them as followed in the figure 1:

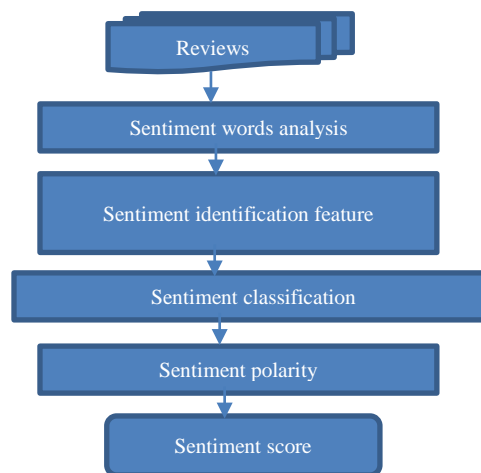


Fig. 1. Sentiment analysis approach

The difference between the four levels of sentiment classification declares in sentiment classification polarity level on document as a whole [9, 10] or on each sentence in a document or text [11] or on aspect or entity level [12] or on each word in text. In this paper, we discuss the sentiment classification and evaluation on word level in one topic domain. We present the sentiment classification regard to word polarity with the specific topic of features or keywords.

The first step is to identify the features and entities the scientific topic. The review writers can give different reviews for different features of the same feature like this sentence "the journal of publication of this paper is not good, but it has a big number of citation". The sentence is subjective, Word-level SA [13] will determine whether the review expresses positive or negative based on evaluate each word polarity related with the feature.

In this paper, we present a new technique which called Sentiment Analysis Of Online Papers "SAOOP". It can evaluate online sentiment reviews for research paper domain. This technique is a new technique which introduces an enhancement of Bag-of-words model to solve major weaknesses of the Bag-Of-Words model in sentiment analysis evaluation. It depends on the word level of sentiment analysis in one topic domain. Additionally, we can extract features and keywords of the domain to classify sentiments reviews and reach the accurately meaning of each review. The proposed technique also can introduce solutions for sentiment analysis challenges to improve accuracy.

The rest of this paper is organized as follows: Section 2 represents related works. Section 3, the presentation of the proposed technique "SAOOP". In Section 4, outlines of the Experiment comparison between the standard and enhancement proposed models. Section 5 highlights the evaluation and discussion results. Finally, Section 6 conclusion and proposes directions for future work.

II. RELATED WORKS

The target of this paper evaluates sentiment analysis and classifies the sentiment polarity automatically and more accurately. Although the sentiment analysis is a hot area to research, No research finds enough in this field till now.

The authors [14] discusses that sentiment analysis becomes the most motivating research area among natural language processing (NLP) community. There are many tools and applications for opinion Mining or sentiment analysis. They also face many research challenges. There are some innovative and effective techniques required to be invented which should overcome the current challenges faced by Opinion Mining and Sentiment Analysis.

In movie domain, the research works on analyzing sentiments on the document level. There is the most distinguished work [15] by using "Poor" and "Excellent" seed words to compute the semantic orientation, point wise the mutual information approach used to calculate the semantic orientation. The sentiment orientation (SO) of the document was computed as the average SO of all such phrases. The accuracy results achieved to 66%.

But when using a lexicon-based approach [16] on also movie review domain, the approach is used to implement the sentiment polarity classification. For this sentiment classification positive and negative words lexicons is used and semantic orientation calculator (SO-CAL) is built that combine intensifiers and negative words. The approach's results showed to have 59.6% to 76.4% accuracy on 1900 documents.

Further research of machine learning [17], which develops a new approach for extracting product features and opinions from a group of free customers;' sentiment reviews about a service or product. This approach depends on a language-modeling framework that, using a seed set of opinion words, can be compatible with any domain reviews and language. The proposed approach combined both a statistical mapping between words and a kernel-based model of opinion words learned from the seed set to approximate a model of product features from which the retrieval is performed

III. PROPOSED TECHNIQUE

In this paper, we propose the enhancement bag-of-words (BOW) model on sentiment analysis by presenting a new technique is called Sentiment analysis of online papers "SAOOP" [18]. The proposed technique SAOOP is based on a word weight instead of the term frequency of each word in the standard BOW. The standard BOW model uses a huge lexicon [19] which has duplications of word and repetition. This lexicon is built manually which requires to create a 'positive' and 'negative' words list [20] by recognize the sentiment polarities based on the personal observation. This approach takes a big time and efforts to compute the total score of sentiments reviews. Another problem of BOW is low accuracy [21] because the standard BOW model neglects text grammatically and ordering of words. So we introduce a new miniature lexicon to reduce the standard lexicon of BOW and deal with adjectives, nouns, verbs, adverbs, adjectives, prefixes, suffixes or other grammatical classes as a word by similarity [22] and differences algorithms. The proposed lexicon is constructed automatically which is based on hierarchical database model [23] to give the correct scores with respect a topic features and keywords. The new lexical approach uses for saving time and ease searching process for each word.

The target of SAOOP is for inferring the polarity of common meaning and polarity concepts from natural language text at a word level, rather than at the syntactic level. With caring topic features with word level of sentiment analysis, the proposed technique also can classify reviews into some categorizations. These categorizations are based on scientific papers topic features and keywords parameters as (place of publication, citation number, topic, publishing paper date, and authors). Although SAOOP aims at the evaluation of words but it can handle some cases of expressions and phrases with respect the order of each word. SAOOP also computes the total score of each review by calculating the aggregate score of review words. For measure accuracy, we make the comparison between our proposed enhancement BOW technique and the standard BOW model on the scientific domain.

A. SAOOP Architecture

he SAOOP technique is presented here which illustrated in fig.2. The architecture has five phases for reaching to sentiment score. The input is online reviews. The phase one called "Analyze Data" which includes some functions as web scraping and extracting data, text analysis, NLP linguistics, and Enhancement BOW.

Web Scarping and extracting data which extracts the papers data and their parameters data and creates rows records. Text analysis contains that splits sentiment reviews into sentences and tokenizes each review into some words. Natural language processing (NLP) Linguistics makes normalization functions and reformat data. Our proposed technique introduce an Enhancement Bag-of-words (BOW) algorithm which is based on a word weight.

The phase two called "Lexicon data" illustrates that creating a lexicon, topic features and keywords, finding names, and classify sentiment reviews. This phase explains that classify reviews from one or more class in assuming five classes (Topic, citation number, the publishing date of paper, authors, and place of publications). In the second function, extract the features and keywords of scientific domain as the names of authors, the names or shortcuts of conferences and journals. Finding names declares that how to recognize some

names of each class. And the last function is the classification reviews by using the previous two functions [24]. In the third phase entitled "sentiment analysis score and polarity", the proposed technique can detect the polarity based on one of sentiment classification (very negative, negative, neutral, positive, and very positive). Although the proposed technique is based on the word-by-word evaluation, it can handle expressions, wish words, some special cases, with caring grammar by using a newly constructed lexicon. The fourth phase is the solutions of sentiment challenges. This depends on the word level, it contains proposed solutions to deal with some challenges "Spam and fake detections", "Implicit and Explicit Negation", and "World knowledge" based on topic features. Additionally creating a map guide which is based on the sentiment scores related to the most related papers according to keywords and fields classification. The output declares in the total sentiment score and polarity of each paper.

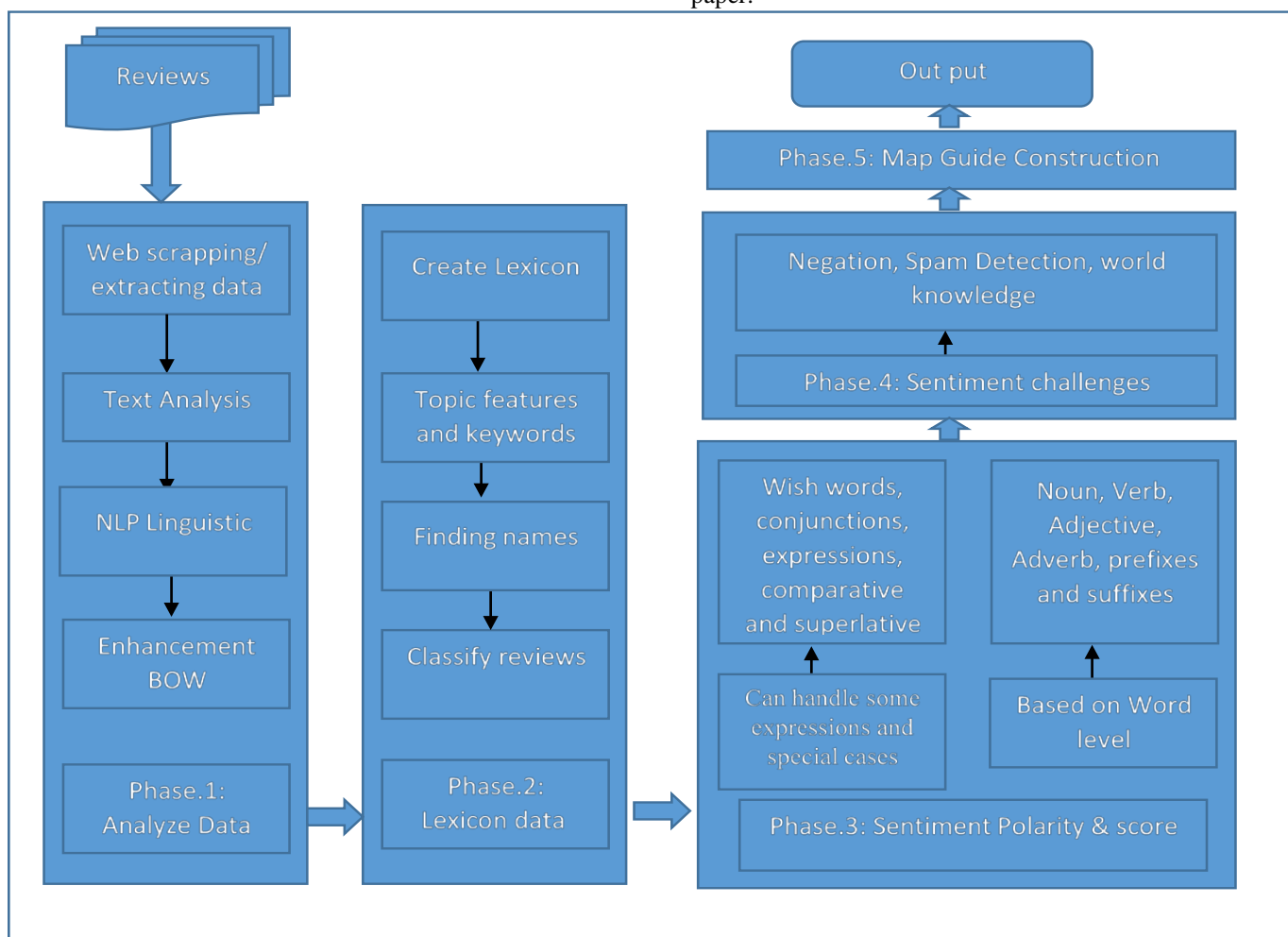


Fig. 2. Proposed technique (saoop) architecture

B. SAOOP Lexicon

The proposed technique creates a new lexicon for sentiments reviews which is based on the hierarchal database model. This model is a data model where the data is organized like a tree. The structure allows repeating

information using parent/child relationships: each parent can have many children but each child only has one parent. All attributes of a specific record are listed under a feature type. The lexicon has words, prefixes and suffixes and hierarchal nouns to produce a solution for bi-polar words and evaluate topic features of reviews. We use Part-of-speech (POS)

tagging [24] model to recognize nouns in constructing the nouns tree in lexicon. The advantage of this hierarchical model of nouns is each parent can hold the same name of child with different value that supports us in evaluating bi-polar and fuzzy words or the topic features which declares in Figure.3:

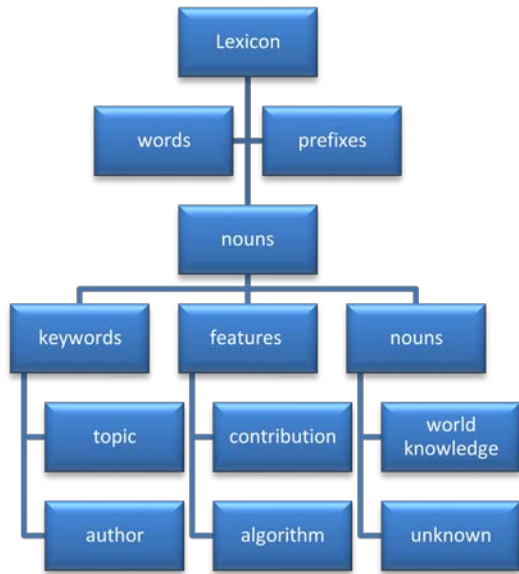


Fig. 3. Proposed lexicon structure

For example if keyword is "topic" and the before word is [old] the polarity will be [negative] but if the keyword is "author" the polarity to the same word [old] will be [positive]. The same case in features but other nouns we splits to identify the world knowledge words and polarities such as a name of famous scientist [Einstein] holds a positive polarity or unknown words.

The proposed lexicon saves a positive Wp and negative Wn score for each word with the recommended polarity, by assuming the equation,

$$\sum Wp + Wn = 1 \quad (1)$$

This model expresses words into positive or negative polarities with saving two scores to support the accurate meaning such as [not great] polarity does not equal [bad] polarity but it equals [good] polarity. SAOOP proposed technique evaluate sentiments with a word weight. It also assumes a desirable state of each word based on the meaning (e.g., "great" and "good") have a positive polarity, while words that encode an undesirable state have a negative polarity (e.g., "bad" and "worst"). Although sentiment polarity normally applies to adjectives and adverbs, there are verbs, expressions, conjunctions, prefixes or suffixes and noun sentiment words as well. We can compile sets of sentiment words and phrases for adjectives, adverbs, verbs, expressions, prefixes, suffixes and nouns respectively.

C. SAOOP Enhancement BOW

In this section, we will explain the comparison between the standard and enhancement bag of words model. We also can

compare between the methodology and challenges, they can face.

1) Standard BOW:

The input of the first technique is some documents, and the output is the sentiment scores for each word. Each document is a bag of words, meaning: Assumes order of words has no significance (the word "home made" has the same probability as "made home"). The first standard BOW algorithm follows the next steps: Bag of words representation (or vector space representation) [25] is the main methodology proposed by information retrieval researchers to represent text corpus, which is an easy approach to converts unstructured text to structured data based on word by word, and neglecting the grammar. This algorithm declares the relationship between documents and evaluates the words based on the term frequency in these documents. There are many algorithms to calculate term weight, we apply here the term frequency-inverse document frequency (tf-idf) which is a numerical statistic that aim at reflecting the importance word is to a text in a groups or corpus.

There are some goals for this algorithm and it can't only to evaluate sentiment score. But it depends on supervised or unsupervised algorithms to compute the score. It can't deal with the words order. We declare the standard algorithm in the following:

Algorithm 1: Pseudo code of Standard BOW

1. The input: Given a corpus of K documents, comprising a dictionary of M words, find the "relations" of words and documents (usually cluster the documents)
2. Get the number of reviews.
3. Create dictionary for all words for all reviews.
4. Calculate the number of frequencies (occurrences) in each review manually.
5. This following formula declares the word frequency in the text.
6. $tf_{i,j} = \frac{n_{i,j}}{\sum_k n_{k,j}}$
7. Where $n_{i,j}$ is the number of occurrences of the considered term in document d_j , and the denominator is the number of occurrences of all terms in document d_j .
8. With $|D|$ is a total number of documents in the corpus and $\{d_j: t_i \in d_j\}$ is the number of documents where the term t_i appears.
9. Then,
10. $tfidf_{i,j} = tf_{i,j} * idf_i$
11. Create vector of frequencies for each review.
12. Create dictionary of positive, negative and neutral words in three files manually.
13. Check on each word manually positive or negative or neutral.
14. It also neglects the grammar and the meaning of the grammar.
15. It can't evaluate the score separately, but it must apply one of the supervised or unsupervised algorithm as K-Nearest Neighbor or Naïve Base Classifier,
16. Compute the sentiment score for each word.
17. Then try to classify the total score polarity class between positive or negative classes.

2) Enhancement BOW

By study of analyzing the online scientific reviews [26], we identify the nature of review structure which is short and formal. So the BOW is the most suitable model to deal with these reviews but we need to improve the accuracy and avoid its weaknesses. Hence, we present an enhancement BOW model in the proposed technique (SAOOP) which is explained steps in the following: The input is online sentiment reviews, and the output is the sentiment scores of each word. Algorithms.2 discusses the pseudo code of the enhancement.

Algorithm 2: Pseudo code of Enhancement BOW

1. For each paper P do
2. Web scrapping data
3. Reformat data
4. Calculate number of sentiment S
5. For each review R in P
6. Check and delete fake or spam sentiments reviews.
7. Get the real number R or reviews.
8. For sentence sent \in classification reviews data do.
9. for review category a \in A do, class='Topic', and Score= 0.
10. For word w \in s do.
11. If $O(w) > 0$ then.
12. Remove stop and punctuation lists.
13. Convert all w into UPPER case.
14. Create a new lexicon for all positive and negative words.
15. Assume each word w has two values (positive and negative), and The total score of w equal 1,
$$\sum Wp + Wn = 1$$
- Assuming each word has 2 values (W p=positive value, W n=negative value)
16. If having explicit negative words (such as not)?
17. Check on the next word w to detect score.
18. We assume the negative value for positive word, And assume the positive value for the negative value for the negative word,
$$V(W) = W(N) - 0.2.$$
19. Each w has class from five sentiment classification levels (very negative, negative, neutral, positive, and very positive).
$$V(W) = W(N) + 0.2.$$
20. Detect sentiment score and polarity.
21. End If.
22. Else if having a word from second negative list such as never.
23. Convert the polarity of the sentence by,
$$V(Sent) = S(Sent) * -1.$$
24. End if.
25. Else if check on future words such as (wish, hope).
26. Check on next word and detect polarity and score (go to step17).
27. Else If.
28. Use POS tagging to check on nouns.
29. If w is noun?
30. If w is feature?
31. Detect sentiment score and polarity
32. End if
33. Else if w is keyword?
34. Detect sentiment score and polarity
35. Else if w is world knowledge
36. Detect sentiment score and polarity
37. Else if go to step 17.
38. End for.
39. Assign review classification of each S in R.

40. If s \in review classes
41. Determine class.
42. End if
43. Else If class = 'topic'
44. Compute sent score and polarity.
45. Calculate R (SA) is a total sentiment score of each review r.
46. End for
47. End for
48. Calculate T (SA) is a total sentiment score of each paper p.
Calculate AVG (SA) is an average sentiment score of each p.
T (SA): is the real total score of all reviews.
r: is the number of real reviews without spam or fake reviews.
49. End For.

D. Sentiment Polarity Detection

The main goal of this word level reaches the accurate polarity for the sentiment review. The proposed technique can detect the polarity values for each input word of the evaluation dataset (with the summation of positive and negative is equal 1). But the polarity depends on the meaning of the review characteristics. SAOOP can assign a polarity based on this approach, considering the words weight replacing term frequency, by assuming each word has two values and polarity with this assumption equation.1. But the sentence contains negative that differs in the word value. If the word is positive, convert to negative polarity and the negative score will be as in the equation,

$$V(w) = W(p) - 0.2. \quad (2)$$

And if the word is negative, the score will be calculated by $V(w) = W(n) + 0.2$. The selection of 0.2 because this division is suitable for the five sentiment class's levels [18]. There is two scores of sentiment analysis, a real sentiment and the average sentiment scores. The r (SA): is a total score of sentiment score of all reviews on each paper with caring of the number of positive reviews. In the next equation,

$$r(SA) = \sum_{i=1}^n \frac{P(SA(R))}{n} \quad (3)$$

The AVG (SA): is a total score of sentiment score of all reviews on each paper. In the next equation,

$$AVG(SA) = \sum_{i=1}^n \frac{P(SA(R))}{n} \quad (4)$$

The calculation of the total score of all reviews depends on the score of each review. There is a difficult problem between a large number of reviews and evaluating sentiment polarity of each one, this problem is improper the most review number having assessment higher score.

E. Sentiment Analysis Challenges Scope

With scanning a data set of CiteuLike [27] scientific reviews for papers, we can detect also the most essential challenges [28] in evaluating sentiments and opinions that are implicit and explicit negative, world knowledge and spam or fake reviews.

The discussion of the solutions in the following:

1) Implicit and Explicit Negative challenge

Negation is the biggest challenge in sentiment analysis [29]. The new technique produces a solution to improve evaluation negative with the enhanced bag of words technique. This research handles the two techniques: explicitly and implicitly negative. First: explicitly is deliberately formed

and are easy to self-report and by keywords. Second implicitly is the unconscious level, are involuntarily formed and are typically unknown to us without any keywords of negative. In addition, the handling the negative meaning of some conjunctions such as “not only”, and “But”. The dual negative is the most important case which cares to achieve the total sentiment polarity. Reverses polarity of mid-level terms: great V.S not great.

2) World knowledge requirement Challenge

The proposed technique presents a solution for Knowledge [30] about worlds’ facts, events, people are often required to correctly classify the text. Trying to achieve higher accuracy and get the evaluation for some neutral reviews. The World knowledge challenge solution is based on the hierarchical database of nouns. Hierarchal model between nouns to achieve the polarity, score and meaning. Also to differ between them and keywords or features. In a hierarchical model, data is organized into a tree-like structure, implying a single parent for each record. A sort field keeps sibling records in a particular order. Hierarchical structures were widely used in the early mainframe database management systems. This structure allows one one-to-many relationship between two types of data. This structure is very efficient to describe many relationships in the real world; recipes, table of contents, ordering of paragraphs/verses, any nested and sorted information.

3) Spam and Fake Reviews Challenge:

The Internet includes both realistic and spam contents [31]. For effective Sentiment classification, this spam content should be eliminated before processing. The proposed technique can be done by empty or identifying duplicates, by detecting outliers and by considering the reviewer reputation. The proposed Technique enhances reviews spam and fake. The proposed SAOOP can avoid and cure the most of them by deleting empty reviews and removing the duplicate sentiment reviews by considering the same reviewer for computing the real number of reviews.

IV. EXPERIMENT

The discussion of this experiment explains the comparison between the proposed technique and the standard BOW technique in online scientific papers domain. This comparison shows the accuracy results based on real dataset. A real set (1000 reviews) from the CiteULike website in computer science branch. This comparison also discusses the challenges solutions impact on evaluating sentiment analysis. We compare between accuracy [32] with the next metrics [33]:

TABLE I. MEASUREMENT METRICS

	Classified as positive	Classified as Negative
Class Positive	TP	FN
Class Negative	FP	TN

Let in a group of reviews have positive sentiment (belong to class positive) and reviews have negative sentiment (belong to class negative). N p n N After classifying these sentiment reviews, class positive had reviews correctly classified under it and reviews wrongly classified under it, while class negative

had documents correctly classified under it and documents wrongly classified under it. Then, in relation to class positive: TP FP TN FN.

$$precision = \frac{TP}{TP+FP} \tag{5}$$

$$Recall = \frac{TP}{TP+FN} \tag{6}$$

The accuracy equation declares in the next equation,

$$Accuracy (acc.) = \frac{TP+TN}{TP+TN+FP+FN} \tag{7}$$

By reporting, all the measurement mentioned above by practical interpretation. The true positive rate or recall can be understood as the rate at which positive reviews are predicted to be positive (R), whereas the true negative rate is the rate at which negative reviews are predicted to be negative. The accuracy represents the rate at which the method predicts results correctly (A). The precision also called the positive predictive rate, calculates how close the measured values are to each other (P).

V. EVALUATION & DISCUSSION

With the examination of the percentage degree of different techniques accuracy [33] on text reviews content. For computing the accuracy of each model, by calculating the intersections of the positive or negative proportion given by each technique. Table.4 presents the percentage of accuracy for the two compared models. Table 4 shows techniques recall, precision, and accuracy.

TABLE II. AVERAGE RESULTS FOR ALL DATASETS

Metric	BOW	SAOOP
Precision	0.834	0.856
Recall	0.560	0.867
Accuracy	0.618	0.817

SAOOP gets a better results of accuracy (82%) than standard (62%). So the enhancement bag of words increases the accuracy with around 20%, as figure.

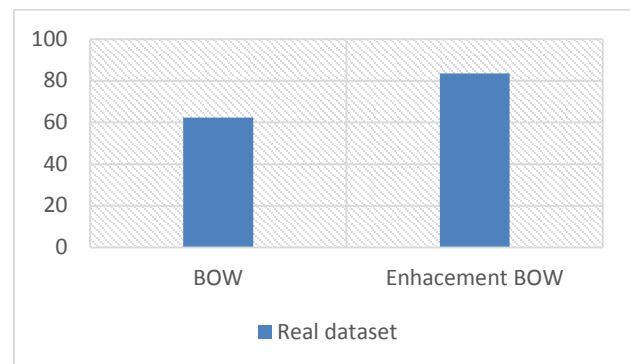


Fig. 4. differences between standard BOW algorithm and proposed SAOOP based on real dataset

In the next table, we discuss the comparison between the two algorithms standard BOW and proposed Enhanced BOW with SAOOP technique. This table comparison relies on

several features, the goal of the algorithm, sentiment classification levels, and the input is reviews, the data size of the input data, data set scope, clarity, efficiency, memorability, simplicity [34]. The definition of memorability is the quality or state of being easy to remember or worth remembering, which can help to declare the relationship between the data and its features. The Algorithm will be clear if it is familiar and easy to use. The efficiency of algorithms depends on the accuracy results in figure.4. The last issue simplicity of algorithm which is simple if it is concise to write down and easy to grasp. Simplicity” of an algorithm is affected by “cultural” factors: Means of presentation (notation, assumptions...etc.) and Previous knowledge of the reader.

TABLE III. COMPARISON BETWEEN STANDARD BOW AND ENHANCEMENT BOW

Algorithm	Standard BOW	Enhancement BOW
Goal	Text analysis and give polarity for words in text	Evaluate sentiment score for reviews
Sentiment classification	2 or three classes	5 classes
Input type	Documents, text or images	Reviews
Data size	Small number of texts or review	Large number of reviews
Data set	Any scope, refer topic domain to minimize dictionary	Topic domain is the best to minimize dictionary and can extraction features and entities
Clarity	No	Yes
Efficiency	No , less accuracy and manually dictionaries	Yes, high accuracy
Memorability	No	Yes
Simplicity	Yes	Yes

Although the last comparison is illustrated, we find the SWOT [35] analysis comparison is very useful to show strengths, weaknesses, opportunities and threats. SWOT [37] analysis [36] became one of the most popular tools for strategic planning or making a decision. It can help in improving our models. Strengths are those features of the business which allow you to operate more effectively than your competitors. Weaknesses are areas capable of improvement. Opportunities identify any new opportunities for techniques. Threats can be external or internal, and are anything which can adversely affect the techniques. With applying SWOT analysis on the compared two algorithms, the results presents in table that discuss the weaknesses of the bow and how can handle them in the strengths of the proposed enhanced BOW model in a new SAOOP technique.

TABLE IV. COMPARISON BETWEEN STANDARD BOW AND ENHANCEMENT BOW IN SWOT ANALYSIS

Algorithm	BOW	SAOOP
SWOT		
Strengths	<ul style="list-style-type: none"> - Ease to use - Using for small reviews - Topic domain - Deal with images, text, and documents 	<ul style="list-style-type: none"> -Improve Bag of words and combine with POS tagging algorithm -categorize reviews - extract features - identify objects and evaluate it Applied KNN- Naive base classifiers to measure accuracy. Graphic reports -Handle some sentiment analysis challenges Easy Clarity High accuracy Topic domain or any domain based on dictionary Memory ability Scale classification -1,0,1
Weaknesses	<ul style="list-style-type: none"> - Less accuracy - Manually dictionary - neglect grammar - neglect ordering - Don't deal with Numbers Questions - Fake or spam review -World knowledge User mention -Hash tags -Emotions 	<ul style="list-style-type: none"> - Not fast enough -Don't deal with Numbers -Questions
Opportunities	<ul style="list-style-type: none"> Automate algorithm High accuracy 	<ul style="list-style-type: none"> -More fast -Arabic sentiment analysis for scientific papers -Create some lexicons to suitable with some domains
Threats	<ul style="list-style-type: none"> Binary words World knowledge Numbers Questions User mention Hash tags Emotions 	<ul style="list-style-type: none"> Numbers (10/10, or 100%) Questions Words not splitting Emotions Deal with hash tags, user mentions and emotions.

Enhancement BOW model in the proposed SAOOP technique has been shown to extremely effective, since it captures more contextual meaning based on word weight, resulting a classification accuracy of 83.5%. Our observation the precision and recall for each sentiment category separately, since the effect of our proposed technique has a significantly different impact on the negative and positive class. Although, the proposed technique increases the precision and recall of both the classes, we could observe a significantly higher improvement of precision and recall in dealing with sentiment challenges. This is a clear indication of the effectiveness of incorporating the impact of world knowledge, spam detection, and negation, by interesting the

topic domain features and keywords and constructing the newly miniature lexicon. Although the proposed technique is based on the word-by-word model, it can understand some phrases as do not directly through caring with the classification of reviews.

VI. CONCLUSION & FUTURE WORK

The technique described in this paper proposes an approach to evaluate sentiment score at the word level. Our contributions include the enhancement of Bag-of-Words model on online scientific papers reviews and the incorporate contextual polarity and effect of sentiment analysis challenges to improve the sentiment accuracy. SAOOP aims at evaluating for reviews of scientific papers and from scientific papers is called CiteULike website, analyzes and classifies the textual content of the sentiment reviews of each paper. The proposed SAOOP can classify sentiment reviews and visualize the relationships between them based on extract features and keywords of scientific domain.

This paper makes a comparison between the standard bag-of-words model and our proposed enhancement bag-of-words and test the impact on sentiment analysis challenges and the accuracy. The experimental results show that our technique obtains sentiment classification accuracy with (83.5 %) that significantly better than the standard BOW (62%). Further, the efficiency of the proposed algorithm improves over standard BOW algorithm. Future research will focus on enhancing the proposed technique further by working on phrases in order to have sufficient local information to determine the polarity. Further, working on the proposed technique (SAOOP) to apply on the Arabic language in scientific paper research.

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QRS Detection Based on an Advanced Multilevel Algorithm

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Abstract—This paper presents an advanced multilevel algorithm used for the QRS complex detection. This method is based on three levels. The first permits the extraction of higher peaks using an adaptive thresholding technique. The second allows the QRS region detection. The last level permits the detection of Q, R and S waves. The proposed algorithm shows interesting results compared to recently published methods. The perspective of this work is the implementation of this method on an embedded system for a real time ECG monitoring system.

Keywords—ECG Signal; QRS Complex; multilevel algorithm; thresholding technique

I. INTRODUCTION

The electrocardiogram (ECG), as illustrated in Fig. 1, presents the electrical activity of the heart. This activity is collected on a patient by electrodes placed on the surface of his skin. The heart is made of muscle cells that conduct electrical impulses. In addition, there are specialized cells organized into a preferential conduction tissue and endowed with the power to depolarize spontaneously, creating cardiac automatism [1]. The ECG is an essential element both in patient monitoring or diagnosis of cardiovascular disease. The theoretical basis and practice of cardiac electrical activity recording were set out by Einthoven. in 1901.

The analyses of the ECG signal permit to evaluate the heart condition of the patients. This signal contains different waves that present repolarization and depolarization of the heart muscle [2-5]. Among these waves, the QRS complex corresponds to the depolarization of the ventricles. This complex shows a large amplitude compared to other waves. This gives it a top priority during the diagnosis of the ECG signal. The QRS complex represents three contiguous waves following the P wave, namely the waves Q, R and S. By definition, the Q wave is the first negative wave, the R wave is the first positive wave, and the S wave is first negative wave after the R wave [6-7]. The average duration of the QRS complex is 0.08 s. It must remain below 0.12 s. Above, it is most often an asynchronous depolarization of two ventricles associated with intraventricular conduction disorder [1].

The automatic processing of the ECG signal is a major challenge to researchers and the engineers in the different tasks of research that deal with this signal, likewise, the ECG signal

denoising, the QRS complex detection and the real time monitoring [8-9]. The ECG signal could be affected by several types of noise, which influence negatively on this signal [10-12]. e.g., the baseline wandering, the high frequency noise and the power line interference. These noises are attested in the recording of the ECG signal. Fluctuations of the baseline are internal noises that disturb the ECG signal. These fluctuations are due to respiration and patient motion during the recording of this signal. The high frequency noises beyond the frequency of the normal ECG signal, which is variable between 0.5 Hz and 150 Hz. These noises are caused by the extra-cardiac muscle activity. The power line interferences are induced by the electrical power supply of appliances. After the ECG signal denoising, the QRS complex detection is the first steps of any analyses of ECG signal's waves. The extraction of this complex is a major issue of the ECG signal processing and it presents an important task for researchers from long time. This difficulty is due to the different morphology of the ECG signal, as illustrated in Fig. 2. Several research works have been proposed to deal with this task. e.g., derivative algorithms [13-15], artificial neural networks [9], DWT [16-17], filter banks [9]. The problem with the majority of these methods is the high complexity of the implementation in the embedded systems. As a solution for this problem, this paper proposes an efficient method based on a multilevel algorithm to solve this issue. This method is based on three levels, namely the extraction of higher peaks using an adaptive thresholding technique, the QRS region detection and the detection of Q, R and S waves. This method is tested on some of the MIT-BIH arrhythmia signals.

The perspective of this work is the implementation of this method on an embedded system for the ECG monitoring system. This gives the possibility of evaluating the patient's cardiac status in real time.

This paper is organized as follows, after the introduction, the next section presents the advanced multilevel algorithm based on adaptive thresholding technique. Next, the results section shows the qualitative and the quantitative results and comparisons of this method over some of the MIT-BIH arrhythmia signals as presented in [1]. Afterwards, the discussion section proposes a detailed analysis of the results. Finally, the last section concludes this paper.

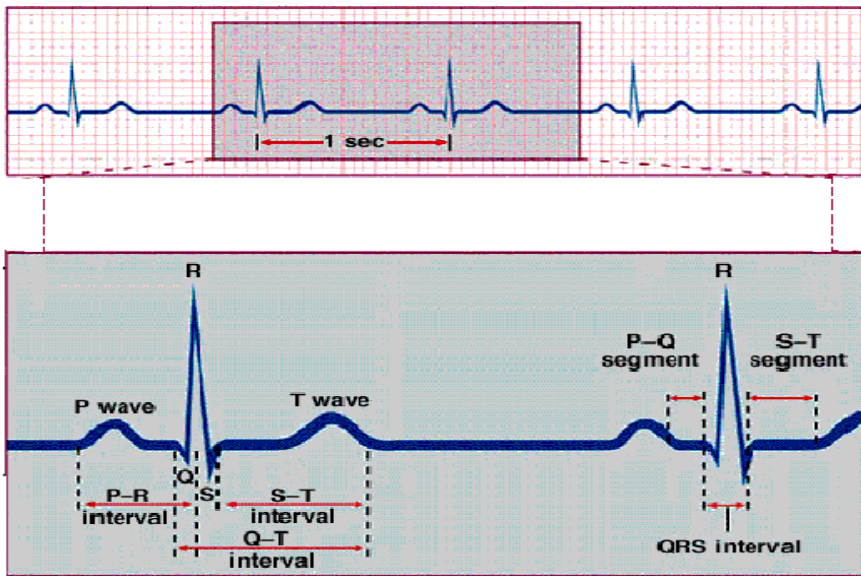


Fig. 1. Normal ECG signal with his different features

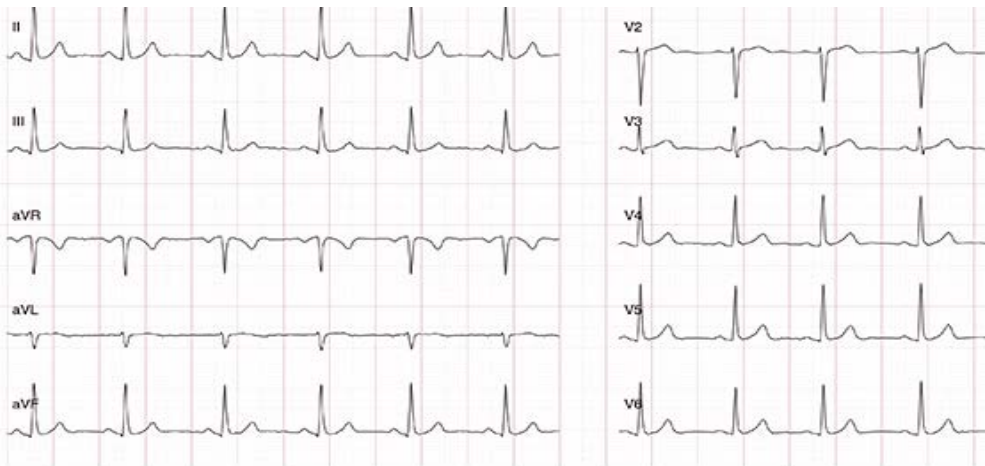


Fig. 2. Example of different morphologies of the ECG signal

II. PROPOSED METHOD

A. ECG Signal denoising

The ECG signal is fragile to different kinds of noises. This makes of the filtering step an essential element of every sort of analysis of the ECG signal.

In our case, we have proposed an efficient solution of the baseline wandering correction and the high frequency noises, which will be presented in further works. The power line interference (50Hz or 60Hz) is not treated with this solution considering the perspective of implementation of this method in an embedded system with a DC supply.

This denoising approach shows interesting results in the case of the baseline wandering issue, as shown in Fig. 3, as well as the high frequency noises as presented in Fig. 4.

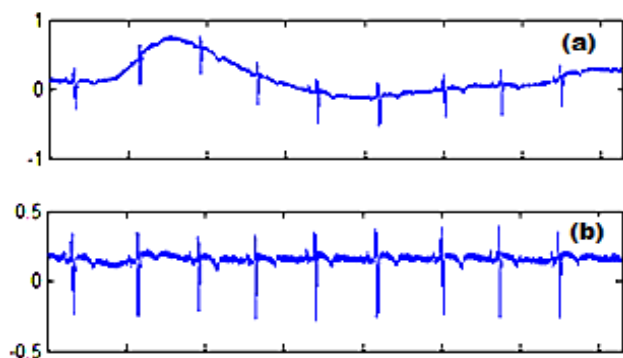


Fig. 3. The baseline wandering correction. (a) noisy signal, (b) corrected signal

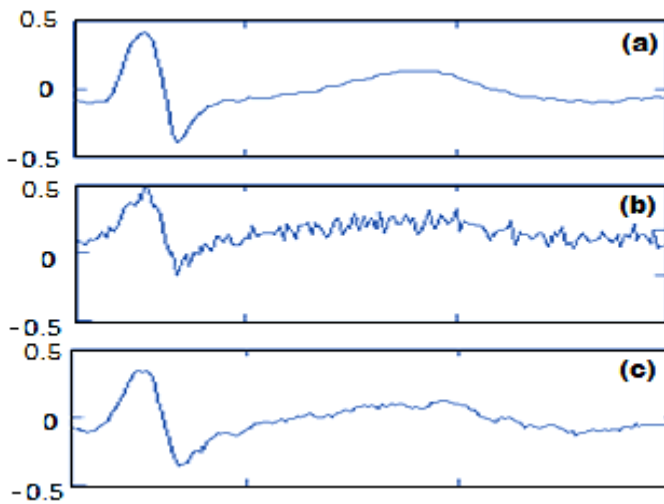


Fig. 4. The high frequency noise correction. (a) original signal, (b) noisy signal, (c) corrected signal

B. Advanced multilevel algorithm

Fig. 5 presents the diagram of the proposed method. The first step presents the ECG signal denoising. Then, the advanced multilevel algorithm, which consists of three levels, namely the extraction of higher peaks using an adaptive thresholding technique, the detection of the QRS region and the detection of Q, R and S waves.

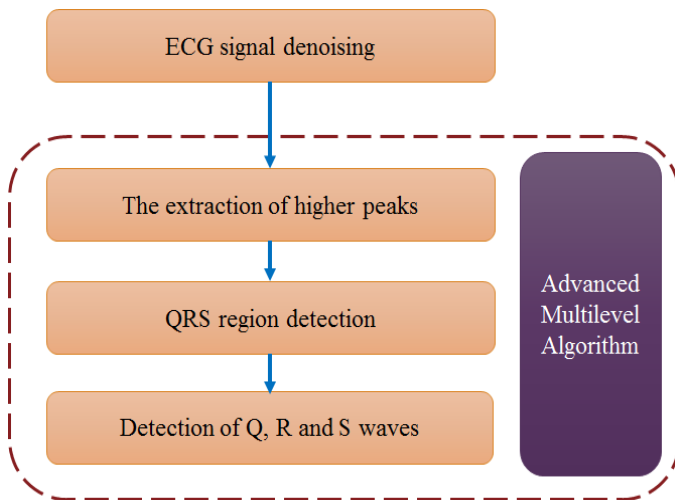


Fig. 5. The diagram of the proposed method

1) *The extraction of higher peaks:* This level is the major element of this method. It permits to locate the position of the QRS complex in the ECG signal. In this level we propose an adaptive thresholding technique. The aim of this technique is to introduce an adaptive threshold, which vary according to the maximum of a moving window. The threshold value is chosen as follows:

$$\beta = M_A \times \alpha \quad (1)$$

Where β presents the threshold value, M_A is the maximum value of the moving window and α is the adaptive thresholding coefficient.

The length of the moving window is 5 seconds that permits to compare every QRS to its neighbours and extract the majority of these complex. In addition, this length doesn't allow to have a large difference between the largest and smallest QRS in this window. This due to the behaviour of the ECG signal, which changes gradually. Therefore, the right choice of the window's length guarantees the high performance of this method.

The MIT-BIH arrhythmia database is frequently used in the evaluation of different algorithms in several thematic interested by the ECG signal [18]. This database contains 48 records; each record is of 30 min length with 360 Hz sampling frequency [18]. This means that the window's length is 1800 samples (5 seconds) and the signal length is 650000 samples (30 minutes). The maximum heartbeats are estimated at 160 beats per minute (bpm). Therefore, the minimum interval between two complexes corresponds to 145 samples for this method.

Following is the algorithm of the adaptive thresholding technique, where $X(i)$ is the absolute value of the original signal and $W(i)$ is the result of this level:

Algorithm of the higher peaks extraction:

```

1. i=1, j=1,  $\alpha=50\%$ , m=1800, n=650000, d=145,
   Loop1: c=1;
2. For i < n
   j=i;
   2.1. For j < m+i
      $M_A = \text{Maxima of } (X(i : i+m))$ ;
     Loop2:  $\beta = M_A \times \alpha$ ;
     2.1.1 For (j=i; j=m+i; j++)
       if  $X(j) \geq \beta$ 
          $W(j) = X(j)$ ;
         j=j+d;
         c=c+1;
       else  $W(j) = 0$ ;
       end if
     end
     2.1.2 if c < 5
       if  $\alpha > 30\%$ 
          $\alpha = \alpha - 5\%$ ;
         j=i;
         goto Loop2;
       end if
     end if
   end
   2.2. i=i+m;
   goto Loop1;
End

```

As presented in this algorithm, the threshold coefficient starts at 50% and decreases by 5% until it reaches at 30%. This coefficient permits to extract the maximum of the peaks present in the moving window.

The condition of the decrease is that the numbers of peaks should be lesser than 5 peaks, which is corresponding to 60 bpm.

To simplify the analysis, it is recommended to divide the original signal. This permits to readily analyse the conduct of the proposed method.

2) *The detection of the QRS region:* After the extraction of the higher peaks, this level consists of the detection of the QRS regions around these peaks. As presented in Fig.3, the S waves in some cases could be higher than the R waves in the absolute value of the original signal. In order to select the right QRS region around a peak, this method proposes to select this region differently depending on the nature of the peaks.

Following is the algorithm of the adaptive thresholding technique, where $U(i)$ is the original signal, $X(i)$ is the absolute value of the original signal and $Q(i)$ is the result of this level:

Algorithm of the QRS region extraction:

```
1.i=1, n=650000, d=145,
Loop1: c=1;
2.For i < n
    if W(i) > 0
        2.1. if U(i) > 0
            Q(i-30:i+30)= U(i-30:i+30);
        else
            Q(i-45:i+15)= U(i-45:i+15);
        end if
        i=i+d;
    else Q(i)=0;
        i=i+1;
    end if
End
```

As presented in this algorithm, the length of the QRS region is 60 samples, which presents to the maximum duration of the QRS complex. This duration corresponds to 160 milliseconds. This coefficient permits to extract the maximum of the peaks present in the moving window.

The decrease condition of this coefficient is that the numbers of peaks should be lesser than 5 peaks, which is corresponding to 60 bpm.

As shown in the condition {2.1}, the position of the QRS region depends to the nature of the peaks. If the peak represents the R wave, then the length of the QRS region will be divided equally around this peak. Otherwise, if the peak represents the S wave, then the position of the QRS region will be selected to make the R wave in the middle of this region.

3) *The detection of the waves Q, R and S:* This level permits the detection of the waves present in the QRS complex. The algorithm of the detection of the waves Q, R and S is as follows:

Algorithm of the QRS region extraction:

```
1.i=1, n=650000, d=145,
Loop1: c=1;
2.For i < n
    if Q(i) ≠ 0
        MR = Maxima of (Q(i : i+60));
        j=i;
        For j <= i+160
            if Q(j) = MR
                Then the correct R peak is detected
                R= j;
                j= j+160;
            else j=j+1;
            end if
        end
        MS = Minima of (Q(i : R));
        j=i;
        For j <= R
            if Q(j) = MS
                Then the correct S peak is detected
                j= j+R;
            else j=j+1;
            end if
        end
        MQ = Minima of (Q(R : R+30));
        j=R;
        For j <= R+30
            if Q(j) = MQ
                Then the correct Q peak is detected
                j= j+30;
            else j=j+1;
            end if
        end
        i=i+160;
    else i=i+1;
    end if
End
```

III. RESULTS

The simulation results have been drawn using MATLAB R2014a. The use of a standard database is recommended to assess the effectiveness of the proposed method. In this paper, the MIT-BIH Arrhythmia signals from n°100 to n°108, as presented in [1], are used in the evaluation of the proposed algorithm.

A. Qualitative results:

The qualitative results show some figures taken from the simulation using MATLAB. These figures allow to perceive the detection of all peaks separately in order to analyse the quality of the extraction. Fig. 6,7 and 8 presents the results of the R peaks detected in the signals n°107, 104 and 106 respectively. Fig. 9 and 10 presents the process of the proposed method in the signals n°105 and 106 respectively.

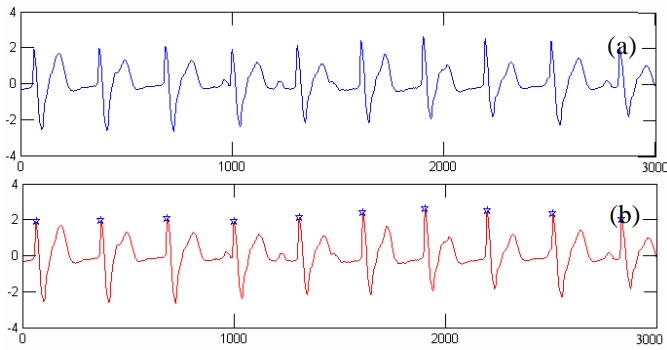


Fig. 6. The R peaks detection of the signal n°107: (a) Original signal, (b) Detection of the higher peaks

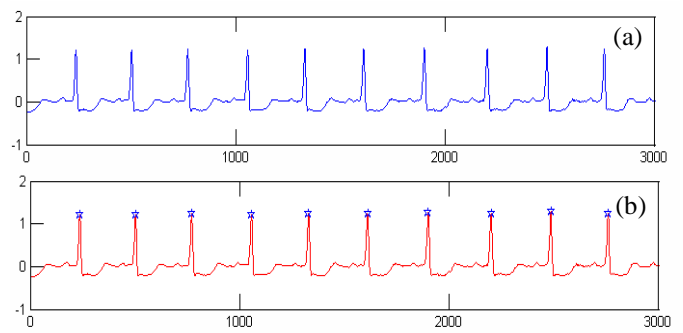


Fig. 7. The R peaks detection of the signal n°104: (a) Original signal, (b) Detection of the higher peaks

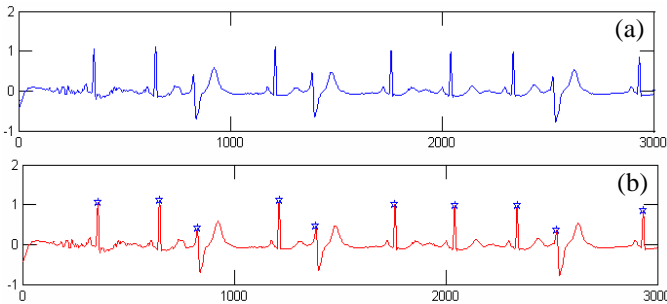


Fig. 8. The R peaks detection of the signal n°106: (a) Original signal, (b) Detection of the higher peaks

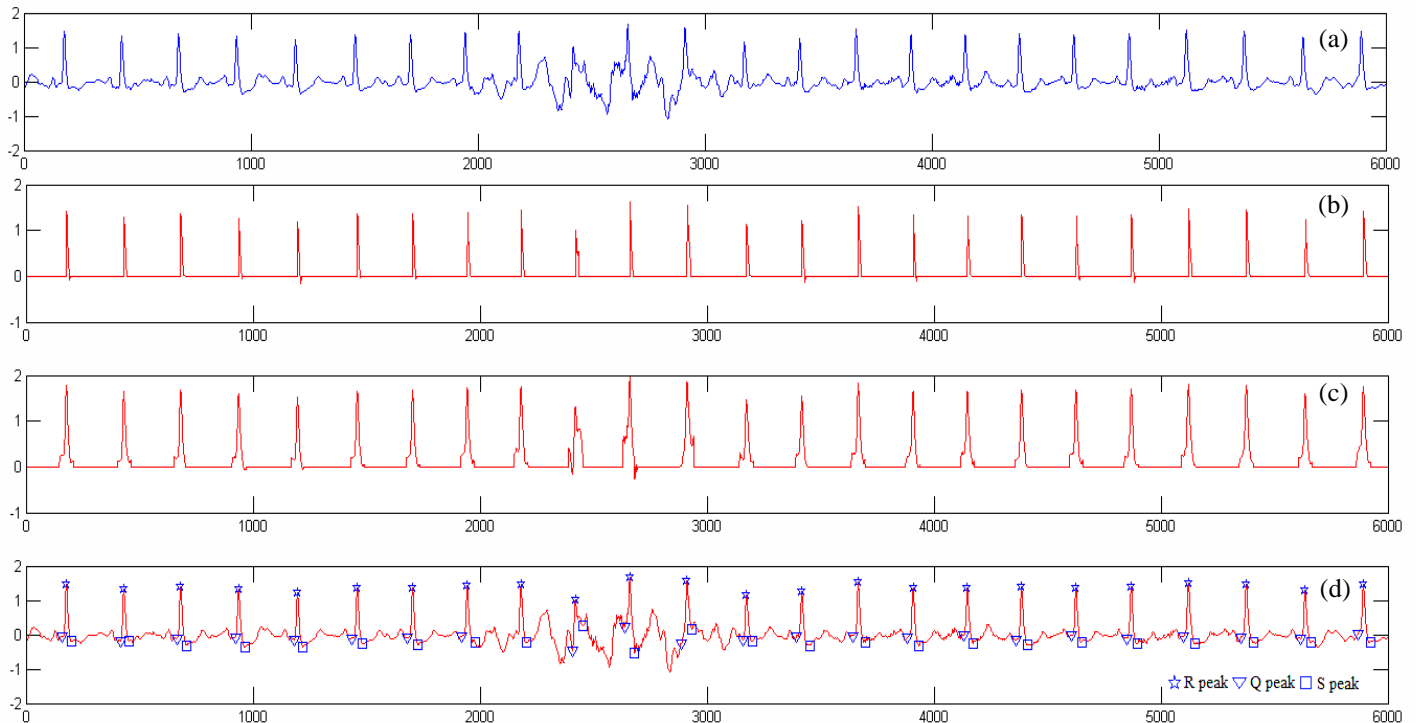


Fig. 9. The process of the proposed method of the signal n°105: (a) Original signal, (b) First level result, (c) Second level result, (d) Third level result

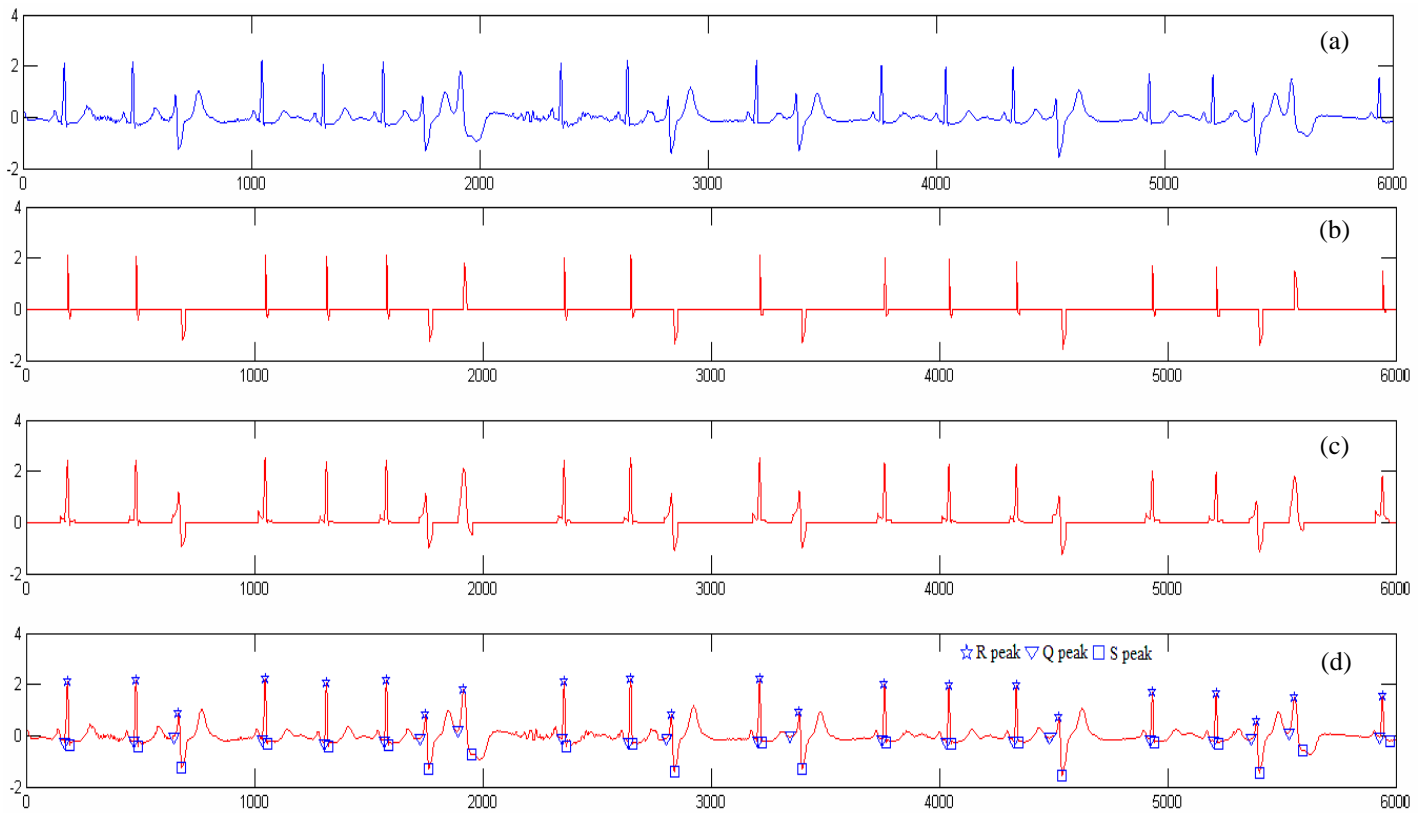


Fig. 10. The process of the proposed method of the signal n°106: (a) Original signal, (b) First level result, (c) Second level result, (d) Third level result

B. Quantitative results:

The quantitative results permit to evaluate statistically the efficiency of the proposed method. The evaluation is established by using the following statistical parameters, true beats (TB), true positive (TP), false negative (FN), False positive (FP), positive predictivity (Pp), sensitivity (Se) and error rate. The true positive is the true detected beats. The false positive is the false detected beats. The false negative is the undetected beats. The total beats is the total number of beats presented in the original signal. The positive predictivity, the sensitivity and the error rate are defined as follows:

$$Se (\%) = 100 \times TP / (TP+FN) \quad (2)$$

$$Pp (\%) = 100 \times TP / (TP+FP) \quad (3)$$

$$Err (\%) = 100 \times (FN+FP) / TB \quad (4)$$

Table 1 presents the statistical results of the proposed method compared to some theoretical methods recently published [1,19,20]. This results are related to the detection of the QRS regions over these methods and the proposed one.

IV. DISCUSSION

The proposed approach provides an important solution of the detection of the QRS complex. This method consists of three levels. The first permits the extraction of higher peaks using an adaptive thresholding technique. The second allows

the QRS region detection. The last level permits the detection of Q, R and S waves.

Fig.6, 7 and 8 present the extraction of the R peaks. As shown in these figures, the proposed method provides an important solution for the complex morphologies of the ECG signal. Fig.9 and 10 show the process of the proposed method. As presented in these figures, this process assures a high extraction's quality of the QRS waves presents in the analysed signal.

The quantitative results, as presented in the table 1, show competitive statistics compared to some theoretical results where the implementation is complex. This comparison is focused in the database signals from 100 to 108. The average error of the proposed method in these signals is 0.53%, which is lesser that the comparative methods. The error of the method presented in [19] is closer to the result of the proposed approach. Whereas the Advanced Multilevel Algorithm presents a simple algorithm to deal with the QRS extraction task, the implementation of this algorithm in an embedded system is simple, unlike the method proposed by [19] which is complex. The sensitivity result of the proposed approach is higher that the comparative methods. This signified that the proposed method permits the minimum of the false negative in the signals presented in the table 1. The result of the positive predictivity presents a competitive result of 99.71%.

TABLE I. COMPARAISON OF QUANTITATIVE RESULTS

MIT-BIH arrhythmia	Methods	TB	TP	FP	FN	Se	Pp	Err
100	The proposed method	2273	2273	0	0	100,00	100,00	0,00
	Z. Zidelmal <i>et al.</i> (2014) [19]		2273	0	0	100,00	100,00	0,00
	Z. Zidelmal <i>et al.</i> (2012) [20]		2273	0	0	100,00	100,00	0,00
101	The proposed method	1865	1864	1	1	99,95	99,95	0,11
	Z. Zidelmal <i>et al.</i> (2014) [19]		1864	2	1	99,95	99,89	0,16
	Z. Zidelmal <i>et al.</i> (2012) [20]		1864	2	1	99,95	99,89	0,16
102	The proposed method	2187	2187	0	0	100,00	100,00	0,00
	Z. Zidelmal <i>et al.</i> (2014) [19]		2185	0	2	99,91	100,00	0,09
	Z. Zidelmal <i>et al.</i> (2012) [20]		2185	0	2	99,91	100,00	0,09
103	The proposed method	2084	2084	0	0	100,00	100,00	0,00
	Z. Zidelmal <i>et al.</i> (2014) [19]		2084	0	0	100,00	100,00	0,00
	Z. Zidelmal <i>et al.</i> (2012) [20]		2084	0	0	100,00	100,00	0,00
104	The proposed method	2229	2229	0	0	100,00	100,00	0,00
	Z. Zidelmal <i>et al.</i> (2014) [19]		2220	5	9	99,60	99,78	0,63
	Z. Zidelmal <i>et al.</i> (2012) [20]		2221	12	18	99,20	99,46	1,35
105	The proposed method	2572	2550	16	22	99,14	99,38	1,48
	Z. Zidelmal <i>et al.</i> (2014) [19]		2550	10	22	99,14	99,61	1,24
	Z. Zidelmal <i>et al.</i> (2012) [20]		2528	15	44	98,29	99,41	2,29
106	The proposed method	2027	2023	0	4	99,80	100,00	0,20
	Z. Zidelmal <i>et al.</i> (2014) [19]		2027	8	0	100,00	99,61	0,39
	Z. Zidelmal <i>et al.</i> (2012) [20]		2027	8	0	100,00	99,61	0,39
107	The proposed method	2137	2137	23	0	100,00	98,94	1,08
	Z. Zidelmal <i>et al.</i> (2014) [19]		2137	2	0	100,00	99,91	0,09
	Z. Zidelmal <i>et al.</i> (2012) [20]		2134	4	3	99,86	99,81	0,33
108	The proposed method	1763	1743	15	20	98,87	99,15	1,99
	Z. Zidelmal <i>et al.</i> (2014) [19]		1740	20	23	98,70	98,86	2,44
	Z. Zidelmal <i>et al.</i> (2012) [20]		1728	25	35	98,01	98,57	3,40
Total	The proposed method	19137	19090	55	47	99,75	99,71	0,53
	Z. Zidelmal <i>et al.</i> (2014) [19]		19080	47	57	99,70	99,75	0,54
	Z. Zidelmal <i>et al.</i> (2012) [20]		19044	66	103	99,46	99,65	0,88
	S. Banerjee <i>et al.</i> (2012) [1]		19022	40	76	99,60	99,79	0,61

The qualitative and the quantitative results show that, spite of the simplicity of this method, the advanced multilevel algorithm presents a very important solution for the extraction of the QRS complex.

V. CONCLUSION

This paper presents an advanced multilevel algorithm used for the QRS complex detection. This method is based on three levels. The first permits the extraction of higher peaks using an adaptive thresholding technique. The second allows the QRS region detection. The last level permits the detection of the waves Q, R and S.

The aim of this method is to present a simple algorithm to deal with the QRS complex extraction. The result of this method should be competitive with the high quality of the extraction presented in the recent research works, despite of the complexity of these works. These results allow us to implement this method in an embedded system with an assured quality of the extraction.

As presented in the qualitative and the quantitative results, the proposed approach shows interesting results. The high quality of the extraction as well as the competitiveness of the statistical results is assured. As a conclusion, spite of the simplicity of this method, the advanced multilevel algorithm presents a very important solution for the extraction of the QRS complex.

The next step of this work is to implement this method in a real-time system. The purpose of this step is to develop a wireless monitoring prototype using an embedded system. The system proposed for this work is a Digital Signal Processor (DSP). This gives the possibility of evaluating the patient's cardiac status in real time.

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FPGA Implementation of Adaptive Neuro-Fuzzy Inference Systems Controller for Greenhouse Climate

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Abstract—This paper describes a Field-programmable Gate Array (FPGA) implementation of Adaptive Neuro-fuzzy Inference Systems (ANFIS) using Very High-Speed Integrated Circuit Hardware-Description Language (VHDL) for controlling temperature and humidity inside a tomato greenhouse. The main advantages of using the HDL approach are rapid prototyping and allowing usage of powerful synthesis controller through the use of the VHDL code. The use of hardware description language (HDL) in the application is suitable for implementation into an Application Specific Integrated Circuit (ASIC) and Field tools such as Quartus II 8.1. A set of six inputs meteorological and control actuators parameters that have a major impact on the greenhouse climate was chosen to represent the growing process of tomato plants. In this contribution, we discussed the construction of an ANFIS system that seeks to provide a linguistic model for the estimation of greenhouse climate from the meteorological data and control actuators during 48 days of seedlings growth embedded in the trained neural network and optimized using the backpropagation and the least square algorithm with 500 iterations. The simulation results have shown the efficiency of the implemented controller.

Keywords—Neuro-Fuzzy; ANFIS; VHDL; FPGA; Quartus; ASIC

I. INTRODUCTION

Under greenhouse production, the climate control is a tool used for yield crop manipulation that maximizes the entrepreneurial benefits. Once the objectives that optimize crop growth and development are defined, the control engineer must design and implement automatic control systems that make possible to obtain a maximum crop yield at minimum production costs. In this sense, control engineering has undergone a considerable development. Researchers have used many control techniques in different fields, from the conventional or classic strategies [proportional integral derivative (PID) control, cascade], artificial intelligence (AI) (fuzzy control, neural networks and genetic algorithms), advanced control techniques (predictive control, adaptive), to robust control strategies, non-linear and optimal control. Specifically, they have been applied in the area of greenhouse climate control [1][2][3]. Conventional control techniques are

difficult to implement in greenhouse systems due to their multi-variable and non-linear nature. Where interrelations between internal and external variables are complex (non-linear physical phenomena that govern these systems dynamics are complicated). This provides justification for the use of intelligent control techniques as a good alternative. In this way, fuzzy logic as part of AI techniques is an attractive and well-established approach to solving control problems [4].

We were brought to develop a Neuro-Fuzzy control of the internal humidity and internal temperature of the greenhouse. This last characterizes the operation of the complex system that the greenhouse constitutes. The identification that is in the center of this step is a process of search for a mathematical representation that minimizes the variations of the real system compared to the modeled system. The development of the plant is influenced mainly by the environmental, climatic variables. The greenhouse, which is a closed circle in which the climatic variables can be controlled, constitutes the ideal medium for the control of the plants growth. The greenhouse must not only create the favorable conditions of the plants growth, but it must moreover be able to ensure certain flexibility in the calendar of production: precocity and spreading out of the calendar. To carry out this objective a robust model using the Artificial Neural Networks and the fuzzy logic can be well adapted to control the nonlinear comportment of greenhouse climate accurately is more than necessary [5].

For the implementation of agricultural technologies (innovations in control systems, remote monitoring, information management), robustness, low-cost and real-time capabilities are needed. In this sense, field programmable gate arrays (FPGAs) proved as a good option for greenhouse technology development and implementation, because FPGAs allow fast development of prototypes and the design of complex hardware systems. These devices are used in many real applications [6]. Through FPGAs, rapid tests, modifications accomplishment, up-dates using single software modifications and an effective production cost (relation performance-price is very favorable) are obtained. In the same sense, reduction in development and commercialisation time is

accomplished. On the other hand, for neuro-fuzzy control implementation, which based on software or hardware, FPGAs are an alternative that keep both benefits, hardware speed and software flexibility. Research made about these devices has experienced an enormous development, in the academic field as well as in the industrial area. There is a great number of contributions about FPGAs applications in different fields [7][8][9]. Also, there are some contributions reported about hardware implementations of neuro-fuzzy control [10]. Moreover, problems of digitized neuro-fuzzy control have been studied [11]. The approach proposed here is focused on greenhouse technologies development, based on AI techniques, particularly fuzzy logic, cascaded with a feed-forward a neural network, and system-on-a-chip (SoC) applications using FPGA technology, with the purpose of obtaining complete engineering solutions on a single Integrated circuit. In our case, an intelligent SoC was developed to carry out the perfect functionality for the greenhouse climate control due to an ANFIS system that seeks to provide a linguistic model for the estimation of greenhouse climate from the meteorological data and control actuators during 48 days of seedlings growth embedded in the trained neural network and optimized using the backpropagation and the least square algorithm with 500 iterations.

II. NEURONAL METHODS IN THE FUZZY SYSTEMS

In a conventional fuzzy inference system, the number of rules is decided by an expert who is familiar with the system to be modeled. In this particular case study the rules generated by an agriculture expert and the number of membership functions assigned to each input is chosen from real data. This is carried out by examining the desired and real input-output data. This situation is much the same as ANN's. In this section ANFIS topology and the learning method used for this neural network are presented. Both neural network and fuzzy logic are model-free estimators and share the common ability to deal with the uncertainties and noise. It is possible to convert fuzzy logic architecture to a neural network and vice versa [12]. This makes it possible to combine the advantages of neural network and fuzzy logic [13-14].

Layer 1: Every node in i in this layer is a square node with a node function

$$o_i^1 = \mu_{A_i}(x) \quad (1)$$

Where x is the input node i , and A_i is the linguistic label (Minimum, Moderate, Maximum) associated with this node function. In other words, o_i^1 is the membership function and it specifies the degree to which the A_i given x satisfies the quantifier A_i . Usually we choose $\mu_{A_i}(x)$ to be bell shaped with maximum equal to 11, moderate equal to 00 and minimum equal to 10, such as

$$\mu_{A_i}(x) = \frac{1}{1 + \left[\left(\frac{x - c_i}{a_i} \right)^2 \right]^{b_i}} \quad (2)$$

Where $\{a_i, b_i, c_i\}$ is the parameter set. As the values of these parameters change, the best bell-shaped functions vary

accordingly, thus exhibiting various forms of membership functions on linguistic label A_i . In fact, any continuous and piecewise differentiable functions, such as commonly used trapezoidal or triangular-shaped membership functions are also qualified candidates for node functions in this layer. Parameters in this layer are referred to as premise parameters.

Layer 2: Every node in this layer is a circle node labeled Π which multiplies the incoming signals and sends the product out. For instance,

$$w_i = \mu_{A_i}(x) * \mu_{A_i}(y), i = 1, \dots, 40 \quad (3)$$

Each node output represents the firing strength of a rule (In fact, other T-norm operators that perform generalized AND can be used as the node function in this layer).

Layer 3: Every node in this layer is a circle node labeled N . The i^{th} node calculates the ratio of the i^{th} rule's firing strength to the sum of all rules firing strengths:

$$\bar{w}_i = \frac{w_i}{w_1 + \dots + w_{40}}, i = 1, \dots, 40 \quad (4)$$

For convenience, outputs of this layer are called normalized firing strengths.

Layer 4: Every node in this layer is a square node with a node function

$$O_i^4 = \bar{w}_i f = \bar{w}(p_i x + q_i y + r_i) \quad (5)$$

Where \bar{w}_i is the output of layer 3, and (p_i, q_i, r_i) is the parameter set. Parameters in this layer will be referred to as consequent parameters.

Layer 5: The single node in this layer is a circle node labeled Σ that computes the overall output as the summation of all incoming signals.

$$O_1^5 = \text{overalloutput} = \sum \bar{w}_i f = \frac{\sum_i w_i f}{\sum_i w_i} \quad (6)$$

Thus we have constructed an adaptive network which is functionally equivalent to a fuzzy inference system [14-15]. The hybrid algorithm is applied to this architecture. This means that, in the forward pass of the hybrid learning algorithm, functional signals go forward up to fourth layer and the consequent parameters are identified by the least squares estimation. In the last backward and the premise parameters are updated by the gradient descent [14].

A. ANFIS Predictive Architecture

Using a given input/output data set, the ANFIS method constructs a fuzzy inference system (FIS) whose membership function parameters are tuned (adjusted) using either a backpropagation algorithm alone, or in combination with a least squares type of method. This allows fuzzy systems to learn from the data they are modeling. FIS Structure is a network-type structure similar to that of a neural network, which maps inputs through input membership functions and associated parameters, and then through output membership functions and associated parameters to outputs [16].

In our case ANFIS is a four-layer neural network that simulates the working principle of a fuzzy inference system. The linguistic nodes in layers one and four represent the input and output linguistic variables, respectively. Nodes in layers two are term nodes acting as membership functions for input variables. Each neuron in the third layer represents one fuzzy rule, with input connections representing preconditions of the rule and the output connection representing consequences of the rules. Initially, all these layers are fully connected, representing all possible rules.

Six feature variables, internal temperature, internal humidity, external temperature, external humidity, global radiation and wind speed, are selected as inputs of the ANFIS. Three membership functions (Mfs) are assigned to each linguistic variable. The suggested ANFIS model is shown in “Fig. 1”.

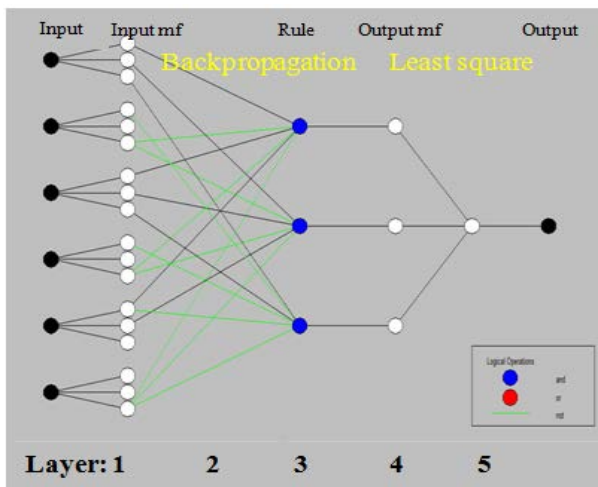


Fig. 1. ANFIS model structure of greenhouse climate

B. ANFIS Modeling, Training and Testing

ANFIS modeling process starts by obtaining a data set (input-output data) and dividing it into training, testing and checking data sets. Training data constitutes a set of input and output vectors. The data is normalized in order to make it suitable for the training process. This was done by mapping each term to a value between 00, 01 and 10 using the Min, moderate and Max method. This normalized data was utilized as the inputs (internal climate conditions and meteorological data) and outputs (actuators conditions) to train the ANFIS. In other words, two vectors are formed in order to train the ANFIS. Input vector = [internal temperature, internal humidity, external temperature, external humidity, global radiation and wind speed]. The output vector = [Ventilating and heating]. The ANFIS registers the input data only in the numerical form therefore the information about the control actuators, internal and external climate of the greenhouse must be transformed into numerical code.

The training data set is used to find the initial premise parameters for the membership functions by equally spacing each of the membership functions. A threshold value for the error between the actual and desired output is determined. The consequent parameters are found using the least-squares method.

Then an error for each data pair is found. If this error is larger than the threshold value, update the premise parameters using the gradient decent method as the following ($Q_{next} = Q_{nov} + \eta d$, where Q is a parameter that minimizes the error, η the learning rate, and d is a direction vector). The process is terminated when the error becomes less than the threshold value. Then the checking data set is used to compare the model with actual system. A lower threshold value is used if the model does not represent the system.

“Fig. 2”, shows the uniform falling of the value of testing error ETest with the number of iterations during the testing process for the ANFIS configuration with traingular Mf and with gaussian Mf. The smallest error of testing (ETest) is reached at iteration 145 (traingular Mf) and at iteration 107 for Gaussian Mf. It can be seen in the “Fig. 2”, that error converges not to zero but to 12% and 2%. This is caused by the presence of some contradicting examples in the training and testing set.

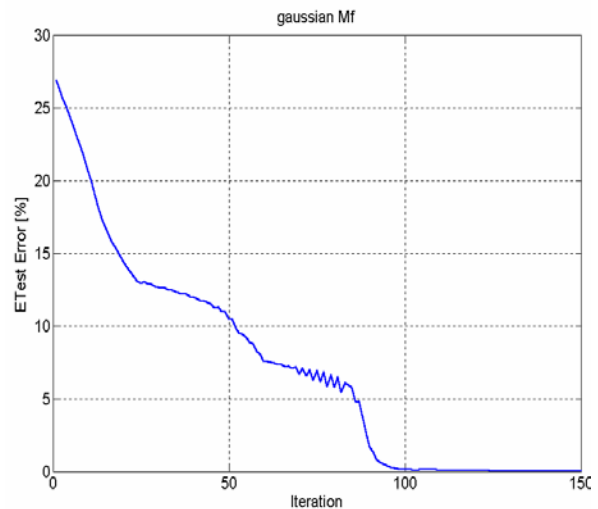


Fig. 2. Decrease of error during the testing process for the ANFIS configuration with Traingular Mf and with Gaussian Mf

Training of the ANFIS can be stopped by two methods. In the first method, ANFIS will be stopped to learn only when the testing error is less than the tolerance limit. This tolerance limit would be defined at the beginning of the training. It is obvious that the performance of the ANFIS that is trained with lower tolerance is greater than ANFIS that is trained with higher tolerance limit. In this method the learning time will change with the architecture of the ANFIS. The second method to stop the learning is to put constraint on the number of learning iterations. In our study, the ANFIS architecture is stopped to learn after 500 training iterations.

III. NEURO-FUZZY CLIMATE CONTROLLER

As is already known from neuro-fuzzy principles, a neuro fuzzy controller acts as a non-linear system capable of implementing expert reasoning for computation of the control values. Indeed, a neuro fuzzy controller which is defined by a set of linguistic rules and fuzzy sets were trained by neural network and optimized using the back-propagation and the least square algorithm is able to compute appropriate values

for greenhouse actuators (heating, ventilating) taking into account information data from the system for control proposes. In the experimental greenhouse, the temperature is controlled by means of heaters, while the humidity is controlled indirectly with the ventilating index regulation. That affects on the temperature and the humidity. Using the physical model, a complete system simulator is shown in “Fig. 3”. With this simulator, a first experiment was carried out using a conventional controller (on-off) with a dead band of 2°C; this is based on a heating system that is activated or deactivated when the error exceeds the fixed regulation range. The humidity depends on the internal air temperature and the ventilating index. This variable is regulated by windows opening in the greenhouse according to the wind speed measurements.

In this case, a multiple inputs, multiple outputs (MIMO) non-linear controller for temperature regulation was used. A MIMO neuro fuzzy controller can be distributed in several multiple inputs, single output (MISO) controllers keeping the same performance. These controllers are independent and can be executed in parallel, which is advisable for the climate controller implementation in a FPGA. The Neuro Fuzzy Controller has six input variables and two output variables, characterized by three fuzzy sets in the universe of discourse.

Input variables are inside and outside temperature (T_i , T_{ext}), inside and outside humidity (H_i , H_{ext}), global radiation (G_r) and wind speed (W_s). Membership functions sets and their appropriate modifications were obtained following a test and error strategy by making exhaustive simulations in Matlab until reaching a good performance through a careful tuning. “Fig. 4”, shows an example of a membership functions set for the input. For this one, three linguistic variables were used (MIN, minimum; MOY, medium; MAX, maximum). The set of fuzzy rules to develop the controller for each variable has been obtained from the expert grower. For tuning the fuzzy rules as well as for membership functions sets a trial-and-error strategy (manual tuning) was used, this is modifying control rule sets until we reaching a good performance of the controller by using the ANFIS editor (simulation system). Each possible linguistic value of inputs is assigned to a consequential action.

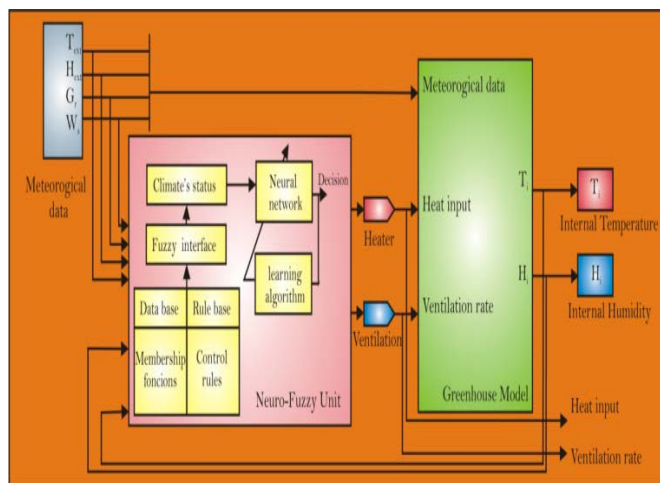


Fig. 3. The model of the greenhouse control system

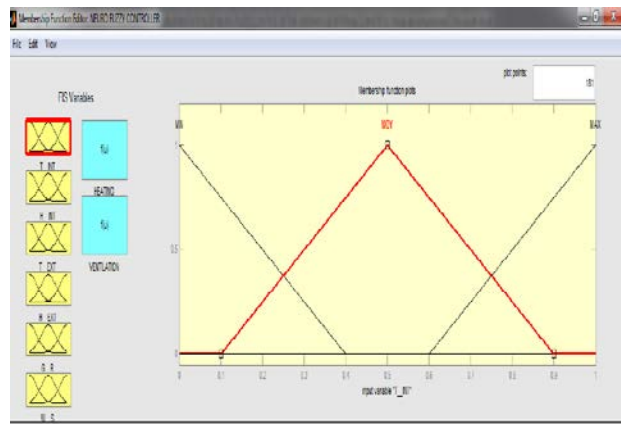


Fig. 4. Membership function of internal temperature

IV. DESIGN AND HARDWARE IMPLEMENTATION

The Neuro Fuzzy Controller shown in “Fig. 5”, has been implemented on an FPGA. The hardware platform used is the Altera DE2 development and education board that is based on the Altera Cyclone II EP2C35F672C6 FPGA.

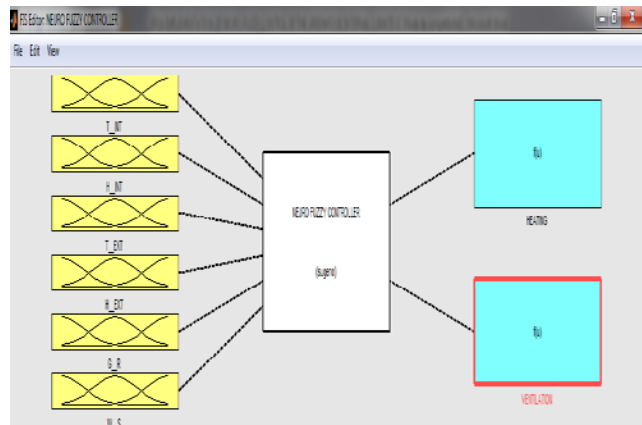


Fig. 5. Neuro Fuzzy Controller

In order to implement our application effectively the design is broken down into modules.

A. The de-multiplexer component

The system should accept multiple inputs with 8-bits in total of 48-bits. In order to reduce the number of pins used in FPGA we have made a de-multiplexing as shown in “Fig. 6”, it has one input of 8-bits and three selection lines, in order to learn at each clock pulse one input and he settles it into a buffer. After six clock top it will acquire all inputs. At the seventh clock pulse it delivers the enable signal and the values of multiples inputs to the rest of the system.

B. The Fuzzier module

In this section we have realized six blocks, where each block is intended for one of memberships functions. The example of such block is presented in the figure it used for the external temperature given in “Fig. 7”. The blocks transformed the numerical data to three linguistic variables (MIN, MOY, MAX). For easy implementation and as we have three cases two bits are used to materialize these case as follows (min => 10, moy => 00 and max => 11).

V. RESULTS AND DISCUSSION

The next step is the simulation of the design to illustrate how it works. "Fig. 9", shows the global simulation timing obtained by Quartus II version 8.1 SJ Web edition. 'Data' are the input values information, (T_int, T_ext, H_int, H_ext, W_s and R_g) are the values of the deferent parameters, (fuz_Tint, fuz_Text, fuz_Hint, fuz_Hext, fuz_Ws and fuz_Rg) are the resultant of all the membership functions.

'Cmd-H-W' is the finally output value represent the ventilating and heating.

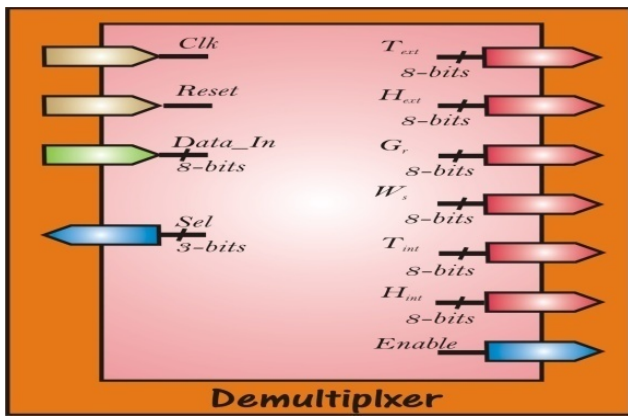


Fig. 6. The de-multiplexer component

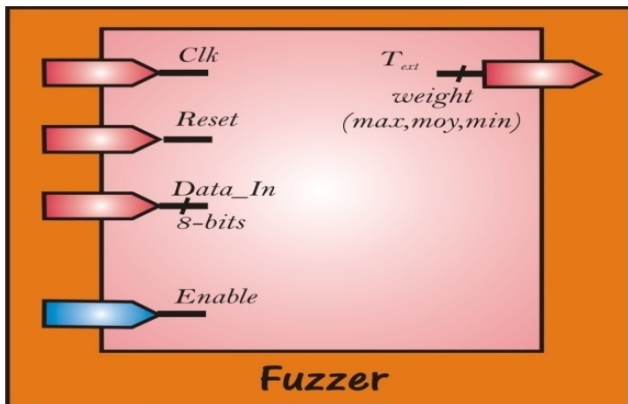


Fig. 7. The fuzzer module

C. The command module

The following operation is the order of the ventilating and the heating. This component shown in "Fig. 8", admits at the inputs the various decisions for the multiple inputs and it will computing the rules of our FIS structure obtained by Matlab Fuzzy Logic Toolbox. To reduce the use of the hardware resource, finite state machine (FSM) is adopted to model this computing process. Finally it will transform the linguistics values on the binary values.

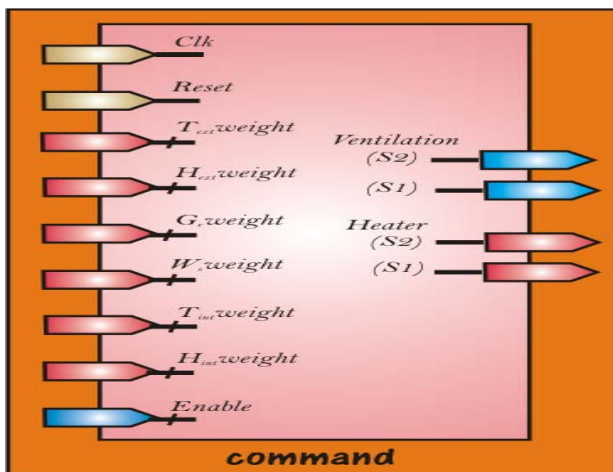


Fig. 8. The command component

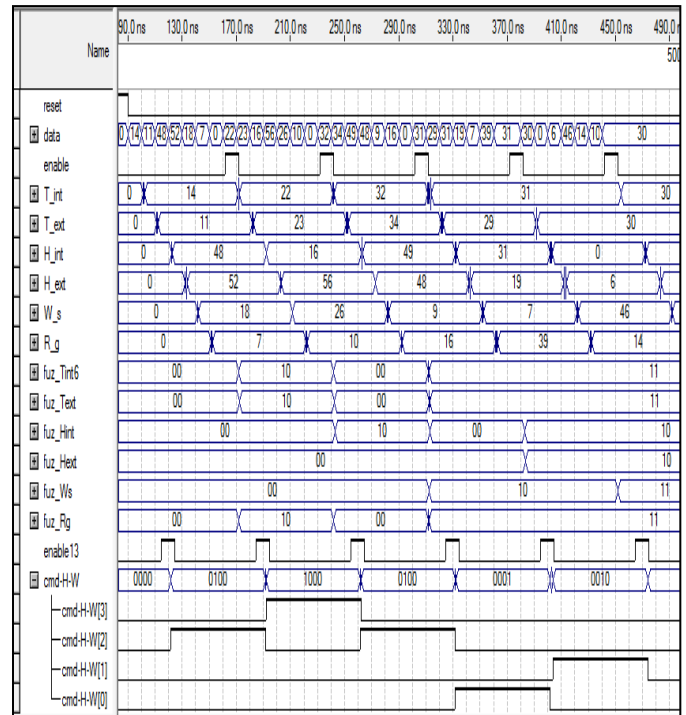


Fig. 9. Neuro Fuzzy Controller Quartus II simulation

The table I shows the strong similarity between the results obtained by Matlab Fuzzy Logic Toolbox environment and those obtained in "Fig. 9". It shows the best operation of all modules. We can also see how the transformation of these data from the linguistic values to numerical values.

Synthesis of fuzzy neural network on FPGA:

We have implemented the design using the DE2 board, contain Cyclone®II 2C35Altera FPGA device, EP1C6Q240. The principal features of Cyclone II EP2C35 FPGA are as follows:

- 33216 Logic elements.
- 105 M 4K RAM blocks.
- 483,840totalRAMbits.
- 35 embedded 1818 multipliers.
- Four PLLs.
- 475 user I/O pins.

TABLE I. COMPARISON OF THE RESULTS GIVEN BY MATLAB FUZZY LOGIC TOOLBOX AND THOSE OBTAINED WITH QUARTUS II

Meteorological data				Greenhouse data		Decision control			
T_ext	H_ext	G_r	W_s	T_int	H_int	Heating S2	Heating S1	Ventilation S2	Ventilation S1
MIN	MOY	MIN	MOY	MOY	MOY	MAX		MIN	
10	00	10	00	00	00	1	0	0	0
MOY	MOY	MOY	MOY	MOY	MIN	MOY		MIN	
00	00	00	00	00	10	0	1	0	0
MAX	MOY	MAX	MIN	MAX	MOY	MIN		MOY	
11	00	11	10	11	00	0	0	0	1
MAX	MIN	MAX	MIN	MAX	MIN	MIN		MAX	
11	10	11	10	11	10	0	0	1	0

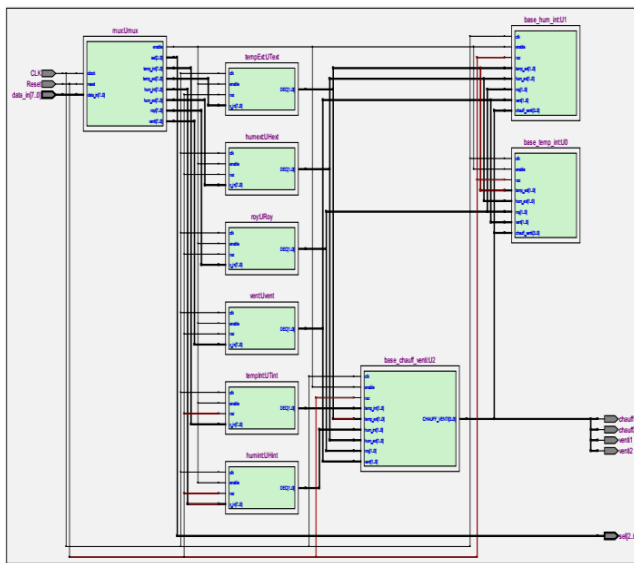


Fig. 10. RTL schematics of neuro-fuzzy controller

VI. CONCLUSION

The current work focuses on the application of neuro-fuzzy control of a greenhouse internal climate. It successfully demonstrated the performance through co-simulation by using ANFIS and ModelSim. This implementation accurately reproduces the theoretical behavior of the system, thus is

ready to be used. The future work will be destined to improve the design of our work including the number and the type of inputs membership functions.

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MR Brain Real Images Segmentation Based Modalities Fusion and Estimation Et Maximization Approach

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Abstract—With the development of acquisition image techniques, more data coming from different sources of image become available. Multi-modality image fusion seeks to combine information from different images to obtain more inferences than can be derived from a single modality. The main aim of this work is to improve cerebral IRM real images segmentation by fusion of modalities (T1, T2 and DP) using estimation et maximization Approach (EM). The evaluation of adopted approaches was compared using four criteria which are: the standard deviation (STD), entropy of information (IE), the coefficient of correlation (CC) and the space frequency (SF). The experimental results on MRI brain real images prove that the adopted scenarios of fusion approaches are more accurate and robust than the standard EM approach

Keywords—component; Data fusion; Segmentation; Estimation and Maximization; MRI images

I. INTRODUCTION

In last decades, biomedical and medical image processing has become one of the most challenging fields of image processing and pattern recognition.

Brain segmentation consists of separating the different tissues: gray matter (GM), white matter (WM), cerebrospinal fluid (CSF) and probably abnormal (tumor) tissue.

The aim of segmentation of MRI Brain images is to: Study anatomical structure, Identify region of interest: locate tumor, others abnormalities, measure tissue size (to follow the evolution of tumor) and help in treatment planning prior to radiation therapy (radiation dose calculation).

However, the segmentation of MRI Brain images has remained a challenge in image segmentation. And this is due to partial volume effects, motion (patient movement, blood circulation and respiration), the existence of image noise, the presence of smoothly varying intensity in-homogeneity, the fact that different anatomical structures may share the same tissue contrast and large amounts of data to be processed. For these and others many approaches have been studied, including Methods based edge [1][2][3], methods based region [4][5], Methods based on thresholding [6][7], methods based artificial neural networks [8], data fusion methods [9], Markov random field methods [10] and hybrid Methods [11][12][13].

Segmentation process also helps to find region of interest in a particular image. The main goal is to make image more simple and meaningful. Estimation and Maximization approach (EM) is a unsupervised classification algorithm. EM algorithm has been developed by Dempster et al. in 1977 [14] and it has been used in several fields, in particular in images segmentation. The main idea of this approach is to approximate an image histogram by a set of Gaussian functions. So, the problem is how to estimate the parameters of the different Gaussian to approaching best the histogram. This estimation is done based on maximum likelihood. The EM approach provides probabilities of membership in different tissue class. Thus, EM can be used to model the partial volume effects, where a pixel may contain multiple tissue classes.

The data fusion, in imaging, is used mainly on radar images, satellite images, and aerial images. Recently, it is also applied in medical image. The increasing diversity of the medical image acquisition techniques motivated recent years much research aimed at developing models increasingly effective data fusion. Indeed, medical imaging, it may happen that no images available alone does not contain sufficient information. On the other hand the medical community entrusts each image type to an expert who has a partial diagnosis of the modality of his specialty and specialists exchange experiences and this confrontation comes the final diagnosis.

In this paper, the contribution is mainly to propose an architecture of a information fusion system guided by the prior knowledge and based on Estimation and Maximization approach to segment human MRI real Brain images.

The organisation of the paper is as follows. In section 2 the Estimation and Maximization approach of segmentation is reviewed and in section 3 describes briefly data fusion. Section 4 present a complete description of proposed segmenting approach using data fusion, where each step of the algorithm is developed in detail. Section 5 illustrates the obtained experimental results and discussions and section 6 concludes this paper.

II. ESTIMATION AND MAXIMIZATION TECHNIQUE

Estimation and Maximisation approach is one is a classical and unsupervised method, it has been proposed by Dempster in 1977 [14]. The idea of EM approach is inspired from a well

know approach frequently used in statistic to describing distributions which is approximation using a combination. In this case, a no know distribution is estimate per a sum of parametric distributions. So the problem become to determinate the parameters of each distribution and its weight poids. In image segmentation, the EM approach is used to approximate an image histogram by a set of Gaussian distributions. So, the problem is how to estimate the parameters of the different Gaussian to approaching best the shape of histogram. Maximum likelihood is used for this estimation.

For each pixel, its intensity is x_i . The intensity distribution of each tissue class is approximated by a Gaussian G_k of mean μ_k and σ_k covariance matrix. Let π_{ki} the priori-probability membership to a class k for each pixel i and R_{ki} posteriori-probabilities membership to a class k calculated for each pixel i which are the searched labels.

EM algorithm has three main phases: Initialization phase, Estimation phase and maximization phase which are described as follows:

Initialization phase: initialize R_{ki} to the value given by the π_{ki}

Estimation phase: calculus of classes parameters μ_k , σ_k using the flowing expressions:

$$\mu_k = \frac{\sum_{i=1}^N R_{ki} x_i}{\sum_{i=1}^N R_{ki}} \quad (1)$$

$$\sigma_k = \frac{\sum_{i=1}^N R_{ki} x_i(x_i - \mu_k)(x_i - \mu_k)^T}{\sum_{i=1}^N R_{ki}} \quad (2)$$

Maximization phase: update of posterior probabilities R_{ki} as follows:

$$R_{ki} = \frac{\pi_{ki} G_k(x_i)}{\sum_{i=1}^k \pi_{ki} G_k(x_i)} \quad (3)$$

Repeat the two phases: estimation and maximization until convergence.

III. DATA FUSION

Information fusion is to combine information (often imperfect and heterogeneous) from multiple sources to obtain better complete global information, to improve decision making and make better act. The terms "information" (numeric or symbolic) and "sources" cover many possibilities. In the same way, the notion of improvement depends wholly on the application.

Information fusion has evolved considerably in recent years in various fields, especially in vision and robotics, information sources have increased (sensors, a priori information, generic knowledge ... etc.). In general, each source of information is imperfect, it is important to combine several to get a better understanding of the all of the system.

MRI is a powerful tool to improve clinical diagnosis because it can provide various information in the form of image intensities related to the anatomy through a variety of excitation sequences (for example: T1, T2, and PD).

The proposed fusion involves the aggregation MRI images from different acquisition techniques. Data to be combined are so homogeneous, and depending on the type of image acquisition will provide more or less pronounced contrast between tissues or between parenchyma and pathology. One of the main interests of the fusion will be to exploit in particular the complementarity between the different images. Many applications can benefit from this technique. These include:

The detection of tumour regions:

MRI provides easy assess tumour extension, especially when contrast media are used. With certain acquisition techniques, the specificity is also greater in some cases to distinguish between tumor and oedema. The whole point is going to reside in a combination of these techniques with a more anatomical acquisition (weighted T1 type) to measure the tumor extension.

Quantification of brain tissue volumes

Because of its anatomical accuracy and variety of acquisition techniques, MRI is used to assess the distribution of different brain tissues following several contrasts. The volume quantification of these tissues is clinically fundamental to the study of many pathologies that affect the white matter, gray matter or cerebrospinal fluid, or simply for the measurement of volumes in healthy subjects. Information fusion can be doing at three conceptual levels corresponding to three types of information:

- Data fusion: it is essentially to marry low-level information such as primitives, in order to make information less noisy than that obtained with a single source of information.
- Decision fusion: it performs the combination of sophisticated information (numeric or symbolic) that can be considered as proposals for a decision.
- Models fusion: in this case, different approaches are set apart to fill imperfections affecting each of them independently.

A general information fusion problem can be stated in the following terms : given L sources S_1, S_2, \dots, S_L representing heterogeneous data on the observed phenomenon, take a decision d_i on an element x , where x is higher level object extracted from information, and D_i belongs to a decision space $D = \{d_1, d_2, d_3, \dots, d_n\}$ (or set of hypotheses). In numerical fusion methods, the information relating x to each possible decision d_i according to each source S_j is represented as a number M_{ij} having different properties and different meanings depending on the mathematical fusion framework. In the centralized scheme, the measures related to each possible decision i and provided by all sources are combined in a global evaluation of this decision, taking the form, for each i : $M_i = F(M_{i1}, M_{i2}, M_{i3}, \dots, M_{in})$, where F is a fusion operator. Then a decision is taken from the set of $M_i, 1 \leq i \leq n$. in this scheme, no intermediate decision is taken and the final decision is issued at the end of the processing chain. In decentralized scheme decisions at intermediate steps are taken with partial information only, which usually require a difficult control or arbitration step to diminish contradictions and conflicts [15][16].

The three-steps fusion can be therefore described as:

Modeling of information: in a common theoretical frame to manage vague, ambiguous knowledge and information imperfection. In addition, in this step the M_{ij} values are estimated according to the chosen mathematical framework.

Combination: the information is then aggregated with a fusion operator F . This operator must affirm redundancy and manage the complementarities and conflicts.

Decision: it is the ultimate step of the fusion, which makes it possible to pass from information provided by the sources to the choice of a decision d_i . [17]

I. Bloch [18] classified these operators in three classes defined as:

- Context independent and constant behaviour operators (CICB);
- Context independent and variable behaviour operators (CIVB);
- Context dependent operators (CD).

IV. METHODOLOGY

Segmentation of brain images can separate different brain structures and detect possible pathologies, namely brain tumors. A good segmentation helps the doctor for making a final decision before his surgery. The main applications of the segmentation are morphometry, functional mapping and surface or volume visualization. Morphometry is the quantitative measurement of the positions, shapes and sizes of brain structures. It requires prior segmentation of these structures, and can identify, understand and follow the progression of diseases such as Alzheimer's or different tumors

The figure 1 shows the implementation of the proposed approach with its various stages:

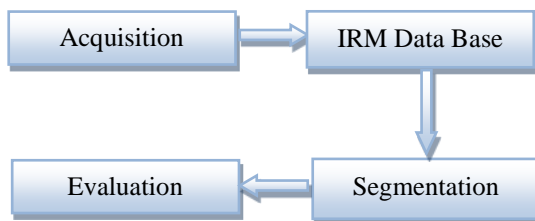


Fig. 1. Diagram of the various steps of the analysis system MRI images

A. Acquisition:

MRI Brain images are obtained by a Magnetic resonance imaging (MRI). Examination performed on a machine of high field 1.5 T according to the sequences:

- Axial and Sagittal T1
- Axial T2 * Flair and diffusion.
- Coronal T2.
- Examination with and without gadolinium injection.

These MRI images are of different sections (axial, sagittal, and coronal) of healthy and pathological subjects. They are grouped into several sections.

B. IRM DataBase:

The format of images is the format DICOM (Digital Imaging and Communication in Medicine). This latter is a file used by most of the manufacturers of medical imaging; this standard was issued by the ACR (American College of Radiology) in association with the NEMA (National Electrical Manufacturers Association). The DICOM format is a file containing the image and patient data compressed (patient name, exam type, hospital, examination date, type of acquisition...etc.). To validate the proposed segmentation algorithms, a real database is used. These images are encoded in the DICOM format size 256x256 pixels. These images are grouped into several sections. Each image DICOM used has the following details:

- Format : 'DICOM'
- Color Type : 'grayscale'
- Modality : 'MR'
- Manufacturer: 'GE MEDICAL SYSTEMS'
- Institution Name: 'Medical Imaging Center of M'sila (DR S. F. Ghabbane)'
- Study Description: 'CEREBRAL'
- Series Description: 'FL:A/3-pl T2* FGRE S'
- Slice Thickness: 5
- Repetition time: 55.500
- Magnetic Field Strength: 15000
- Echo Time: 2.1000
- Spacing Between Slices: 10
- Spatial Resolution: 1.8750
- Flip Angle: 0
- Pixel pacing:[2x1 double]

The modality T1 is used to distinguish the different tissues such as: gray matter (GM), white matter (WM) and cerebrospinal fluid (CSF). However, The T2 modality do not allow to distinguish the GM from WM but highlight lesions and CSF.

C. Segmentation

Normally MRI Brain image can be classified in three classes: gray matter (GM), white matter (WM), cerebrospinal fluid (CSF). Each region has a certain gray level, for the T1 modality, the WM region has the gray level which tend to white one, the CSF region has the gray level which tend to black one and the grey level of GM region is between the both. The process of segmentation is done with EM using modalities fusion to separating the different tissues of MRI Brain images.

For MRI images fusion, context-based conjunctive operators are chosen because in the medical context, both images were supposed to be almost everywhere concordant, except near boundaries between tissues and in pathologic areas.

In addition, the context-based behaviour allowed to taking into account these ambiguous but diagnosis-relevant areas. Then four operators of this class are retained, three of them are introduced in [18][19][20]:

$$\text{OP1: } \pi_T(v) = \min(\pi_T^{T1}(v), \pi_T^{T2}(v)) + 1-h \quad (4)$$

$$\text{OP2: } \pi_T(v) = \max\left(\frac{\min(\pi_T^{T1}(v), \pi_T^{T2}(v))}{h}, 1-h\right) \quad (5)$$

$$\text{OP3: } \pi_T(v) = \min\left(1, \frac{\min(\pi_T^{T1}(v), \pi_T^{T2}(v))}{h}\right) + 1-h \quad (6)$$

with :

$$h = 1 - \sum_{v \in \text{Image}} |\pi_T^{T1}(v) - \pi_T^{T2}(v)| / |\text{Image}| \quad (7)$$

$$\text{OP4: } \pi_T(v) = \frac{\pi_T^{T1}(v), \pi_T^{T2}(v)}{2} \quad (8)$$

The general algorithm for the EM using modalities fusion approach can be formulated as follows:

- 1) Set the parameters of the algorithm: C: number of cluster, ε : convergence error.
- 2) For each image of section j Xj: j=1...SC, which SC is the section number.
- 3) Segmentation of each image section of each modality (T1, T2, PD) using EM provide posteriori-probabilities membership Rjt, t=(T1, T2, PD), which t is the modality.
- 4) Use one operator fusion OPi(t1,t2) then the output is membership matrix Rj.
- 5) Assign all pixels to clusters by using the maximum membership value of every pixel.

D. Evaluation :

In addition to visual analysis, a quantitative evaluation is used on the above experimental results by different fusion algorithms. The selected quantitative criterions are standard deviation (SD), entropy (EN), spatial frequency (SF) and coefficient correlations (CC).

Standard deviation (SD): standard deviation is the square root of the variance, the variance of an image reflects the degree of dispersion among the grayscale values and the average value of gray levels. The larger the value is, the better fusion results are obtained.

$$SD = \sqrt{\frac{\sum_{i=0}^{N-1} \sum_{j=0}^{M-1} F(i,j)}{M}} \quad (9)$$

Entropy (EN): Entropy can effectively reflect the amount of information in certain image. The larger the value is, the better fusion results are obtained [21]:

$$EN = \sum_{i=0}^{L-1} P_f(i) \log_2 P_f(i) \quad (10)$$

Where P_{ii} is the normalized histogram of the fused image to be evaluated, L is the maximum gray level for a pixel in the image.

Spatial frequency (SF): Spatial frequency can be used to measure the overall activity and clarity level of an image. Larger SF value denotes better fusion result [21]:

$$SF = \sqrt{RF^2 + CF^2} \quad (11)$$

With

$$RF = \sqrt{\left(\frac{1}{M(N-1)}\right) \sum_{i=0}^{M-1} \sum_{j=0}^{N-2} (F(i, j+1) - F(i, j))^2} \quad (12)$$

And

$$CF = \sqrt{\left(\frac{1}{N(M-1)}\right) \sum_{i=0}^{M-2} \sum_{j=0}^{N-1} (F(i+1, j) - F(i, j))^2} \quad (13)$$

Coefficient correlation (CC): Coefficient correlation can show similarity in the small structures between the original and reconstructed images. Higher value of correlation means that more information is preserved [21]:

$$CC = \frac{\sum_{j=1}^N \sum_{i=1}^M (x_{i,j} - \mu(A))(x'_{i,j} - \mu(B))}{\sqrt{\sum_{j=1}^N \sum_{i=1}^M (x_{i,j} - \mu(A))^2 (x'_{i,j} - \mu(B))^2}} \quad (14)$$

Where $\mu(A)$ and $\mu(B)$ are the mean value of the corresponding dataset.

V. EXPERIMENTAL RESULTS

The proposed approach Estimation and maximization using modalities fusion have been tested on real MRI brain images to certify their efficiency. It was acquired in medical imaging center of M'sila (DR S. F. Ghadbane). These MRI images are of different sections (axial, sagittal, and coronal) of healthy and pathological subjects of size (256×256). These images are grouped into 26 sections. The format of images is the format DICOM (Digital Imaging and size Communications in Medicine).

Examples of real images MRI present in this paper are extracted on axial sections of three modalities (T1, T2 and PD). The healthy images are segmented in four parts (c=4): background, white matter WM, gray matter GM and cerebrospinal fluid CSF. For tumoral subject, the images are segmented in five classes: background, WM, GM, CSF and the tumor. Standard deviation (SD), entropy (EN), spatial frequency (SF) and coefficient correlations (CC) are used to compare the performance of the adopted techniques for segmentation of MRI brain images.

To evaluate the performance of the proposed approach, two slices are presented for each adopted fusion operator and each fusion system. And obtained results are compared to those of EM. The two slices are: slice 16 from the healthy subject (Figure 2) and slice 22 from the tumor one (Figure 4) with three modalities (T1, T2 and PD). Figure 2, 3, 4 and 5 present for each example the segmented image using the EM and EM using system fusion ((T1, T2) (T1, PD), (T2, PD) and (T1, T2, PD)) with each operator(OP1, OP2, OP3 and OP4). Tables I, II, III, IV and V present for each example and for EM and EM using system fusion with each operator the detailed quantitative evaluation: (standard deviation (SD), entropy (EN), spatial frequency (SF) and coefficient correlations (CC)).

Table I and IV show that modality T2 provide the best segmentation. The use of modalities fusion has improved segmentation in terms of evaluation criteria (Table II, III and V). Segmentation by modalities fusion depends on modalities themselves and the used fusion operator: for example for the min operator (OP1) the best combination is T1 with T2.

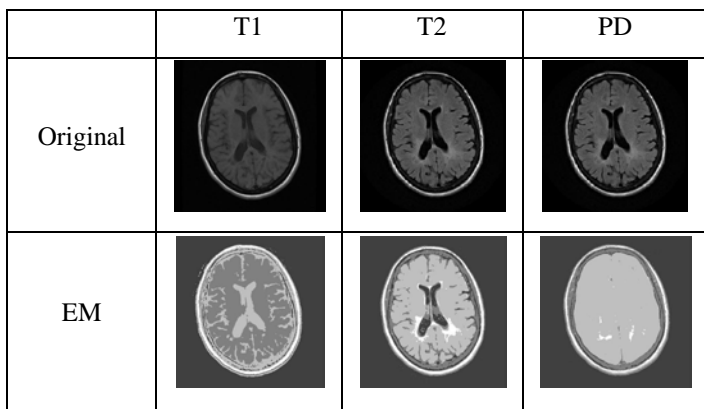


Fig. 2. Slice 16 from healthy MRI brain images and corresponding segmented image using EM algorithm for $c=4$

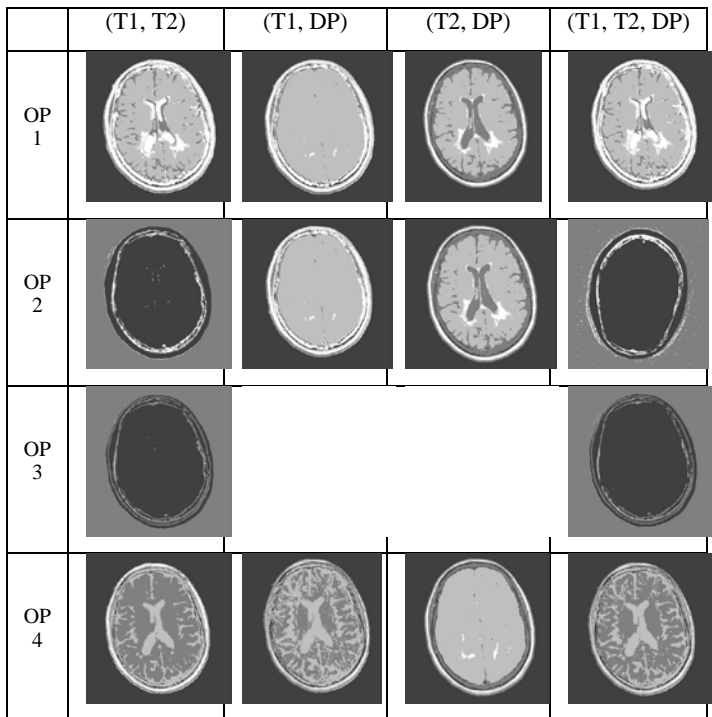


Fig. 3. The segmented results of the Slice 16 from healthy MRI brain images using the estimation and maximization algorithm using the four fusion operator and the four fusion system ($c=4$)

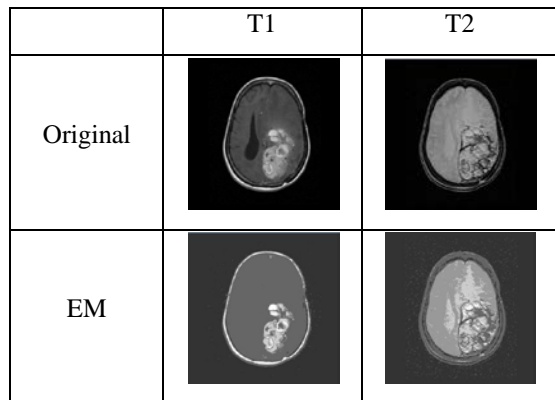


Fig. 4. Slice 22 from tumor MRI brain images and corresponding segmented image using EM algorithm for $c=5$

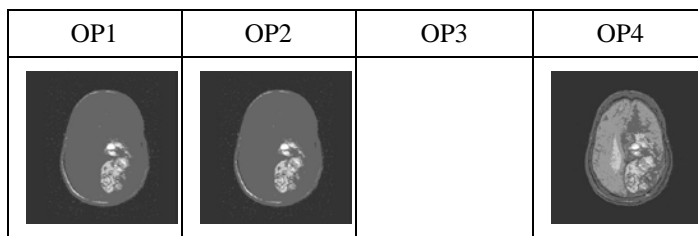


Fig. 5. The segmented results of the Slice 22 from tumor MRI brain images using the estimation and maximization algorithm using the four fusion operator and the fusion system (T1, T2) with ($c=5$)

TABLE I. EXPERIMENTAL RESULTS USING EM FOR THE SLICE 16 FROM HEALTHY MRI BRAIN

	CC	EN	SF	STD
T1	0.957	1.468	26.661	10.085
T2	0.990	1.430	18.548	10.320
DP	0.992	1.365	15.512	10.514

TABLE II. EXPERIMENTAL RESULTS USING EM WITH THE FOUR FUSION OPERATOR AND THE FUSION SYSTEM (T1, T2) FOR THE SLICE 16 FROM HEALTHY MRI BRAIN

		CC	STD	SF	EN
Op1	T1	0.932	10.953	35.400	1.537
	T2	0.892			
Op2	T1	-0.976	10.357	32.027	1.113
	T2	-0.925			
Op3	T1	-0.840	10.251	16.567	1.004
	T2	-0.868			
Op4	T1	0.959	10.047	27.925	1.453
	T2	0.848			

TABLE III. EXPERIMENTAL RESULTS USING EM WITH THE FOUR FUSION OPERATOR AND THE FUSION SYSTEM (T1, T2, PD) FOR THE SLICE 16 FROM HEALTHY MRI BRAIN

		CC	STD	SF	EN
Op1	T1	0.923	10.975	34.846	1.546
	T2	0.891			
	DP	0.942			
Op2	T1	-0.964	10.430	34.468	1.182
	T2	-0.918			
	DP	-0.965			
Op3	T1	-0.830	10.235	17.5468	1.034
	T2	-0.862			
	DP	-0.932			
Op4	T1	0.945	9.974	29.026	1.386
	T2	0.881			
	DP	0.943			

TABLE IV. EXPERIMENTAL RESULTS USING EM FOR THE SLICE 22 FROM TUMOR MRI BRAIN

	CC	EN	SF	STD
T1	0.991	1.237	13.808	8.619
T2	0.984	1.445	18.661	9.063

TABLE V. EXPERIMENTAL RESULTS USING EM WITH THE FOUR FUSION OPERATOR AND THE FUSION SYSTEM (T1, T2) FOR THE SLICE 22 FROM TUMOR MRI BRAIN

		CC	STD	SF	EN
Op1	T1	0.975	8.520	13.047	1.174
	T2	0.927			
Op2	T1	0.975	8.520	13.047	1.174
	T2	0.927			
Op3	T1	0	15.968	0	0
	T2	0			
Op4	T1	0.949	8.796	18.166	1.346
	T2	0.981			

The fusion using the three modalities with the operator min (OP1) offer the best segmentation with a rate of correlation $CC=0.923, 0.891$ and 0.942 standard deviation $STD=10.975$ spatial frequency $SF=34.846$ and information entropy $EN=0.546$. The segmentation using the third operator provide white image for the tumour subject and for the fusion T1 with DP and T2 with DP. Fusion operator that has the best performance is the fourth one which confirms the qualitative improvement. As it can be seen, the adopted approach is very good and allows a good segmentation (Figure 3 and 5).

It can see that the main tissues (GM, WM and CSF) of the brain images are well separated for healthy subject and tumor region is well extracted for pathological subject. Fusion modalities and EM algorithm perform equally well in terms of the quality of image segmentation and leads to a good visual result.

VI. CONCLUSION

Segmentation of medical images is still a vast field of research. The aim of this work is devoted to brain tissue segmentation from magnetic resonance images, in order to extract the tumor and also all other tissues (white matter, gray matter and CSF) by using of estimation and maximization with modalities fusion approach. This aggregation was performed by fusion operators that model doctor daily analysis confronted heterogeneous clinical data. The proposed approach Estimation and maximization using data fusion has been tested on real MRI brain images (healthy and pathological) to certify their efficiency. Experimental results show that: modalities fusion improves the segmentation of brain images. The fusion operators min and mean are the best for the segmentation of brain images and can deliver satisfactory performance to separating the different parts of an MRI brain real image. Further research is needed to improve the proposed approach. At level of modelling, other numeric or symbolic information can be integrated to increase the mass of available knowledge and at the combination one to design adaptive fusion operators for the combination of data in the medical field.

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A Leveled Dag Critical Task Firstschedule Algorithm in Distributed Computing Systems

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Abstract—In distributed computing environment, efficient task scheduling is essential to obtain high performance. A vital role of designing and development of task scheduling algorithms is to achieve better makespan. Several task scheduling algorithms have been developed for homogeneous and heterogeneous distributed computing systems. In this paper, a new static task scheduling algorithm is proposed namely; Leveled DAG Critical Task First (LDCTF) that optimizes the performance of Leveled DAG Prioritized Task (LDPT) algorithm to efficiently schedule tasks on homogeneous distributed computing systems. LDPT was compared to B-level algorithm which is the most famous algorithm in homogeneous distributed systems and it provided better results. LDCTF is a list based scheduling algorithm which depends on sorting tasks into a list according to their priority then scheduling one by one on the suitable processor. LDCTF aims to improve the performance of the system by minimizing the schedule length than LDPT and B-level algorithms.

Keywords—Task scheduling; Homogeneous distributed computing systems; Precedence constrained parallel applications; Directed Acyclic Graph; Critical path

I. INTRODUCTION

Distributed systems have emerged as powerful platforms for executing parallel applications. A distributed system can be defined as a collection of computing systems that appears to its users as a single system, these systems collaborate over a network to achieve a common goal [1]. There are two types of distributed systems; homogeneous (in which processors are identical in capabilities and functionality) and heterogeneous (in which processors are different).

In distributed computing environment, an application is usually decomposed into several independent and/or interdependent sets of cooperating tasks. Dependent tasks are represented by a Directed Acyclic Graph (DAG). DAG can be defined as a graph consists of a set of vertices or nodes and a set of edges $G(V, E)$ in which each node represents a task and each edge represents a communication between two tasks (the two tasks are dependent on each other). The computation cost of the task is represented by a weight associated with each node and the communication cost between two tasks is represented by a weight associated with each edge. The communication cost between two dependent tasks is considered to equal zero if they are executed on the same processor. Figure 1 shows an example of a simple task graph (DAG). In the Figure, t_0 is called predecessor (or parent) of t_2

and t_2 is called successor (or child) of t_0 . The edge between t_0 and t_2 means that t_2 can start execution only after t_0 finishes its execution. Efficient task scheduling of application tasks is essential to achieve high performance in parallel and distributed systems. The basic function of task scheduling is to determine the allocation of tasks to processors and their execution order in order to satisfy the precedence requirements and obtain minimum schedule length (or make span) [2]. Task-scheduling algorithms are broadly classified into two basic classes: static and dynamic. In static scheduling, the characteristics of an application, such as execution time of tasks and data dependencies between tasks are known in advance (during compile time before running the application). In dynamic scheduling, some information about tasks and their relations may be undeterminable until run-time [3].

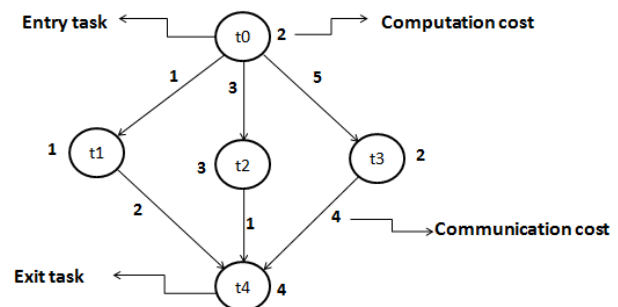


Fig. 1. Example of a DAG

Over the past few decades, researchers have focused on designing task scheduling algorithms for homogeneous and heterogeneous systems with the objective of reducing the overall execution time of the tasks. Topcuoglu et al. [2] have presented HEFT and CPOP scheduling algorithms for heterogeneous processors. Luiz et al. [4] have developed lookahead-HEFT algorithm, which look ahead in the schedule to make scheduling decisions. Eswari, R. and Nickolas, S. [5] have proposed PHTS algorithm to efficiently schedule tasks on the heterogeneous distributed computing systems. Rajak and Ranjit [6] have presented a queue based scheduling algorithm called TSB to schedule tasks on homogeneous parallel multiprocessor system. Ahmed, S.G.; Munir, E.U.; and Nisar, W. [7] have developed genetic algorithm called PEGA that provide low time complexity than standard genetic algorithm (SGA). Xiaoyong Tang; Kenli Li; Renfa Li; and Guiping Liao [8] have presented a list-scheduling algorithm called HEFD for

heterogeneous computing systems. Nasri, W. and Nafti, W. [9] have developed a new DAG scheduling algorithm for heterogeneous systems that provide better performance than some well-known existing task scheduling algorithms.

In homogeneous distributed systems, researchers have developed many heuristic task-scheduling algorithms such as ISH [10], ETF [11], DLS [12], MH [13], B-level [14] and some heuristics that depend on the critical path such as MCP [15], FCP [16], and CNPT [17]. Among these algorithms, B-level provides the best performance in terms of schedule length, speedup, and efficiency. LDPT (Leveled DAG Prioritized Task) algorithm [18] was compared to B-level algorithm which is the most famous algorithm in homogeneous distributed systems and it provided better results.

In this paper, the problem of scheduling precedence constrained parallel tasks on homogeneous physical machines (PMs) is addressed. A new static scheduling algorithm called LDCTF is proposed. The goal of LDCTF is to optimize the performance of LDPT [18] algorithm in order to provide better system performance. LDCTF is a list scheduling algorithm. It depends on dividing the DAG into levels then sorting tasks in each level into a list according to their priority and finally, picking tasks from the list one by one to schedule it on the suitable processor. LDCTF is compared to LDPT and B-level algorithms and it provided better results in terms of schedule length, speedup, and efficiency.

The remainder of this paper is organized as follows. Section II provides an overview of the related work algorithm. The proposed algorithm is discussed in section III. Section IV presents performance evaluation results of the proposed algorithm. Finally, conclusion and future work is reviewed in section V.

II. LDPT ALGORITHM

LDPT is a list based scheduling algorithm. It depends on dividing the DAG into levels with considering the dependency conditions among tasks in the DAG. The algorithm has two phases: (1) Task prioritization phase, (2) Processor selection phase. LDPT algorithm depends on giving a priority to each task as shown in Figure 2 then; scheduling each task on one processor with taking into consideration the insertion-based policy. Figure 2 shows the pseudo code of LDPT algorithm.

III. LEVELED DAG CRITICAL TASK FIRST (LDCTF) ALGORITHM

LDCTF is a theoretical task scheduling algorithm. LDCTF, LDPT, and B-level algorithms are applied on Standard Task Graph STG [19] as a bench mark, and it was found that LDCTF algorithm is more efficient than LDPT and B-level algorithms.

LDCTF is a list based scheduling algorithm. It depends on dividing the DAG into levels with considering the dependency conditions among tasks in the DAG then, applying the Min-min method [20] which means calculating the minimum completion time (MCT) for each task on all processors then selecting the task with the lowest MCT to schedule. The algorithm has two phases: (1) Task prioritization phase, (2) Processor selection phase.

```
Generate the DAG
Divide the DAG into levels according to their
communicated dependency
Sort the constructed levels according to dependency
ordering
Sort tasks according to [their computation costs then their
direct communication of its next level] in descending order
While there are unscheduled levels do
    While there are unscheduled tasks do
        For each level do
            Find the task with the highest computation cost
            If there are tasks have equal computation cost
                Then
            Choose the task with the highest communication cost with
            its Childs in next level
        End if
        Find the processor that minimizes the Earliest
        Start Time of the selected task
        Assign the task to the selected processor
        Remove the selected task from the list
        Repeat
        Until all tasks are scheduled
    End for each
End while
```

Fig. 2. LDPT algorithm [18]

A. Task prioritization phase:

In this phase, the critical path [2] is calculated for the DAG (critical path is the longest path from the entry task to the exit task in the graph) then, the DAG is divided into levels and the tasks in each level will be sorted into a list based on their priority. The priority for each task is given as follow:

1) First, the critical task (task located on the critical path) in each level will have the highest priority.

2) Then, the expected Earliest Finish Time (EFT) is calculated for the other tasks in the same level and the task with the lowest EFT will have the highest priority. If tow tasks have equal EFT value then, the task with the lowest task number will have the highest priority. EFT of a task tion processor p_j is computed as follow:

$$EFT(t_i, P_j) = w_{i,j} + EST(t_i, P_j) \text{-----} (1)$$

3) Finally, tasks in each level are sorted into the list in ascending order according to their EFT value.

B. Processor Selection Phase:

In this phase, the tasks are picked from the list one by one and assigned to the processor that will minimize the earliest start time of the task, with taking into consideration the insertion-based policy. The insertion policy means that if there is an idle time slot on the processor between two already scheduled tasks and it was enough for executing the task, then the task is assigned on that processor in this idle slot without violating precedence constraints. In other words, a task can be scheduled earlier if there is a period of time between two tasks already scheduled on processor (P), where P runs idle. If two processors provide the same start time for the task then, the task is assigned to the first processor that will minimize the EST of it. The Earliest Start Time of a task n_i on a processor P_j is defined as:

$$EST(n_x, P_m) = \max\{T_{Available}(P_m), \max\{AFT(n_i) + c_{x,i}\}\} \quad (2)$$

Where $T_{Available}(P_m)$ is the earliest time at which processor P_m is ready. $AFT(n_i)$ is the Actual Finish Time of a task n_i (the parent of task n_x) on the processor P_m . $c_{n,i}$ is the communication cost from task n_i to task n_x , $c_{k,i}$ equal zero if the predecessor task t_k is assigned to processor P_m . For the entry task, $EST(n_{entry}, P_m) = 0$. Figure 3 shows the pseudo code of LDCTF algorithm.

C. Case Study

Consider the DAG shown in Figure 4; assume the system has two processors (P_0, P_1). The critical path for the DAG in Figure 4 is $(t_0, t_1, t_3, t_6, t_8)$. Table 1 shows the computation cost for each task. Both LDPT and LDCTF algorithms generate a list of tasks that shows the execution order of them. Table 2 shows the lists generated by LDPT and LDCTF algorithms. For LDCTF algorithm the critical task in each level will be scheduled first as shown in table 2. Figure (5.a, 5.b) shows the Gantt chart generated by LDPT and LDCTF algorithms respectively. Both algorithms assign the selected task to the processor that minimizes the start time (EST) of it. For example, in Figure 5.a, the EST for task t_2 on p_0 is 5 and the EST for t_2 on p_1 is 4, so the task t_2 is scheduled on p_1 . In Figure 5.b, the same manner is followed with taking into consideration the insertion-based policy. From Figure 5, it is shown that the schedule length (the finish time of the last task scheduled from the DAG) resulted from LDPT and LDCTF algorithms is 25, and 23 unit of time respectively.

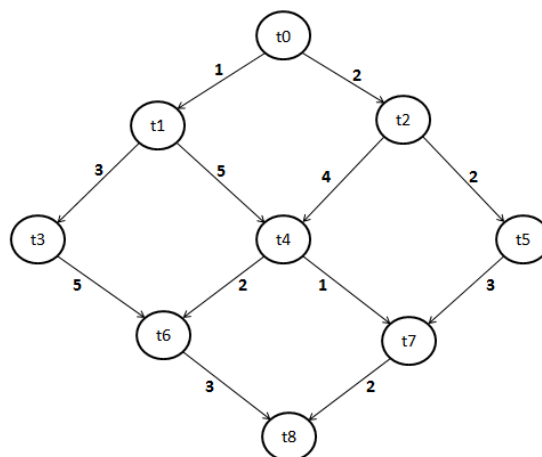


Fig. 4. Sample DAG

TABLE I. COMPUTATION COST

Task	Computation Cost
t_0	2
t_1	3
t_2	1
t_3	4
t_4	3
t_5	5
t_6	2
t_7	4
t_8	6

Figure 5 depicts the Gantt chart generated by LDPT and LDCTF algorithms. From the Figure, it is shown that the schedule length generated from LDPT algorithms is 25 unit time while the schedule length generated from LDCTF algorithm is 23 unit time. In case of LDCTF, we observe that there is less periods in which processors are idle than LDPT. According to this result, the overall running time of the application will be decreased and the efficiency of the system will be improved.

TABLE II. TASK LISTS FOR LDPT AND LDCTF ALGORITHMS

Execution	LDPT	LDCTF
1	t_0	t_0
2	t_1	t_1
3	t_2	t_2
4	t_5	t_3
5	t_3	t_5
6	t_4	t_4
7	t_7	t_6
8	t_6	t_7
9	t_8	t_8

```

Generate the DAG
Calculate the critical path for the DAG
Divide the DAG into levels according to their communicated dependency
Sort the constructed levels according to dependency ordering
Determine the critical task for each level
While there are unscheduled levels do
    While there are unscheduled tasks do
        For each level do
            For each task in level
                Calculate the expected EFT of selected task
            End for
            Sort level tasks in Tasks Ordered List according to
            1-Critical task
            2-Expected EFT in ascending order
            If there are tasks have equal Earliest Finish Time
                Then
                    Choose the task with the lowest task number
                End if
            For each task in Tasks Ordered List
                Find the processor that minimizes the Earliest Start Time
                of the selected task
                Assign the task to the selected processor
                Remove the selected task from the list
            Repeat
            Until all tasks are scheduled
        End for each
    End while
    
```

Fig. 3. Leveled DAG Critical Task First algorithm

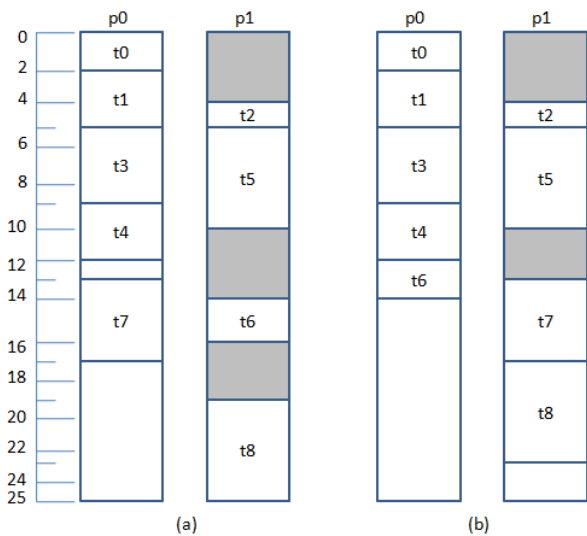


Fig. 5. The schedules generated by (a) LDPT algorithm (b) LDCTF algorithm for sample DAG

IV. RESULTS AND PERFORMANCE EVALUATION

A. Simulation Environment

To evaluate the performance of LDCTF algorithm, a simulator had been built using visual C# .NET 4.0 on machine with: Intel(R) Core(TM) i3 CPU M 350 @2.27GHz, RAM of 4.00 GB, and the operating system is window 7, 64-bit. To test the performance of LDPT and LDCTF algorithms, a set of randomly generated graphs is created by varying a set of parameters that determines the characteristics of the generated DAGs. These parameters are described as follows: DAG size (n: the number of tasks in DAG). Density (d: the probability of existence edge between n_i in level j and n_x in the next level level $j+1$ for DAG. Where, $i, x=1,2,\dots, N$, and N is the number of tasks, $j=1, 2,\dots, T$, and T is the number of levels in DAG). With six different numbers of processors varying from 2, 4, 8, 16, 32 and 64 processors. For each number of processors, six different DAG sizes have been used varying from 10, 20, 40, 60, 80 and 100 nodes.

B. Evaluation Metrics

The most important metrics for evaluating performance of scheduling algorithms are schedule length, speed up, and efficiency. Schedule length is the maximum finish time of the last task (exit task) scheduled from the DAG.

$$\text{Schedule length} = \text{Max}(AFT(\text{next})) \text{-----}(3)$$

Where $AFT(\text{next})$ is the actual finish time of the exit task. Speedup is defined as the ratio of the schedule length generated from executing the application on one processor to the schedule length generated from executing the application on multiple parallel processors.

$$\text{Speed up} = \frac{\text{Min}_{j \in P} [\sum_{i \in V} w(i,j)]}{SL} \text{-----}(4)$$

Where $w(i,j)$ means the weight of task n_i on processor p_j and SL means the schedule length. Efficiency is the inverse of speed up.

$$\text{Efficiency} = \frac{\text{speedup}}{\text{Number of processors}} \text{-----}(5)$$

C. Experimental Results

The schedule length generated by LDPT and LDCTF algorithms is shown in Figure 6, 7, 8, 9, 10, 11 for 10, 20, 40, 60, 80, 100 tasks respectively and the results are recorded in table 3. According to the results, the schedule length is decreased that will minimize the running time of the application. The improvement ratio in schedule length is (2.75%). Figure 12, 13, 14, 15, 16, 17 show a comparative study of the speed up of LDPT and LDCTF algorithms in case of 2, 4, 8, 16, 32, 64 processors respectively. Table 4 shows the speedup results of LDPT and LDCTF algorithms. From the results, we can see that the improving ratio in speed up is (3.2%). Table 5 shows the efficiency results of LDPT and LDCTF algorithms. From Figure 18, 19, 20, 21, 22, 23 we can see that LDCTF is more efficient than LDPT algorithms with improving ratio (1.9%). The schedule length generated by B-level, LDPT, and LDCTF algorithms is shown in Figure 24, 25, 26, 27, 28, 29. Figure 30, 31, 32, 33, 34, 35 shows a comparative study of the speed up of B-level, LDPT, and LDCTF algorithms. The efficiency results of B-level, LDPT, and LDCTF algorithms are shown in Figure 36, 37, 38, 39, 40, 41.

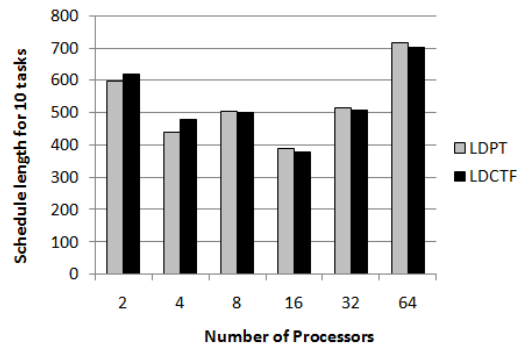


Fig. 6. Schedule length for 10 tasks

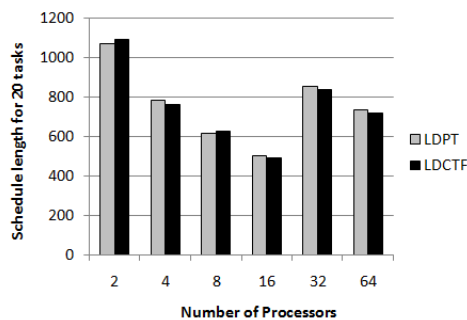


Fig. 7. Schedule length for 20 tasks

Figure 6, 7, 8, 9, 10, 11 depict the schedule length versus number of tasks with varying number of processors 2, 4, 8, 16, and 32 processors. It is shown that the schedule length in case of applying LDCTF algorithm is less than LDPT algorithm. This is because the periods in which processors are idle in case of LDCTF are less than LDPT algorithm.

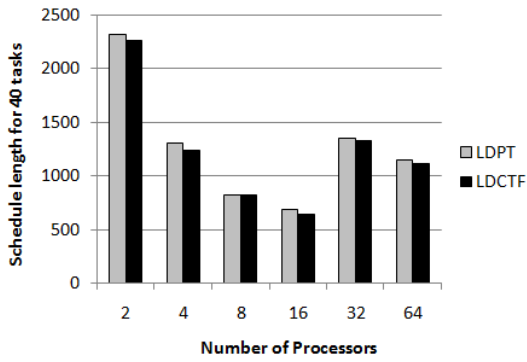


Fig. 8. Schedule length for 40 task

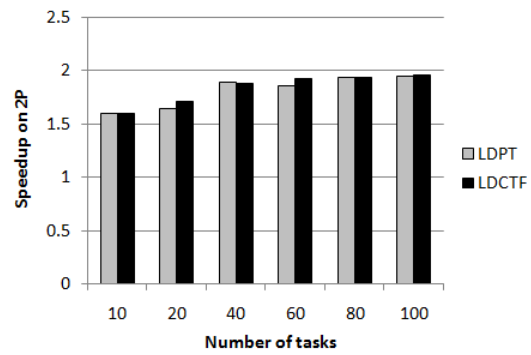


Fig. 12. Speedup on 2 processors

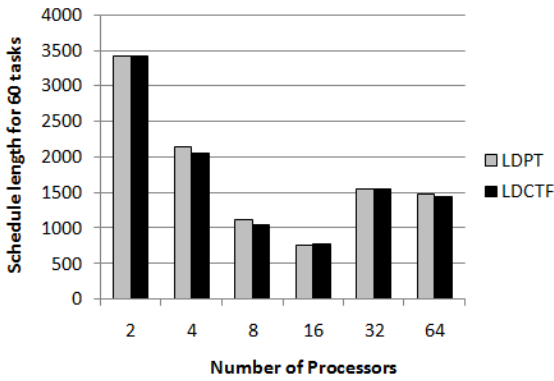


Fig. 9. Schedule length for 60 task

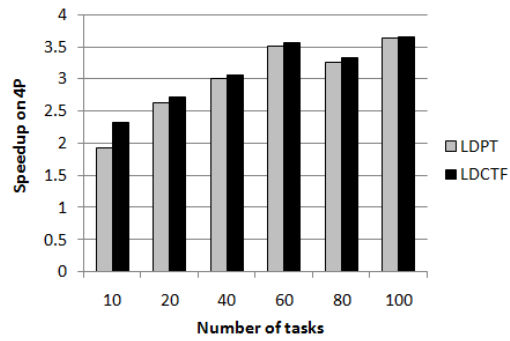


Fig. 13. Speedup on 4 processors

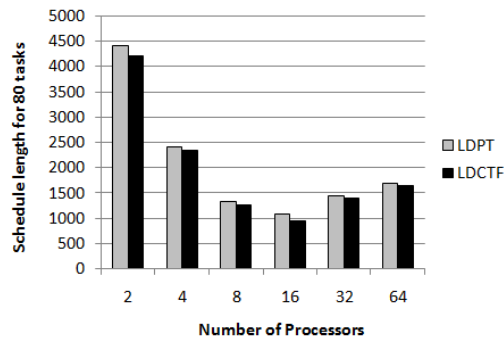


Fig. 10. Schedule length for 80 task

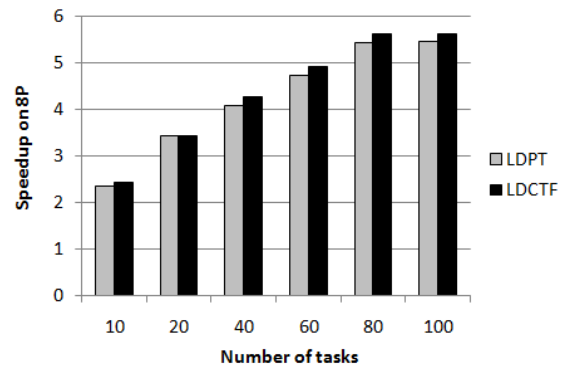


Fig. 14. Speedup on 8 processors

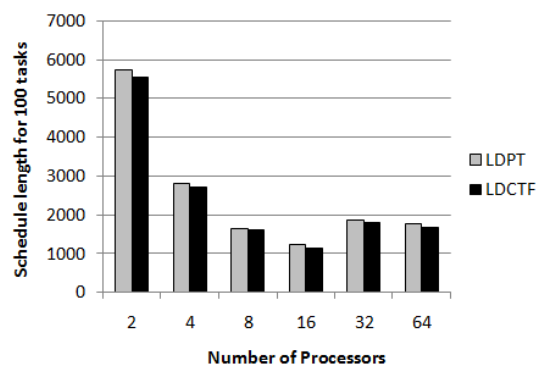


Fig. 11. Schedule length for 100 task

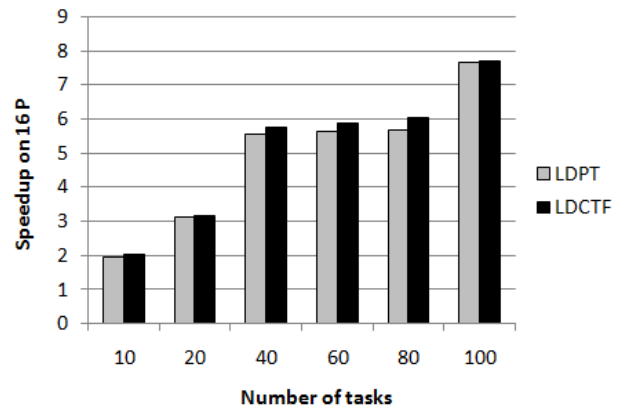


Fig. 15. Speedup on 16 processors

TABLE III. SCHEDULE LENGTH RESULTED FROM LDPT AND LDCTF ALGORITHMS

Number of tasks	2 processor		4 processor		8 processor		16 processor		32 processor		64 processor	
	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF
10	598	619	438	481	503	501	390	377	516	508	718	701
20	1070	1091	784	763	615	629	504	490	856	838	733	720
40	2319	2265	1304	1236	827	818	691	637	1348	1325	1151	1117
60	3427	3430	2141	2055	1114	1053	765	783	1557	1544	1469	1441
80	4408	4204	2403	2349	1330	1261	1072	955	1442	1403	1690	1642
100	5734	5551	2821	2724	1654	1604	1218	1124	1858	1806	1755	1666

TABLE IV. SPEEDUP RESULTED FROM LDPT AND LDCTF ALGORITHMS

Number of processors	10 tasks		20 tasks		40 tasks		60 tasks		80 tasks		100 tasks	
	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF
2	1.597101	1.604076	1.646914	1.710256	1.889304	1.884024	1.861015	1.925334	1.941921	1.938553	1.951211	1.957603
4	1.926573	2.32981	2.625984	2.722449	3.01626	3.05922	3.509259	3.567059	3.261836	3.330391	3.648107	3.658973
8	2.339703	2.421978	3.42637	3.42637	4.071071	4.258639	4.71821	4.918985	5.438854	5.611392	5.46696	5.615385
16	1.965385	2.015779	3.126603	3.172358	5.570213	5.749634	5.649007	5.88856	5.671501	6.030069	7.672043	7.707561
32	1.965385	2.015779	4.363834	4.402198	4.718067	4.892601	5.672481	5.883417	7.498576	7.695907	7.63357	7.985985
64	2.066937	2.088115	2.782361	3	4.601407	4.774939	5.362133	5.529858	6.167453	6.291099	6.755631	7.024131

TABLE V. EFFICIENCY RESULTED FROM LDPT AND LDCTF ALGORITHMS

Number of processors	10 tasks		20 tasks		40 tasks		60 tasks		80 tasks		100 tasks	
	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF	LDPT	LDCTF
2	61.59624	63.94366	55.79577	56.94572	52.51707	53.2254	52.22963	53.03263	51.05813	51.52983	51.24817	51.62996
4	38.20755	40	34.6574	34.95531	30.53475	30.58641	28.24712	28.19847	26.86957	26.90683	27.48918	27.43998
8	40.21632	40.21632	23.46253	24.34109	17.45132	17.8457	15.93451	16.25871	14.7612	15.02103	14.97816	15.22779
16	32.63062	32.63062	20.13423	23.2318	12.44079	13.06407	10.4242	10.83525	8.181933	8.51226	8.97423	9.236988
32	35.92677	35.92677	14.66799	14.66799	10.17964	11.00299	7.395091	7.854315	7.148139	7.394627	5.821122	6.124305
64	37.26462	37.26462	18.53759	18.53759	8.372093	8.372093	5.611434	5.593789	4.385628	5.140007	4.311791	4.666996

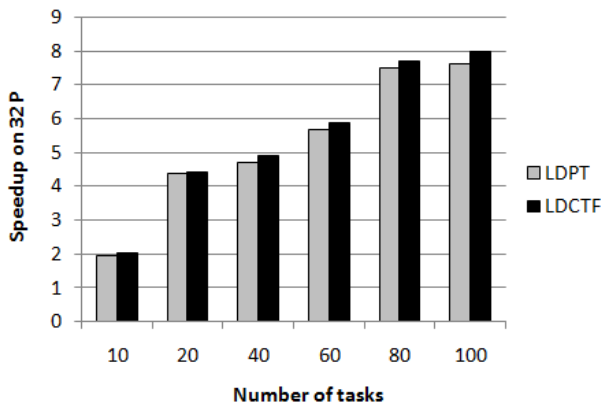


Fig. 16. Speedup on 32 processors

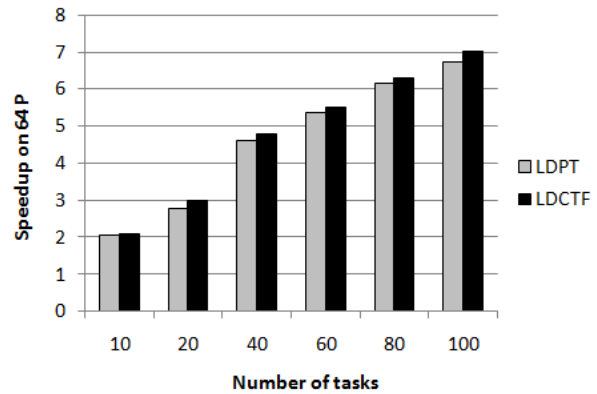


Fig. 17. Speedup on 64 processors

Figure 12, 13, 14, 15, 16, 17 depict speedup versus number of processors with varying number of tasks (20, 40, 60, 80, 100). It is shown that LDCTF algorithm provides better speed up than LDPT algorithm. This is because in case of LDCTF algorithm, all processors have finished the execution of tasks earlier than LDPT algorithm.

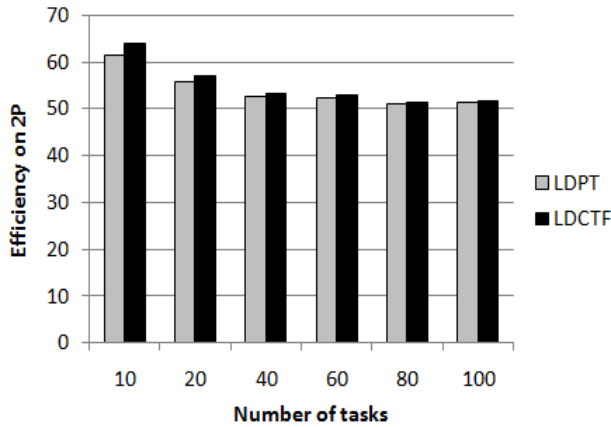


Fig. 18. Efficiency on 2 processors

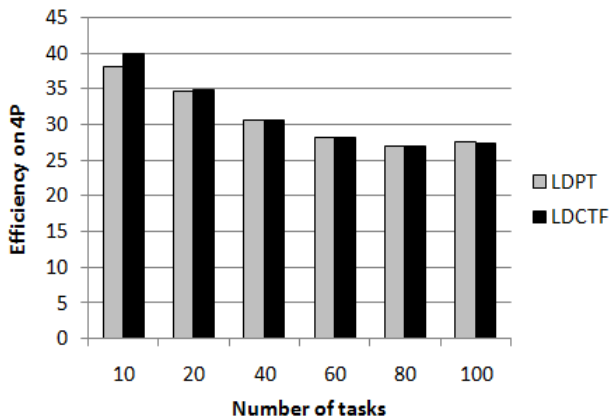


Fig. 19. Efficiency on 4 processors

Figure 18, 19, 20, 21, 22, 23 depict efficiency versus number of processors with varying number of tasks (20, 40, 60, 80, 100). It is shown that LDCTF algorithm is more efficient and provides better performance than LDPT algorithm. Most of processor elements have been perfect utilized in our algorithm because of the communication among tasks is not affected in algorithm breadth procedures.

Figure 24, 25, 26, 27, 28, 29 depicts the schedule length versus number of tasks with varying number of processors 2, 4, 8, 16, 32, and 64 processors. It is shown that the schedule length in case of applying LDCTF algorithm is less than LDPT and B-level algorithms.

Figure 30, 31, 32, 33, 34, 35 depicts speedup versus number of processors with varying number of tasks (10, 20, 40, 60, 80, 100). It is shown that LDCTF algorithm provides better speed up than LDPT and B-level algorithms. This is because in case of LDCTF algorithm, all processors have finished the execution of tasks earlier than LDPT and B-level algorithms.

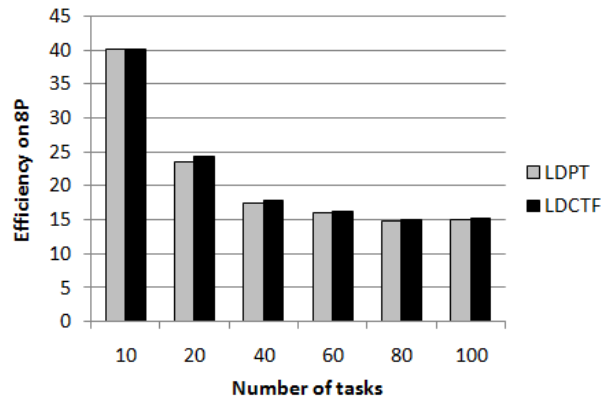


Fig. 20. Efficiency on 8 processors

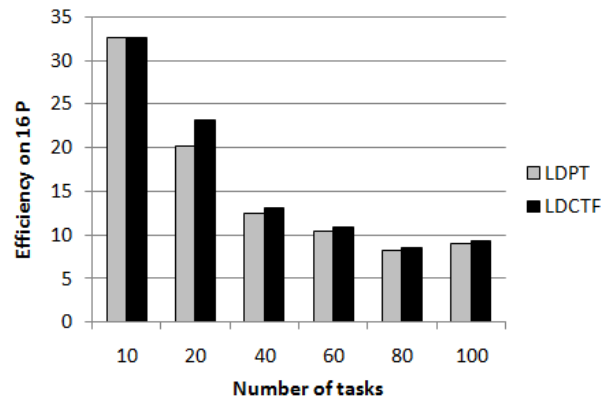


Fig. 21. Efficiency on 16 processors

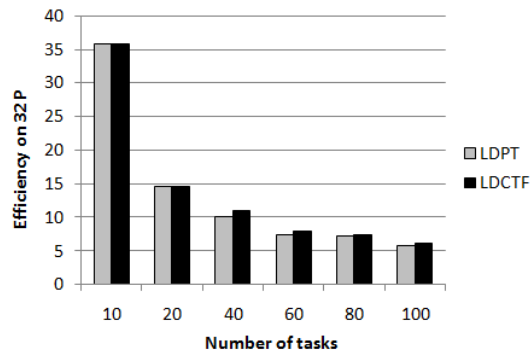


Fig. 22. Efficiency on 32 processors

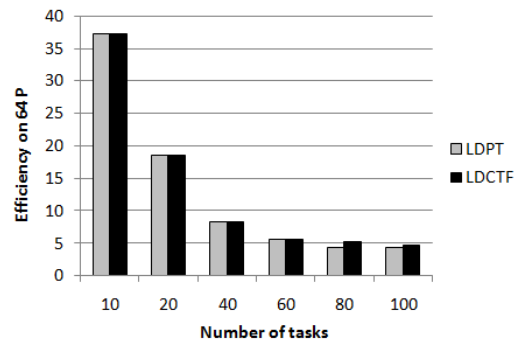


Fig. 23. Efficiency on 64 processors

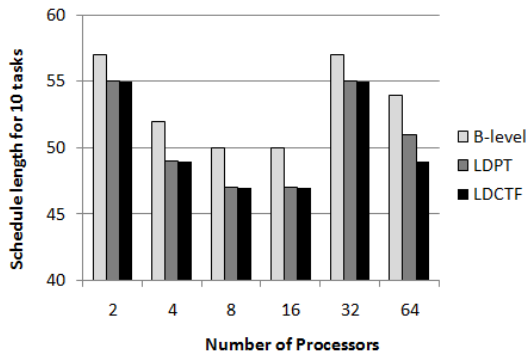


Fig. 24. Schedule length for 10 tasks

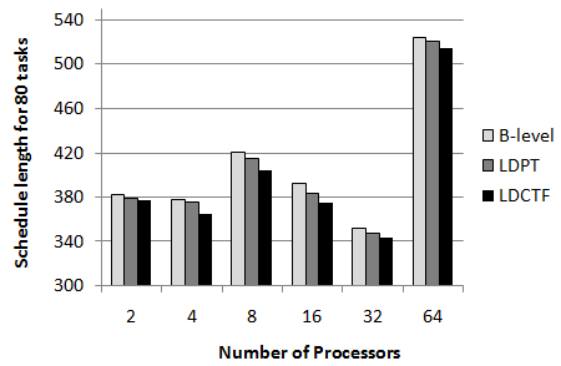


Fig. 28. Schedule length for 80 tasks

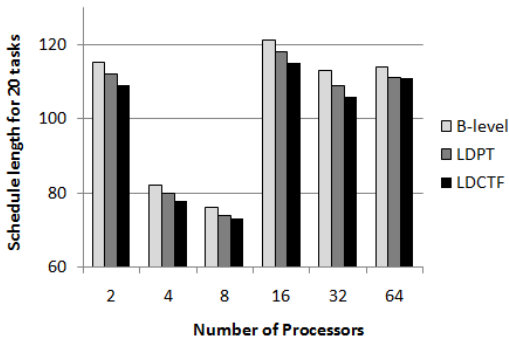


Fig. 25. Schedule length for 20 tasks

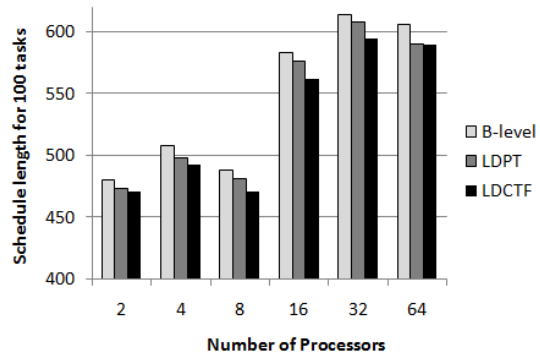


Fig. 29. Schedule length for 100 tasks

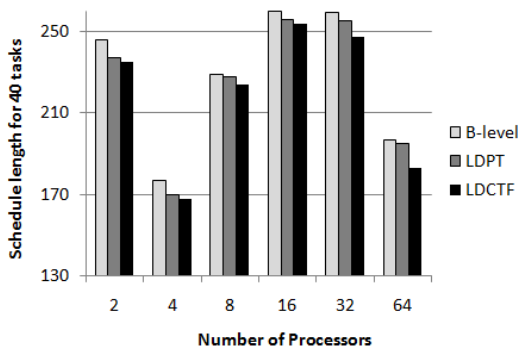


Fig. 26. Schedule length for 40 tasks

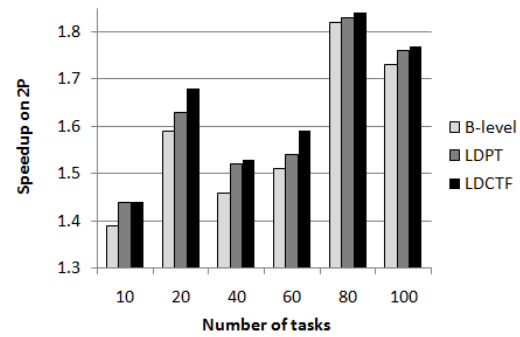


Fig. 30. Speedup on 2 processors

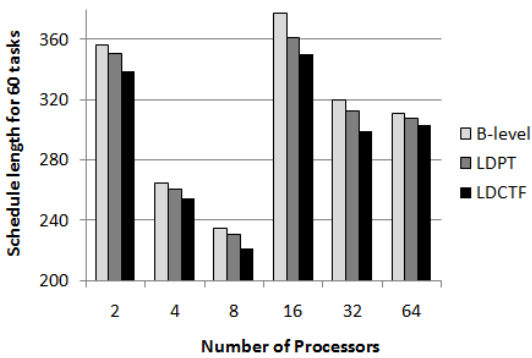


Fig. 27. Schedule length for 60 tasks

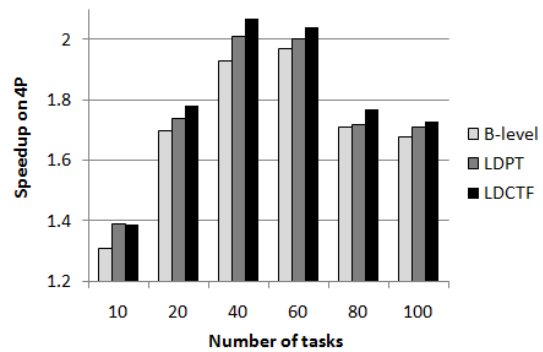


Fig. 31. Speedup on 4 processors

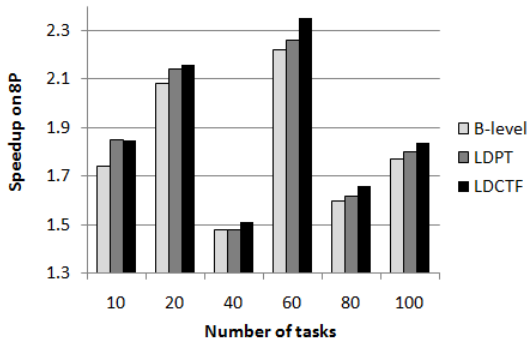


Fig. 32. Speedup on 8 processors

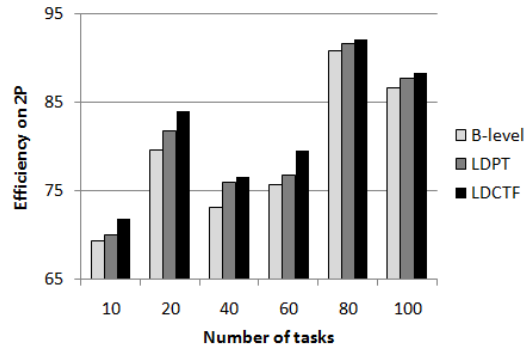


Fig. 36. Efficiency on 2 processors

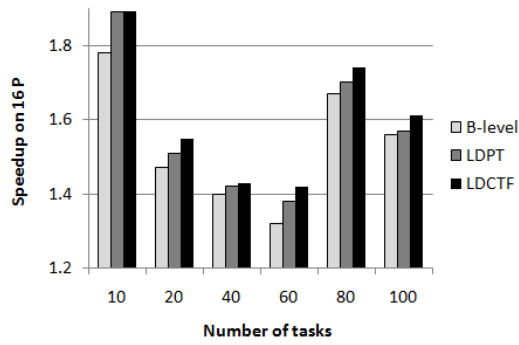


Fig. 33. Speedup on 16 processors

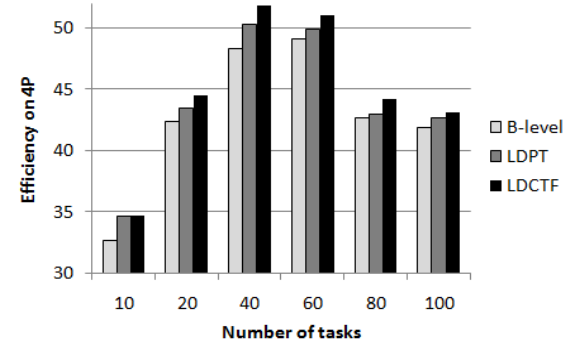


Fig. 37. Efficiency on 4 processors

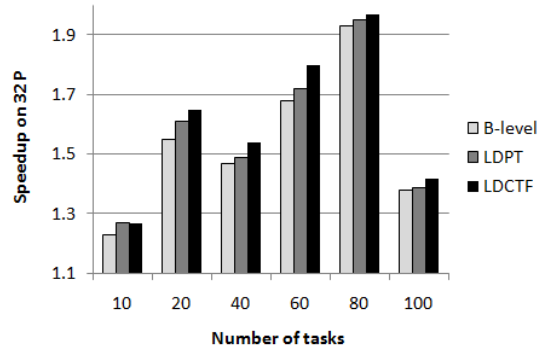


Fig. 34. Speedup on 32 processors

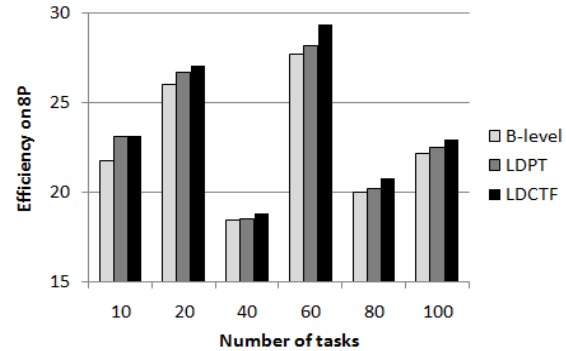


Fig. 38. Efficiency on 8 processors

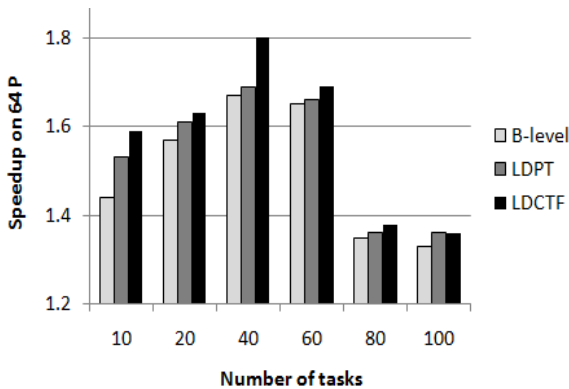


Fig. 35. Speedup on 64 processors

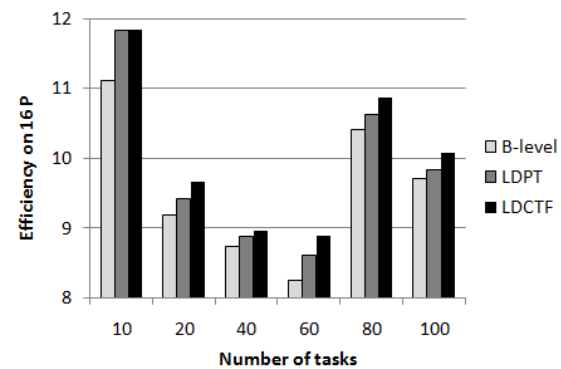


Fig. 39. Efficiency on 16 processors

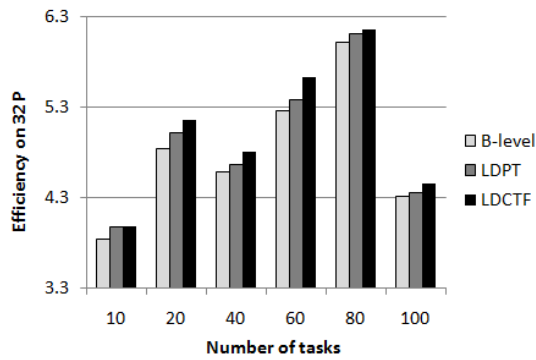


Fig. 40. Efficiency on 32 processors

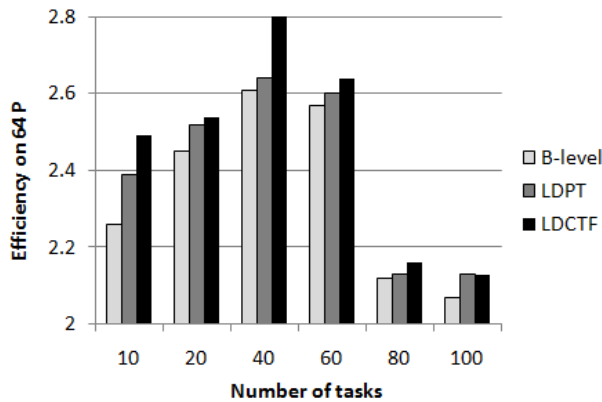


Fig. 41. Efficiency on 64 processors

Figure 36, 37, 38, 39, 40, 41 depicts efficiency versus number of processors with varying number of tasks (20, 40, 60, 80, 100). It is shown that LDCTF algorithm is more efficient and provides better performance than LDPT and B-level algorithms.

D. Discussion of Results

First, LDCTF algorithm is compared to LDPT algorithm and it provided better results in terms of schedule length, speed up, and efficiency. This is because in case of LDCTF, the critical path is taken into account and the critical task will be scheduled first in each level. This means that the task with the highest computation and communication cost will be scheduled first resulting in minimum schedule length, higher speed up, and higher efficiency.

Finally, LDCTF is compared to B-level algorithm and it provided better results in terms of schedule length, speed up, and efficiency. This is because B-level algorithm depends on paths idea and this will increase the communication overhead during assigning tasks on processors. On the other side, LDCTF algorithm depends on levels idea that will minimize the communication overhead during assigning tasks on processors. Another reason is that B-level algorithm must calculate the b-level value for each task before scheduling so that, the arithmetic calculation in LDCTF is less than B-level algorithm which leads to minimize the complexity factor.

V. CONCLUSION AND FUTURE WORK

In this paper, a new static scheduling algorithm (LDCTF) is developed for homogeneous distributed computing systems. The performance of LDCTF algorithm is compared with LDPT algorithm. LDCTF is evaluated for different DAGs and found to be giving better results than LDPT algorithm in terms of schedule length, speed up, and efficiency with improving ratio 2.75%, 3.2%, and 1.9% respectively.

The performance of LDCTF is also compared with B-level and LDPT algorithms and found to be giving better results in terms of schedule length, speed up, and efficiency. LDCTF, LDPT, and B-level algorithms are applied on Standard Task Graph STG as a bench mark, and it was found that LDCTF algorithm is more efficient than LDPT and B-level algorithms. The future scope of the idea can be as follows:

- In this paper LDCTF algorithm is applied on Directed Acyclic Graph (DAG). In the future it can be applied on Directed Cyclic Graph (DCG).
- LDCTF can be applied on Heterogeneous Distributed Computing Systems (HDCS).
- LDCTF can be applied in a dynamic strategy instead of static strategy.
- Finally, duplication technique can be applied with LDCTF algorithm to minimize the communication overhead.

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No-Reference Perceived Image Quality Algorithm for Demosaiced Images

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Abstract—Visual image quality assessment (IQA) plays a key role in every multimedia application, as end user to it is a human-being. Real time applications demand no reference (NR) IQA, due to unavailability of the reference image. Today, most of the perceived/visual NR-IQA algorithms developed are for distortions like blur, ringing, and blocking artifacts. Very few are available for color distortions. Visible color distortions, such as false color, and zipper are produced in the demosaiced image due to incorrect interpolation of missing color values. In this paper, state of the art zipper and false color artifact quantification algorithms, general purpose NR-IQA algorithms are evaluated for visual quality assessment of demosaiced images. Separate NR-IQA algorithms are proposed for zipper and false color artifact quantification scores are then combined to obtain final quality score for demosaiced image. Zipper algorithm quantifies zipper artifact by searching for zipper pixels in an image, while false color algorithm finds correlation between local high frequency region's color planes to quantify false color.

Keywords—Demosaicing; Correlation; False color; Image quality; Regression; Zipper

I. INTRODUCTION

Image quality assessment plays key role in image processing pipeline for testing and validation. End user to every multimedia application is human being, so visual quality assessment is of great importance. Visual quality scores can be obtained either by conducting subjective experiment or by developing objective IQA algorithms. Conducting subjective experiment is very difficult and is not the real time solution. So many objective IQA algorithms, correlating well with subjective ratings, are developed. These objective IQA algorithms are categorized into full Reference (FR), reduced reference (RR), and no reference (NR) algorithms.

FR and RR algorithms require original image or partial information of it respectively for IQA. On the other hand NR algorithms assess quality without any reference image and only with distorted image in hand. For real time applications NR algorithms are needed as making reference image available is not always possible. Most of the NR-IQA algorithms developed are for distortions like, blur, and distortions due to compression algorithms. Few general purpose algorithms are also proposed but they are evaluated only for distortions like blur, white noise, and distortions due to JPEG, JPEG-2000.

Digital cameras capture real world scenes using a single sensor in order to reduce the size and cost of the camera. This single sensor is overlaid with a color filter array (CFA) that

restricts the sensor elements to storing only red (R), green (G), or blue (B) color information. A mosaic/bayer image is subsequently generated at the output of this sensor. Although various types of CFA patterns are available, the Bayer CFA pattern [1] is the one most commonly used. Figure 1a shows the Bayer CFA pattern, where green color information is sampled more because the human visual system is more sensitive to green color information.

Demosaicing process reconstructs full color image from this bayer image by interpolating missing color values. If interpolation for missing color values is incorrect color distortions like false color, zipper, water color, or grid pattern can appear in the demosaiced image. Of these artifacts, zipper and false color are the most common and appear in high frequency areas of the image. Misinterpolation in high frequency areas, owing to limited correlation between neighboring pixels, causes zipper and false color artifacts to appear in those areas. Zipper artifacts appear as on-off-on or off-on-off patterns, whereas false color artifacts appear as aberrant colors. Figures 1b and 1c show examples of zipper and false color artifacts respectively [2].

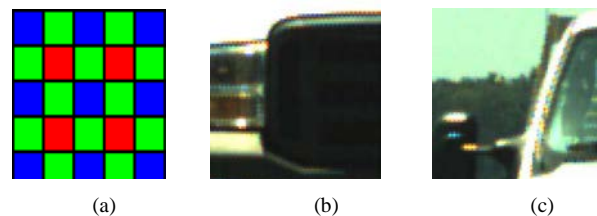


Fig. 1. (a) Bayer CFA pattern. (b) Zipper artifact. (c) False color artifact

Today, quality assessment of demosaiced image and testing of demosaicing algorithm is performed in the RGB color space or International Commission on Illumination's $L^* a^* b^*$ (CIELAB) color space with FR objective quality metrics such as color peak signal to noise ratio (CPSNR), color mean square error (CMSE), CIELAB color difference ΔE (CIELAB ΔE), or spatial extension of the CIELAB color difference ΔE (SCIELAB ΔE) [3].

As these metrics are FR, so they have limitation for real time application. Also, it is stated in the literature that FR color difference metrics in CIELAB space indicate perceived quality, but we obtained very less correlation (around 37%) between FR color difference metrics in CIELAB and subjective opinions.

Considering all the limitations of FR-IQA algorithms for demosaiced image quality assessment, NR-IQA algorithms are proposed in the literature but they are very few. Furthermore, no correlation is exhibited between their objective scores and subjective opinions.

In this paper, state of the art NR demosaicing IQA algorithms and general purpose NR-IQA algorithms are evaluated for visual quality assessment of demosaiced images. Separate novel visual NR-IQA algorithms are proposed for zipper and false color artifacts. These scores are then combined to obtain final visual quality score for demosaiced image. The remainder of this paper is organized as follows: Section 2 offers an overview of previous work. Section 3 presents proposed NR-IQA algorithm. Section 4 outlines the performance evaluation. Section 5 concludes the paper.

II. PREVIOUS WORK

NR-IQA algorithms are categorized in to distortion specific and general purpose ones. Distortion specific NR-IQA algorithms predict perceived quality of an image distorted with specific distortion [4-7], while general purpose algorithms predict perceived quality of an image for any type of distortion. General purpose algorithms like [8-9] are trained on human rated database, while general purpose algorithm like [10] do not require training also for visual quality assessment. Very few algorithms are reported for NR quality assessment of demosaiced image. Also these algorithms are not evaluated for their correlation with subjective ratings.

A. State of the Art NR-General Purpose Algorithms

General purpose algorithms quantify visual quality for any type of distortion. In this section very popular general purpose algorithms [8-10] are presented. General purpose algorithms [8-9] are trained on distortions from LIVE database [11] and general purpose algorithm [10] is not trained on any human rated database. Blind image quality index (BIQI) [8] general purpose algorithm is a two step framework for quality assessment. The two steps are image distortion classification based on a measure of how the wavelet statistics is modified, followed by quality assessment, using an algorithm specific to the decided distortion. Blind referenceless image spatial quality evaluator (BRISQUE) [9] general purpose algorithm uses scene statistics of locally normalized luminance coefficients to quantify possible losses of naturalness in the image due to the presence of distortions thereby leading to a holistic measure of quality. Natural Image Quality Evaluator (NIQE) general purpose algorithm do not require training on human ratings and is based on collection of quality aware statistical features from a corpus of natural undistorted images. The quality of the test image is expressed as the distance between multivariate Gaussian fits of the Natural Scene Statistic (NSS) features extracted from test image and the multivariate Gaussian model of the quality aware features extracted from the corpus of natural images.

B. State of the Art Algorithms for Zipper Artifact Quantification

Very popular state of the art zipper artifact quantification algorithms [12-13] are overviewed in this section.

Lu and Tan [12] proposed a FR algorithm for quantifying zipper artifacts that is currently used by most of the demosaicing algorithms for performance evaluation. Their algorithm is outlined below:

1) Find a pixel with minimum CIELAB color difference ΔE with respect to the center pixel in a 3×3 neighborhood in the original image (image having no demosaicing artifacts is referred to as original image here). Equation (1) gives the CIELAB color difference formula, where $l1$, $a1$, and $b1$ are the CIELAB color values for pixel $l1$ and $l2$, $a2$, and $b2$ are the CIELAB color values for pixel 2.

$$\Delta E^*_{ab} = \sqrt{(l2 - l1)^2 + (a2 - a1)^2 + (b2 - b1)^2} \quad (1)$$

2) Compute CIELAB ΔE^*_{ab} between the same pair of pixels in the demosaiced image.

3) Find the difference between ΔE^*_{ab} of the original and the demosaiced image; if this difference is greater than 2.3, then that pixel is affected by zipper.

4) Compute percentage of such pixels which is the zipper score for that demosaiced image.

Losson, Macaire, and Yang [13] proposed a directional alternation measurement algorithm for zipper artifact measurement. Their proposed algorithm is outlined below:

5) Compute the green color variance in the original image along the vertical as well as horizontal direction at the center pixel in the 3×3 neighborhood.

6) Consider the lowest green color variance direction to find the alternation amplitude at the center pixel. If the lowest variance direction is horizontal, then the alternation amplitude is given by (2). However, if the lowest variance direction is vertical, then the alternation amplitude is given by (3). The alternation amplitude value is always positive if a "high-low-high" or "low-high-low" pattern exists at the considered pixel or the center pixel in the local neighborhood.

$$\alpha^i = |g(i-1, j) - g(i, j)| + |g(i, j) - g(i+1, j)| + |g(i-1, j) - g(i+1, j)| \quad (2)$$

$$\alpha^j = |g(i, j-1) - g(i, j)| + |g(i, j) - g(i, j+1)| + |g(i, j-1) - g(i, j+1)| \quad (3)$$

7) Compare this alternation amplitude to the alternation amplitude at the same pixel in the demosaiced image. If the value obtained is greater, then the alternation amplitude of the green levels has been amplified by demosaicing.

C. State of the Art Algorithms for False Color Artifact Quantification

State of the art false color artifact quantification algorithms [14-15] are presented in this section.

A NR algorithm is proposed in [14] for quantifying false color. The algorithm proposed is as follows:

- 1) Use the Prewitt operator to find edge points in the Gaussian filtered image.
- 2) At each edge point, compute the difference between the G and R values.
- 3) Compute the median of the (G-R) values in a 5×5 neighborhood centered on the edge point.
- 4) Finally, compute the MSE between the median and difference between G and R values at the edge point over all edge points.
- 5) Repeat the above procedure for the G and B planes.
- 6) Compute the average of these MSE values for the (G, R) and (G, B) planes. This average value is the false color measure.

In [15] a FR algorithm for false color quantification is proposed, where the absolute difference between the pixel values of the demosaiced and corresponding original image is computed for the R, G, and B color planes, then the maximum color difference value among the three color planes is found. Subsequently, if that difference is greater than a specified threshold value, then that pixel is considered as being affected by the false color artifact. The percentage of such pixels is quantified as the false color score. Threshold value of 2.3 is used for implementation of this algorithm. All these state of the art zipper and false color algorithms are implemented in Matlab 7.

III. PROPOSED NR-IQA ALGORITHMS FOR DEMOSAICING ARTIFACTS

Proposed perceived NR-IQA algorithm for visual quality assessment of demosaiced images is presented in this section.

A. Proposed NR Algorithm for Zipper Artifact Quantification

Zipper artifacts predominantly appear near high frequency regions such as edges, and zipper pixels appear perpendicular to edge pixels. This fact is utilized for quantification of zipper artifacts. The proposed algorithm derived is outlined below:

- 1) Convert demosaiced image to gray image, compute gradient magnitude and gradient direction.
- 2) Use the Sobel operator to find edge pixels in the horizontal as well as vertical direction.
- 3) Consider a pixel in 3×3 neighborhood around each edge pixel as a zipper pixel, if it satisfies the following conditions:
 - a) It is not an edge pixel.
 - b) Its gradient magnitude is greater than gradient magnitude of the edge pixel.
 - c) It is perpendicular to edge pixel.
- 4) Repeat this process for all edge pixels to find zipper pixels.
- 5) Compute the ratio of the total number of zipper pixels to the total number of edge pixels, which is considered as zipper score for demosaiced image.

B. Proposed NR Algorithm for False Color Artifact Quantification

The high frequency details in an image are more affected by false color than smooth regions. Consequently, for false

color quantification, the correlation of the high frequency details of the green (G) plane to the high frequency details of the blue (B) and red (R) planes is found. G plane is used as reference because green color information is sampled more than red and blue colors in bayer image. The resulting algorithm proposed is outlined below:

- 1) Divide Demosaiced image in to blocks of 64×64 . Block size of 64 is selected as it corresponds to foveal region [6].
- 2) For each block
 - a) Obtain HH, HL, LH, and LL sub-bands for G, and R planes.
 - b) Obtain correlation between the HH, HL, and LH sub-bands of the G plane and the respective HH, HL, and LH sub-bands of the R plane.
 - c) Compute the average of these three correlation values, which is correlation value between block's high frequency details of G and R planes.
- 3) Repeat step 2 for all blocks.
- 4) Take average of block's high frequency details of G and R planes (obtained in step 2C), which is correlation value for high frequency regions of the G and R planes.
- 5) Repeat steps 2-4 for G and B planes.
- 6) Average out the correlation values obtained for high frequency regions of G&R (value obtained in step 4) and G&B planes to obtain proposed false color quantification score.

C. Proposed perceived NR-IQA Algorithm for Demosaiced Images

Zipper and false color scores obtained with proposed algorithms are combined, with non-linear equation as in (4), to obtain final quality score for demosaiced image. In (4) z & fc are zipper and false color scores respectively. Constant values in (4) were obtained with multiple non-linear regression.

$$\text{Quality Score} = c + c1 * z + c2 * fc + c3 * z^2 + c4 * fc^2 + c5 * f * z \quad (4)$$

Table 1 gives details of the regression statistics for the multiple nonlinear regression.

TABLE I. MULTIPLE NON LINEAR REGRESSION STATISTICS

Statistics	Values	Constant Values for (4)	
Multiple R	0.955562	C	37.98871
R Square	0.9131	C1	315.2318
Adjusted R Square	0.90032	C2	-200.859
F	71.45053	C3	-1009.65
Significance F	4.74E-17	C4	300.213
		C5	-213.677

IV. PERFORMANCE EVALUATION AND COMPARISON

For experimentation, forty demosaiced images, which were utilized for subjective experiment1 in [16] are used. The subjective scores for the images are made available from Brainard, one of the authors of [16]. In [16], ten high resolution images were acquired using a Kodak DCS200 camera and 50 mm lens under various daylight conditions. From each captured image, four demosaiced images were then obtained by applying Bilinear, Freeman1 [17], Bayesian1, and Bayesian2

[18], [19] demosaicing algorithms to the bayer image. Forty demosaiced images were thus obtained from the ten original high resolution images.

For performance evaluation four (IQA) performance parameters are computed: 1) Spearman rank order correlation coefficient (SROCC); 2) Kendall's rank order correlation coefficient (PLCC); and 4) Root mean squared error (RMSE). Nonlinear logistic mapping function as given in [20] is used to compute PLCC and RMSE. Tables 2 and 3 give performance evaluation and comparison of proposed zipper and false color algorithms respectively. Table 4 gives performance evaluation of general purpose NR-IQA algorithms on demosaiced images. Performance evaluation of proposed NR-IQA algorithm is given in table 5. Figure 2 gives scatter plot for proposed algorithm.

TABLE II. PERFORMANCE COMPARISON OF ZIPPER ALGORITHMS

Zipper Algorithms	Performance Parameters			
	PLCC	SROC C	KROC C	RMSE
Zipper algorithm in [12]	0.6439	0.3712	0.2662	23.50
Zipper algorithm in [13]	0.5443	0.3344	0.23	25.76
Proposed zipper algorithm	0.6730	0.6901	0.4874	22.72

TABLE III. PERFORMANCE COMPARISON OF FALSE COLOR ALGORITHMS

False Color Algorithms	Performance Parameters			
	PLCC	SROCC	KROCC	RMSE
False color algorithm in [14]	0.4224	0.4567	0.3204	27.84
False color algorithm in [15]	0.5637	0.4422	0.3101	25.40
Proposed false color algorithm	0.9555	0.8078	0.6056	9.05

TABLE IV. PERFORMANCE EVALUATION OF GENERAL PURPOSE ALGORITHMS FOR VISUAL QUALITY ASSESSMENT OF DEMOSAICED IMAGES

General Purpose Algorithms	Performance Parameters			
	PLCC	SROCC	KROCC	RMSE
BIQI [8]	0.5055	0.4261	0.2905	26.50
BRISQE [9]	0.3956	0.4548	0.3163	28.19
NIQE[10]	0.4838	0.5355	0.3783	26.86

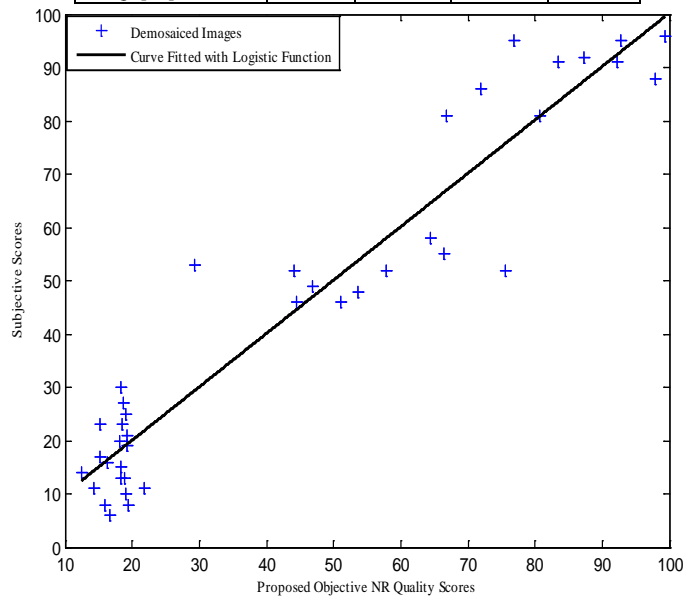


Fig. 2. Scatter plot of proposed NR-IQA algorithm for demosaiced images

TABLE V. PERFORMANCE EVALUATION OF PROPOSED NR-IQA ALGORITHM

Proposed NR-IQA Algorithm	Performance Parameters			
	PLCC	SROCC	KROCC	RMSE
	0.9552	0.8601	0.6770	9.0866

Statistical significance testing is also required in order to evaluate the performance of the IQA algorithm, if the number of tested images are less. So, we performed F-test and t-test between predicted and the actual subjective scores at 5% significance level. The F-test was used to check whether the difference between the variance of the predicted and actual subjective scores is statistically significant or not, whereas the t-test was used to check whether the difference between the means of the two is statistically significant or not. Table 6 gives details of the F-test and t-test for proposed NR-IQA algorithm for demosaiced images. Results in table 6 shows that proposed algorithm's scores are statistically significant as F & t statistic values are less than critical values.

TABLE VI. STATISTICAL SIGNIFICANCE TESTING

Statistical Parameters	Proposed NR-IQA Algorithm for Demosaiced Images
F-statistic	1.090163
F-critical one tail	1.704465
t-statistic	0.00011
t-critical two tail	1.990847036

V. CONCLUSION

Separate NR-IQA algorithms are proposed for quantification of zipper and false color artifacts then these scores are combined to obtain final visual quality score for demosaiced image. Experimental results points towards usability of proposed algorithm for visual quality assessment of demosaiced images. State of the art demosaicing artifact quantification algorithms as well as general purpose NR-IQA algorithms fail to quantify visual quality of demosaiced image. Statistical significance testing also confirms validity of proposed algorithm's scores. So, proposed algorithm can be utilized for testing and validation of demosaicing algorithm and for visual quality assessment of demosaiced image.

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Dynamic Clustering for Information Retrieval from Big Data Depending on Compressed Files

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Abstract—The rapid growth in the database data led to origination a large amount of data. So, it is still a big problem to access this data for answering user queries. In this paper a novel approach for aggregating the required data was proposed, this approach called dynamic clustering. Also, several retrieval methods were used for retrieving purposes. The dynamic clustering method is built clusters according to the user entries (queries). It has been applied to different compressed database files in different size and using different queries. The compressed database file is resulted from applying ICM (Ideal Compression Method) and best compressed algorithm (improved k-mean, k-mean with medium probability and k-mean with maximum gain ratio). The retrieval methods applied to original database file, compressed file and the cluster that result from implementing dynamic clustering algorithm and the results was compared.

Keywords—dynamic clustering; data retrieval methods; compression algorithm; ICM system; improved k-means algorithm and modified improved k-means algorithms

I. INTRODUCTION

Big Data interests with large-volume, complex, growing data sets in multiple and autonomous sources. Data storage and Data collection have become more complex. Big Data is now expanding quickly in all science and engineering domains, including physical, biological and biomedical sciences, etc. The most essential challenge for Big Data applications is to explore the large amount of data and extract useful information or knowledge for future actions [1]. The goals of data compression is the task of providing space on the hard drive, and reduce the use of bandwidth in the transmission network and transfer files quickly. Data compression purposes are to reduce the number of bits used to store or transmit data. Data Compression methods are divided into two types 1) lossless compression method and 2) lossy compression method [2]. In this paper we used Lossless data compression techniques. In lossless data compression, the integration of data is preserved without loss any information. In this paper the ICM system results are used to build the dynamic clusters.

The Data Mining is defined as an extraction of hidden information from large databases. It is a powerful advanced technology. It has great possibility helps the Libraries and information centers to focus on the most important information in their data warehouse. There are several techniques for data mining these are: 1) classification 2) clustering 3) prediction (regression) 4) decision trees 5) sequential patterns and 6) association rules [3]. In this paper

the clustering technique was used to solve the problem of accessing big data. The ICM system was used to specify the best compression algorithm and the results of the compression algorithm is used to build the dynamic clusters. Clustering technique is one of the most important data mining techniques. The aim of cluster analysis is to divide the data (objects, instances, items) into groups (clusters) so that items belonging to the same group are more similar than items belonging to distinct groups [2]. Also, the retrieval methods were used to answering user queries. The most frequently used operations on transactional databases is the data retrieval operation. Data retrieval means obtaining data from a database management system In order to retrieve the desired data the user specifies a set of criteria by a query. Then the Database Management System (DBMS) selects the required data from the database. There are several searching strategies for data retrieval these are keyword, Boolean operators, truncation, phrase searching, and search limiting, and nesting. All search strategies are based on comparison between the query and the stored documents [4]. The retrieved data may be stored in a file, printed, or viewed on the screen. In traditional database management systems, information retrieval is often performed using keywords contained within fields of each record [5]. So, for faster retrieving the compression methods were used to compress the database files and the dynamic clustering method was used to build clusters contain information about the required query data, so the retrieving data become much faster than the retrieving from original and compressed files. Numerous studies and tools already can be found in the scientific literature.

The system proposed by S. Parthasarathy, V. Shakila [4] focuses mainly on a k_means algorithm that characterizes the features of the Big Data revolution, and proposed a Big Data processing model from the Data mining perspective. This provided the most relevant and most accurate social sensing feedback to better understand our society at real time with big data technologies. The system proposed by Hazem M. El-Bakry, Nikos E. Mastorakis, Michael E. Fafalios [6] focuses on using an efficient model for fast retrieving of specific information from big data. Fast neural networks are used to find the best matching between words in query and stored big data. The idea is to accelerate the searching operation in a big data. This is done by applying cross correlation between the given query and the big data in the frequency domain rather than the time domain. The research work proposed by QIAN WeiNing, GONG XueQing and ZHOU AoYing [7] describes using a hybrid-clustering algorithm to solve the problems of

scanning the whole database, pre-specifying the uncertain parameter k and lacking high efficiency in treating arbitrary shape under very large data set environment. The proposed algorithm combines both distance and density strategies, handled any arbitrary shape clusters effectively. It makes full use of statistical information in mining to reduce the time complexity greatly while keeping good clustering quality.

This research is organized as follows. Section one shows the introduction, section two presents data compression, Section three explains major clustering techniques, section four explains major data retrieval methods, Section five shows the methodology of dynamic clustering method and data retrieval methods and system structure, Section six presents experiments and results, section six offers the conclusion and in section seven suggests a future works.

II. COMPRESSION METHODS

Data compression purposes are to reduce the number of bits used to store or transmit data. Data Compression is basically defined as a technique to reduce the size of data by applying different methods that can either be Lossy or Lossless. Data compression is popular for two reasons: (1) People like to collect data and hate to throw anything away. (2) People hate to wait a long time for data transfers. Compression process may be useful if one wants to save the storage space. For example, if one wants to store a 4MB file, it may be best to compress it to a smaller size to provide the storage space. Also compressed files are much more easily exchanged over the internet since they upload and download much faster [2]. The main goal of data compression is to reduce the redundancy in warehouse or communicated data. Lossless Compression is used when the original data from a source are so important retrieved without lose any details. The main purpose of this compression technique is to compress the file by decreasing the information in such a way that there is a no loss when decompress any file back into the original file. Example of lossless data compression technique is text compression [8]. A lossy data compression method is one where the retrieved data after decompression may not be exactly same as the original data, but is "close enough" to be useful in particular purpose [2]. In this paper the clustering technique (Improved K-means, K-means With Medium Probability and K-means With Maximum Gain Ratio) algorithms were used as lossless compression algorithm and the results have been used to build the dynamic clusters.

III. DATA CLUSTERING

Data mining, the extraction of hidden predictive information from large databases, is a powerful technology with great possibility to help companies concentrate on the most important information in their data warehouses. Data mining tools can answer business queries that traditionally consumed too long time to resolve [9]. Cluster analysis is a mechanism for multivariate analysis that assigns items to create groups based on a calculation of the degree of association between items and groups. There are two main types of cluster analysis methods these are the nonhierarchical, which divide a dataset of N items into M clusters, and the hierarchical, which output nested dataset in which pairs of items or clusters are successively linked. In the

information retrieval (IR) field, cluster analysis has been used to create groups of documents with the goal of benefiting the efficiency and effectiveness of retrieval [10]. Clustering is a data mining technique that makes useful cluster of objects which have similar characteristics using automatic technique [3]. In this paper the (Improved K-means, K-means With Medium Probability and K-means With Maximum Gain Ratio) algorithms were used to build clusters for the inputted database and dynamic clustering method was used to build clusters according to the used entries (queries). The benefit of using (Improved K-means, K-means With Medium Probability and K-means With Maximum Gain Ratio) algorithms as compression methods is to reduce the file size then the required answering time for user queries was decreased. As well, the advantage of building dynamic clusters for entries of user query led to speed up the data retrieval operation (speed up answering for user query).

K-means is one of the most commonly used clustering techniques due to its simplicity and speed. It partitions the data into k clusters by assigning each object to its closest cluster centroid (the mean value of the variables for all objects in that particular cluster) based on the distance measure used. The basic algorithm for k -means works as follows:

Algorithm 1 k-mean [11]

Input: C : the number of cluster and D : A data set containing m objects.

Output: A set of C cluster.

Begin:

```
1: Choose  $m$  objects randomly from dataset as the initial
   cluster centers;
2:   Until there are no changes in the mean values
3:     Use the estimated means to classify  $m$  objects into
        $k$  clusters based on similarity measured
4:   For  $i=1$  to  $k$ 
5:     Calculate mean value of the objects for each
       cluster  $i$  and make replacing old mean with
       new mean
6:   End_for
7:   End_until
8: End
```

Usually, the K-means algorithm criterion function depends on the square error criterion, which can be defined in the following equation:

$$E = \sum_{j=1}^k \sum_{x_i \in c_j} \|x_i - m_j\|^2 \quad (1)$$

In which, E is the total square error of all the objects in the data cluster, x_i is the vector of the i -th element of the dataset, m_i is the mean value of cluster C_i (x and m are both multi-dimensional). K-means is the most important clustering technique that has been used widely in the field of IR. It was grouped data objects into k clusters [12]

IV. DATA RETRIEVAL

Databases are electronic collections of information, the databases were used to retrieve items in a catalog or a periodical database. Each item in a database is a record. Each

record consists of a set of fields [13]. A database-management system (DBMS) is a collection of correlated data and a set of programs to access those data. Usually a collection of data referred to it as the database, contains relevant information to an enterprise. The essential goal of a DBMS is to provide a way to store and retrieve database information that is both convenient and efficient [14]. Information Retrieval is the technique of presentation, storage, organization of and access to information items. The representation and organization of information should be in such ways that the users can access information to meet their information need [15]. Data retrieval means obtaining data from a database management system. The retrieved data may be stored in a file, printed, or viewed on the screen. The retrieval process has been begun with the user entering a query. The query entered by the user can be a one word or it can be a sentence [16]. Information retrieval (IR) is finding items (usually documents) of an unstructured nature (usually text) that meets an information need from within large collections (usually stored on computers) [15]. The difference between information retrieval and data retrieval is summarized in the following table:

TABLE I. THE DIFFERENCE BETWEEN IR AND DATA RETRIEVAL [15]

	Data Retrieval	Information Retrieval
Example	Database Query	WWW Search
Matching	Exact	Partial Match, Best Match
Inference	Deduction	Induction
Model	Deterministic	Probabilistic
Query Language	Artificial	Natural
Query Specification	Complete	Incomplete
Items Wanted	Matching	Relevant
Error Response	Sensitive	Insensitive

Information retrieval (IR) systems uses a simpler data model than database systems, it was information organized as a collection of documents (documents are unstructured data). While Database systems deal with structured data, with schemas that define the data organization [14]. Searching strategies include: keyword and subject searching, Boolean operators, truncation, phrase searching, search limiting, and nesting [4]. Boolean searching is a method based on logic. Most online databases and internet search engines based on Boolean searches. The Boolean operators AND, OR, NOT (or AND NOT). Using AND narrows your search. It retrieves records that contain both of the search items or keywords that you specify. The more items (terms or keywords) connected with (AND) the fewer search results will be found. Using OR expand your search. It retrieves records that contain either of the search items (terms) or keywords that you specify, but not necessarily both. The more items (terms) connected with OR, the more search results will be found. Using NOT narrows the search. It retrieves records that do not contain a search item (term) in your search. NOT was used to exclude a term from your search and to find fewer results [17]. The bellows figure shows these operations.

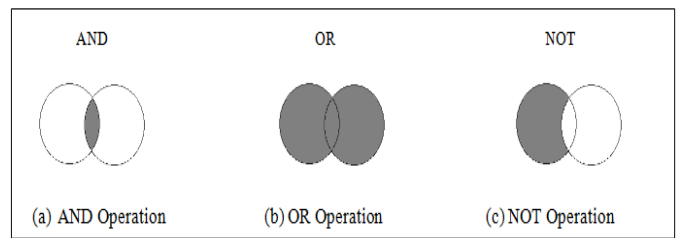


Fig. 1. Boolean Operation

The index is a logical view where documents or data in a collection are represented through a set of index terms or keywords, i.e., any word that appears in the document text or field in the database [18]. An index for a file in a database system works in much the same way as the index in the textbook [14]. Keyword Searching best used method for searching new terms (items), special words, jargon or slang. Phrase searching is a way to retrieve records containing specific phrases. A phrase search will locate only records containing the specified (inputted) words [4]. Keyword query is easy and flexible because it doesn't require from the database user to know details about the database schema. The goal of information retrieval is to identify documents which best match user needs. While the goal of data retrieval is to identify table records which best match user needs [5].

V. PROPOSAL DYNAMIC CLUSTERING SYSTEM

The proposed system is dependent on a dynamic clustering algorithm. Clustering is a collection of items that have high similarity measures in the same cluster and dissimilar to the items belonging to other cluster. The traditional k-means algorithm problems it was needed to specify the center points and the number of the clusters by the user, it was also needed to calculate similarity measures and this will take a long time and it was applied on numerical data only. The (Improved K-means, K-means With Medium Probability and K-means With Maximum Gain Ratio) algorithms were used to compress the database and were built clusters according to the specified centers. The centers have been specified in dynamic way according to the best selected compression algorithm. The selection was conducted by applying the ICM algorithm. The ICM algorithm is worked by analyzing the database, then depending on the extracted features of the database the ICM would be selected best compression algorithm (Improved K-means, K-means With Medium Probability and K-means With Maximum Gain Ratio). We proposed a new algorithm called a dynamic clustering algorithm. This algorithm builds a cluster depending on user entries. The final result of ICM was represented by the compressed file that was resulted from selecting and applying the best compression algorithm. This file was used to build dynamic clusters based on user query (user entries). The dynamic clusters contain data that are relevant to the user queries only. Example voters' information about voters' lives in Baghdad, year nascent like 1990 and name polling station. The results of dynamic clusters have advantages in data retrieval system, it was much faster in retrieving data than retrieving from original database or

compressed file because of the searching was worked in less data than the data contained in original database or in compressed data. The retrieval of data from the large database forms a problem because it was taken much time. So the compression and building dynamic clusters solved this problem, it was taking lesser time in retrieving the required data. Several retrieval methods were used in retrieving data, these methods are: 1) indexing, 2) keyword searching and 3) applying Boolean operations. These retrieving methods have been applied on original database, compressed file and dynamic clustering file and the results were compared. Dynamic clustering includes of making clusters based on specified user query data (entries of the query). The clusters are made from the compressed database file and not from the original database file. There exists a historical file which consists of all the previous user queries for the built clusters. Therefore the stored clusters and its stored query data were not needed to make it again. When the user enters query the system do searches in the historical file, if it was found the query in the historical file then it is returned the answer for the user query directly, else it was making a new cluster for the entering query and then the answer returned for the user query.

The proposed system consists of several phases. These steps consist of:

- 1) Input the original database file
- 2) Apply the ICM algorithm
- 3) Compress the database file with the best compression method
- 4) Returning compressed file
- 5) Input the user query to the dynamic clustering algorithm
- 6) Analyzer: searching the user query if it was existed in the historical file or not and
- 7) Return the results

These steps can be explained in the following phases:

Phase 1: This phase includes selecting the database file to be inputted to ICM. The inputted database can be contained any type of data (numerical data, text data, .etc.).

Phase 2: This phase compromises the selection the optimal compression method. The ICM algorithm was used for this purpose. The ICM algorithm depends on the extracted features of the database by analyzing the inputted database. Also, this algorithm depends on several conditions used to specify the best compression method (improved k-mean, k-mean with medium probability and k-mean with maximum gain ratio). Each one of these algorithms depended on specific calculations to specify the centers of the clusters.

Phase 3: The ideal compression method was selected in the previous phase. In this phase the best compression method can be applied to the entering database.

Phase 4: The results of the compression algorithm are recorded in this phase. These results have been represented by the compressed file contains database data in clusters form.

Phase 5: At this phase the users entering the query data to the dynamic clustering algorithm.

Phase 6: This phase includes analyzing the user query by searching in historical file if the user query exists or not.

Phase 7: this is the final phase consists of displaying the query answers or making new clusters and then displaying query answers. If the query not existed in historical file, then making new clusters and saving the user query in historical file then return the query answer to the user. The proposed system can be explained in the following architecture.

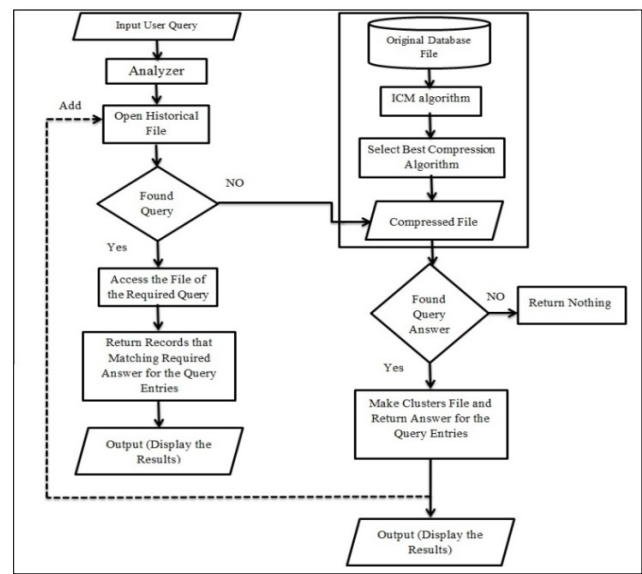


Fig. 2. Proposed System Architecture

The structure of the dynamic clustering system is considered by entering the users for queries data to the analyzer. The analyzer would be done searching in the historical file to specify if the entered queries exist or not. If the queries exist, the system would be returned the required data, else the system would be built new clusters for the required queries and after that the system would be returned the required data. The analyzer has two situations, these are classified as the existed of the user query and not existed of the user query in the historical file. Results can be returned directly or indirectly. The results returned directly if the analyzer in the case of found the query in the historical file. The results returned indirectly if the analyzer in the case of not found the query data in the historical file, this is because of the system was built new clusters and then return the results. The dynamic clustering steps can be explained in the following algorithm:

Algorithm 2: Dynamic clustering algorithm

Input: user query.

Output: answers return the required records that match the user query.

Begin:

- 1: Open the historical file to check if the query exists or not exists in the historical file.
 - 2: If the query exists in the historical file then
 - 3: Fetch the path of the file that contains the data of the entered query and then open this file let it x.
 - 4: While not end of x do
 - 5: Search about the required query using (keyword strategy; indexing strategy or phrase and Boolean operation strategy).
 - 6: Return all the records that match the required query data.
 - 7: End while.
 - 8: Else If the query not exists in the historical file then
 - 9: While not end of the compressed file
 - 10: Search about the required query using (keyword strategy; indexing strategy or phrase and Boolean operation strategy).
 - 11: Open the compressed file and match the query data with data in the compressed file.
 - 12: Open new file for saving clusters that was extracted from matching the user query.
 - 13: Save user query in historical file.
 - 14: End while
 - 15: End If
 - 16: Display the results to the user.
- End.**

The retrieval from the file that was resulted from applying dynamic clustering algorithm much faster than retrieval from original database or compressed database file. In this research the string matching method was used. It was too fast method. The comparison between the words has been applied to find the matching words that match the required entries of user queries. The character comparison is a fast searching method to search the required text or records. The entity search includes of entry from one or more columns. Also, the indexing method was used to search about specific record. In addition to the indexing and string matching search methods, the Boolean operation, keyword searching, SQL query and phrase searching strategies have been used to search about the required queries in clustered and non-clustered data structure. The Boolean (logical) operation was used to search about more than one item, each item located in different columns. The indexing strategy was used to search about distinctive identifier (special ID). Keyword strategy was used to search about records that containing the required keywords (e.g. voters' how have year nascent =1990) and SQL was used to search in the original database file. The structure of dynamic clustering method can be explained in the below figure.

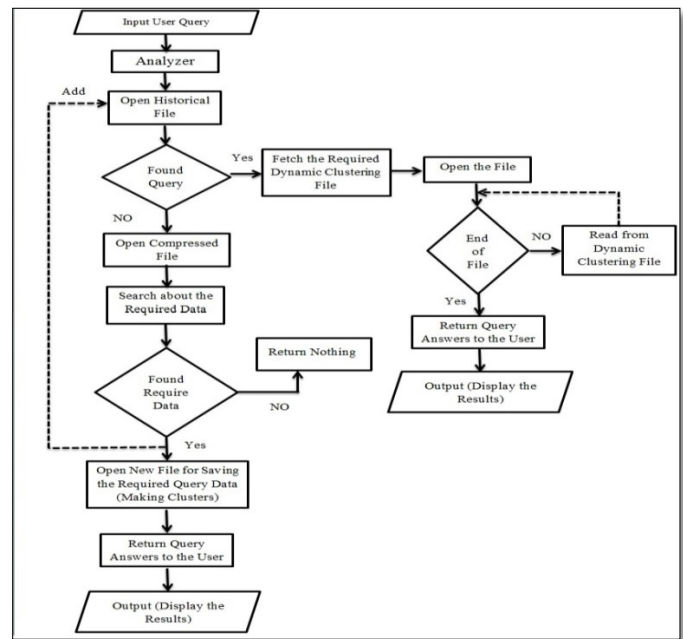


Fig. 3. The structure of dynamic clustering method

VI. RESULTS AND DISCUSSION

This section discusses the results that obtained from the new proposed method (dynamic clustering). The dynamic clustering method was applied on different data sets for different data types, and was conducted different experiments to specify the performance of the proposed method.

The experiments have been conducted on different databases and the results are compared based on the consumed time in retrieving from original database file, compressed database file and dynamic clustering file. The proposed method results showed in the following tables. The time was measured in seconds (e.g. 40.11seconds=40110 milliseconds and 0.016 seconds=16 milliseconds). Tables [2 and 3] show the detailed results for the tested databases such as:

- * DB Name (database name),
- * DB Size (database size),
- * NO of Query (number of query),
- * R.O DB.T (retrieval time from original database file),
- * R.C.T (retrieval time from the compressed file),
- * R.D.T (retrieval time from building dynamic clustering file),
- * P.O DB (preprocessing operation time for original database file),
- * P.C.DB (preprocessing operation time for compressed database),
- * P.D clus (preprocessing operation time for dynamic clustering),
- * T.T.R.O DB (total time for retrieving from original database file),
- * T.T.R.C DB (total time for retrieving from the compressed database file) and

* T.T.R.D clus (total time for retrieving from dynamic clustering resulted file.

From table (2) we would notice that the consumed time to answering about the user query in the state of querying from the dynamic clustering file is too much faster than answering in the situation of querying from the original or compressed database file. This is because of the dynamic clustering file contains the data that had related with only user query. Optimization proposal is advances by reducing the required vocabulary check about the user query.

TABLE II. PROPOSED SYSTEM ANSWERING TIME RESULTS

DB Name	DB Size	No.of Query	R.O DB.T	R.C.T	R.D.T	P.O DB	P.C.DB	P.D clus
dept100	224 KB	9	0.357	0.013	0.007	3.656	2.609	0.004
			0.183	0.007	0.003	3.656	2.609	
			0.174	0.006	0.003	3.656	2.609	
			0.094	0.001	0.001	3.656	2.609	
			0.102	0.001	0.001	3.656	2.609	
			1.15	0.008	0.004	3.656	2.609	
			0.132	0.004	0.002	3.656	2.609	
			0.874	0.312	0.047	3.656	2.609	
			0.005	0.125	0.004	3.656	2.609	
dwc	1,360 KB	9	1.875	2.1	0.424	7.344	8.156	0.036
			0.641	1.866	0.312	7.344	8.156	
			1.234	0.234	0.112	7.344	8.156	
			0.203	0.187	0.031	7.344	8.156	
			0.062	0.062	0.001	7.344	8.156	
			2.437	0.437	0.265	7.344	8.156	
			1.234	0.25	0.187	7.344	8.156	
			1.203	0.187	0.078	7.344	8.156	
			0.062	0.187	0.016	7.344	8.156	
voters	288,192 KB	26	37.423	24.047	0.193	1.891	7.336	3.547
			30.615	17.885	0.087	1.891	7.336	
			30.281	17.638	0.083	1.891	7.336	
			19.434	17.729	0.067	1.891	7.336	
			5.734	17.699	0.051	1.891	7.336	
			5.399	18.112	0.049	1.891	7.336	
			34.972	20.711	0.359	1.891	7.336	
			19.335	18.055	0.031	1.891	7.336	
			19.339	18.164	0.125	1.891	7.336	
			27.09	28.531	0.828	1.891	7.336	
			32.874	18.291	0.176	1.891	7.336	
			5.452	17.821	0.157	1.891	7.336	
			32.039	19.406	0.016	1.891	7.336	
			31.513	18.21	0.177	1.891	7.336	
			79	51.466	1.125	1.891	7.336	
			39.067	32.852	1.032	1.891	7.336	
			18.519	17.136	0.024	1.891	7.336	
			46.921	17.569	0.02	1.891	7.336	
			38.627	17.979	0.172	1.891	7.336	
			38.082	20.776	0.704	1.891	7.336	
			45.53	20.11	0.268	1.891	7.336	
			29.542	21.426	0.474	1.891	7.336	
			10.033	34.443	0.906	1.891	7.336	
			36.953	22.884	0.072	1.891	7.336	
			38.5	23.584	0.066	1.891	7.336	
			35.531	23.282	0.025	1.891	7.336	

TABLE III. PROPOSED SYSTEM TOTAL TIME RESULTS

DB Name	DB Size	No.of Query	T.T.R.O DB	T.T.R.C DB	T.T.R.D clus
dept100	224 KB	9	3.071	0.477	0.072
dwc	1,360 KB	9	8.951	5.51	1.426
voters	288,192 KB	26	787.805	575.806	7.287
Total Time			799.827	581.793	8.785

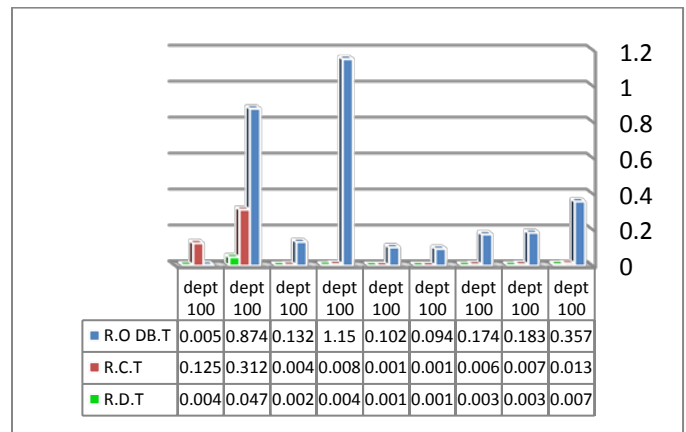


Fig. 4. Retrieval time for database dept100 of size 224 KB

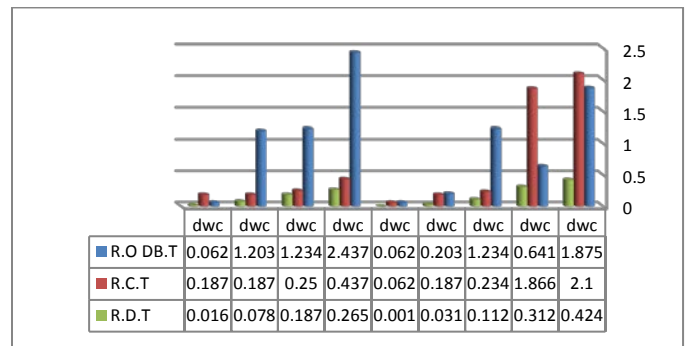


Fig. 5. Retrieval time for database DWC of size 1,360 KB

TABLE IV. RETRIEVAL TIME FOR DATABASE VOTERS OF SIZE 288,192 KB

DB Name	R.O DB.T	R.C.T	R.D.T
voters	37.423	24.047	0.193
voters	30.615	17.885	0.087
voters	30.281	17.638	0.083
voters	19.434	17.729	0.067
voters	5.734	17.699	0.051
voters	5.399	18.112	0.049
voters	34.972	20.711	0.359
voters	19.335	18.055	0.031
voters	19.339	18.164	0.125
voters	27.09	28.531	0.828
voters	32.874	18.291	0.176
voters	5.452	17.821	0.157
voters	32.039	19.406	0.016
voters	31.513	18.21	0.177
voters	79	51.466	1.125
voters	39.067	32.852	1.032
voters	18.519	17.136	0.024
voters	46.921	17.569	0.02
voters	38.627	17.979	0.172
voters	38.082	20.776	0.704
voters	45.53	20.11	0.268
voters	29.542	21.426	0.474
voters	10.033	34.443	0.906
voters	36.953	22.884	0.072
voters	38.5	23.584	0.066
voters	35.531	23.282	0.025

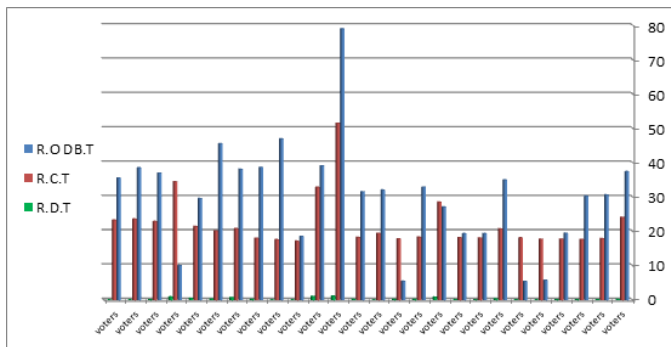


Fig. 6. Retrieval time for database voters of size 288,192 KB

The previous figures showing a comparison in the required retrieval time to retrieving from the original database, compressed database and dynamic clustering files. The following figure explains a total time for retrieving from the original database, compressed database and the file that was resulted from dynamic clustering method.

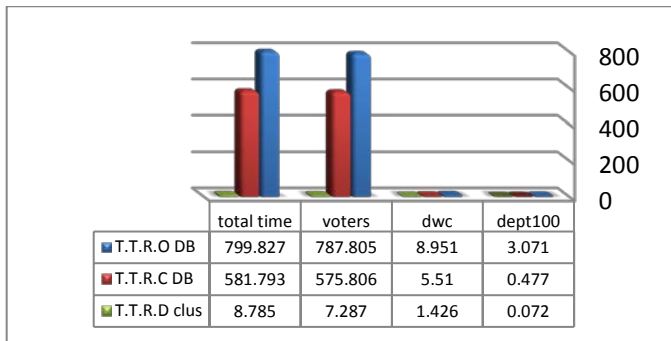


Fig. 7. Total retrieval time

The results showed that: 1) the larger retrieving time from the original database file is 56.763 seconds and the smaller retrieving time is 5 milliseconds, 2) the larger retrieving time from the compressed database file is 34.445 seconds and the smaller retrieving time is 1millisecond and 3) the larger retrieving time from the dynamic clustering file is 1.032 seconds and the smaller retrieving time is 1 millisecond.

VII. CONCLUSION

1) In this research the dynamic clustering method was used to solve the problem of searching. Instead of searching in whole database data, the dynamic clustering enabling the user to search in data that it was relevant to the user query only. The dynamic clustering method built the clusters contained the data about the user query only. So when the user searching in this data, the results (query answers) can be received quickly much faster than searching in the whole database data or compressed database data. By this the dynamic clustering method solved the problem of the slowing in retrieving data.

2) The dynamic clustering algorithm built the clusters according to the user entries. In addition, the retrieval by using the file that is the output of a dynamic clustering process may be needed lesser inputs by the user compared with retrieval

from the original database or compressed database which was needed more inputs.

3) The historical file solved the problem of building many similar copies of the same clusters. Instead of building new clusters similar to one existed, it was building the clusters only once. The user enters the query to the analyzer then the analyzer begins with searching the query in the historical file to specify if the query existed or not. This mechanism provides a saving time for retrieving answers directly if the query was existed in the historical file.

VIII. FUTURE WORKS

Future work includes the proposal of new methods in order to encrypt the data and then retrieving the data from this encrypted data. The encryption of this large data in order to protect them from hackers and preservation from being stolen. As well as new methods based on artificial intelligence will be used for the retrieving purpose from large data.

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A Fast Adaptive Artificial Neural Network Controller for Flexible Link Manipulators

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Abstract—This paper describes a hybrid approach to the problem of controlling flexible link manipulators in the dynamic phase of the trajectory. A flexible beam/arm is an appealing option for civil and military applications, such as space-based robot manipulators. However, flexibility brings with it unwanted oscillations and severe chattering which may even lead to an unstable system. To tackle these challenges, a novel control architecture scheme is presented. First, a neural network controller based on the robot's dynamic equation of motion is elaborated. Its aim is to produce a fast and stable control of the joint position and velocity and damp the vibration of each arm. Then, an adaptive Cerebellar Model Articulation Controller (CMAC) is implemented to balance unmodeled dynamics, enhancing the precision of the control. Efficiency of the new controller obtained is tested on a two-link flexible manipulator. Simulation results on a dynamic trajectory with a sinusoidal form show the effectiveness of the proposed control strategy.

Keywords—Adaptive control; CMAC neural network; artificial neural network; nonlinear control; flexible-link manipulator; dynamic motion equation

I. INTRODUCTION

The trajectory control of a manipulator robot can be decomposed into two parts. Tracking the desired trajectory on the dynamic phase of the movement and positioning the tip of the link on the final phase of the movement. While the majority of the existing researches on the control of flexible link manipulators concentrate on the positioning phase of the movement, very few of them deal with the dynamic phase of the movement.

This paper presents a new control system structure to deal with the tracking control problem of flexible link manipulators on the dynamic phase of the trajectory.

The use of knowledge-based modeling, whereby mathematical equations are derived in order to describe a process, based on a physical analysis, is important to elaborate effective controllers. However, this may lead to a complex controller design if the model of the system to be controlled is more complex and time consuming. Therefore, the controller presented in this paper is based on Artificial Neural Networks (ANNs) that approximate the dynamic model of the robot.

Using ANNs, replacing nonlinear modeling, may simplify the structure of the controller, reduce its computation time and enhance its reactivity without a loss in the accuracy of the tracking control. This is important when real time control is needed.

The main advantage of neural networks control techniques among others is that they use nonlinear regression algorithms that can model high dimensional systems with extreme flexibility due to their learning ability.

To reduce the modeling error between the actual system and its representation, an adaptive Cerebellar Model Articulation Controller (CMAC) is added. The advantages of using a CMAC in the adaptive controller are as follows. The CMAC is fast in terms of convergence speed and computation time. Because of the associative and local generalization properties of the CMAC, the number of training cycles to converge is orders of magnitude smaller with the CMAC than with other neural networks [1]. The learning law and the output function of the CMAC are simple, so the CMAC needs fewer computations and less time to make adjustments and produce outputs than other neural networks, in which complex update laws and nonlinear sigmoid output functions are involved.

CMAC networks were used in many military applications. In [2], they were applied to the control of a parallel hybrid-electric propulsion system for a small unmanned aerial vehicle (UAV). The CMAC controller saves on the required memory compared to a lookup table by two orders of magnitude. The hybrid-electric UAV with the CMAC controller uses 37.8% less energy than a two-stroke gasoline-powered UAV during a three-hour intelligence, surveillance, or reconnaissance mission.

Demand for increased productivity in industry has led to the use of lighter robots with faster response and lower energy consumption. Flexible-link manipulators have relatively smaller actuators, higher payload to weight ratio and, generally, less overall cost. The drawbacks are a reduction in the stiffness of the manipulator structure which results in an increase in robot deflection and poor performances due to the effect of mechanical vibration in the links.

Trajectory following control of flexible-link manipulator system has been an important research area in the last three decades. A non-rigid link bears resemblance to a flexible (cantilever) beam often used as a starting point in modeling the dynamics of a flexible link [3].

Well-known approaches such as Euler-Lagrange's equation and Hamilton's principle commonly used in modeling the motion of rigid-link manipulator have been applied to derive the general equation of motion for flexible link manipulator.

Infinite-dimensional manipulator system is commonly approximated by a finite-dimensional model for controller design. Finite element method is used to derive the dynamic model leading to a computationally attractive form for the displacement bending [4]. However, most of the control techniques for non-rigid manipulators are inspired by classical controls. A bibliographical study presents some of them.

A multi-step control strategy was used in [5-11], which consists of superimposing to the control of the rigid body, the techniques of shaping or correction of the elastic effects. Other algorithms use the techniques of decoupling [12, 13], others are based on the method of the singular perturbation approach [14-16] or use non-collocated feedback [17]. Sliding mode control algorithms were successfully applied for maneuvering planar flexible manipulators while suppressing vibrations [18, 19].

On other hand, much research effort has been put into the design of artificial neural network and fuzzy logic-based controllers as they reduce the complexity and allow a faster computation of the command [20-32].

With recent developments in sensor/actuator technologies, many researchers have concentrated on control methods for suppressing vibrations of flexible structures using smart materials such as shape memory alloys (SMA) [33], magnetorheological (MR) materials [34], electrorheological (ER) materials [35], piezoelectric transducers (PZT) [36-39], or using positive position feedback (PPF) to suppress the vibration amplitude of nonlinear systems [40], etc.

New techniques, based on swarm intelligence have also been used to elaborate optimal controllers for vibration suppression [41].

The majority of the existing non-linear controls nevertheless are subjected to constraints such as: frequency and damping of the mode shapes known exactly a priori or complex online computing like matrix inversion or computation of the dynamics of the manipulator. Other control techniques suffer from the lack of robustness facing significant variation of the dynamic parameters of the manipulator, particularly the payload.

The presented control law has several distinguished advantages. It is easy to compute since it does not require the computation of all or part of the dynamic model. This robust controller design method maximizes the control performance guaranteeing good precision when regulating the tip position of the flexible arm in the presence of large structured and unstructured uncertainties.

The reminder of this article is organized as follows. In Section 2, a two-link planar flexible manipulator is modeled according to Euler-Lagrange's formulation and finite element method for the discretization. Section 3 presents the non-linear control. Stability analysis of this control method is carried out in Section 4. Section 5 presents the approach used to reduce the complexity of the controller. In Section 6 an adaptive controller is added to enhance the precision of the control. Section 7 describes simulation results. Tests have been carried out for the hybrid controller and compared to those obtained with the non-linear controller used solely. Results show the efficiency of the hybrid control strategy facing an important variation of the

dynamic parameters. Finally, conclusion is presented in Section 8.

II. DYNAMIQUE MODELLING

The system considered here consists of two links connected with a rotating joint moving in a horizontal plane as shown in Fig. 1. The first and the second link are composed of a flexible beam cantilevered onto a rigid rotating joint. It is assumed that the links can be bent freely in the horizontal plane but are stiff in the vertical bending and torsion. Thus, the Euler-Bernoulli beam theory is sufficient to describe the flexural motion of the links. Lagrange's equation and model expansion method can be utilized to develop the dynamic modeling of the robot.

As shown in Fig. 1, $\{O_0 \bar{x}_0 \bar{y}_0\}$ represents the stationary frame, $\{O_1 \bar{x}_1 \bar{y}_1\}$ and $\{O_2 \bar{x}_2 \bar{y}_2\}$ are the moving coordinate frames with origin at the hubs of links 1 and 2, respectively. \bar{y}_1 and \bar{y}_2 are omitted to simplify the figure. θ_1 and θ_2 are the revolving angles at the hub of the two links with respect to their frames. f_1, α_1, f_2 and α_2 are the elastic displacements, they describe deflection and section rotation of the tip for the first and the second arm, respectively.

Motion of each manipulator's arm is described by one rigid and one elastic variable [1]:

$$\mathbf{q} = [\mathbf{q}_r \ \mathbf{q}_e]^T \quad (1)$$

where $\mathbf{q}_r = [\theta_1 \ \theta_2]^T$ and $\mathbf{q}_e = [f_1 \ f_2]^T$.

Torques applied to the manipulator joints are given by:

$$\mathbf{\Gamma} = [\Gamma_1 \ \Gamma_2]^T \quad (2)$$

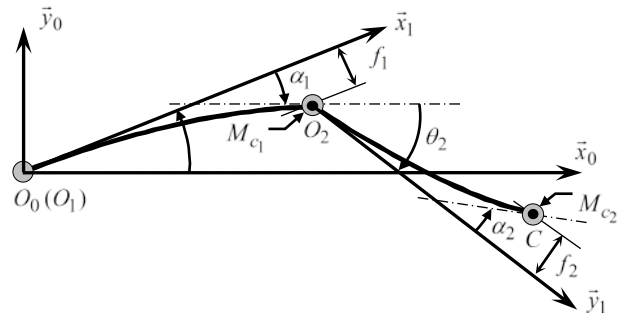


Fig. 1. Two-link manipulator with flexible arms

Let us consider an arbitrary point M_1 on the link 1. The expression of M_1 in the moving coordinate frame $\{O_1 \bar{x}_1 \bar{y}_1\}$ is given by:

$${}_{O_1}M_1 = (x - y f_{,x}) \bar{x}_1 + (y + f) \bar{y}_1 \quad (3)$$

With, (x, y) the coordinates of M_1 in a non deformed link, f the deflection at the abscissa x and $f_{,x} = \frac{\partial f}{\partial x}$.

Velocity of the point M_1 is given by the equation:

$$\begin{aligned} \mathbf{V}(M_1) &= \frac{d}{dt} \mathbf{O}_1 \mathbf{M}_1 \\ &= -\left[(y+f)\dot{\theta}_1 + y\dot{f}_{,x}\right]\bar{x}_1 + \left[(x-yf_{,x})\dot{\theta}_1 + \dot{f}\right]\bar{y}_1 \end{aligned} \quad (4)$$

Finite-element theory allows to write the approximation [4]:

$$f(x,t) = \left(\frac{3x^2}{L_1^2} - \frac{2x^3}{L_1^3}\right)f_1(t) + L_1\left(-\frac{x^2}{L_1^2} + \frac{x^3}{L_1^3}\right)\alpha_1(t) \quad (5)$$

With, L_1 the length of the first link, f_1 and α_1 are the deflection and the section rotation of the tip of the first link, respectively.

Bernoulli–Euler beam theory states that for a link with small elasticity, deflection and section rotation can be considered linearly dependent. Therefore, the section rotation can be written [42]:

$$\alpha_1 \approx \tan(\alpha_1) = \frac{3f_1}{2L_1} \quad (6)$$

With, $\tan(\cdot)$ is the tangent function. The value of α_1 is considered here in radians.

The same reasoning can be made for an arbitrary point M_2 on the link 2.

The kinetic energy T_i of the link i (with $i=1,2$) is then given by:

$$T_i = \frac{1}{2} \rho_i \int_0^{L_i} \int_0^{S_i} [\mathbf{V}(M_i)]^2 ds dl \quad (7)$$

where $\mathbf{V}(M_i)$ is the velocity of M_i on the flexible link i . L_i , S_i and ρ_i are the length, the section and the mass density of link i ($i=1,2$), respectively.

Now, the total kinetic energy T can be written as [43]:

$$\begin{aligned} T &= T_1 + T_2 + \frac{1}{2} J_{A_1} \dot{\theta}_1^2 + \frac{1}{2} J_{B_1} (\dot{\theta}_1 + \dot{\alpha}_1)^2 \\ &+ \frac{1}{2} J_{A_2} (\dot{\theta}_1 + \dot{\alpha}_1 + \dot{\theta}_2)^2 + \frac{1}{2} J_{B_2} (\dot{\theta}_1 + \dot{\alpha}_1 + \dot{\theta}_2 + \dot{\alpha}_2)^2 \\ &+ \frac{1}{2} M_{C_1} \mathbf{V}(O_2)^2 + \frac{1}{2} M_{C_2} \mathbf{V}(C)^2 \end{aligned} \quad (8)$$

Where J_{A_i} and J_{B_i} are, the mass moment of inertia at the origin and at the end of link i , respectively. Note that the first and the second terms on the right-hand side in (8) are kinetic energy of the flexible links 1 and 2, respectively. The third term is due to moment of inertia of the portion of the mass of the first actuator relative to link 1. The fourth and the fifth terms are due to moment of inertia of the portion of the mass of the second actuator in relation to link 1 and portion of the mass of the second actuator in relation to link 2, respectively. The

sixth term is due to moment of inertia of mass at C (payload). The seventh and the eighth terms are kinetic energy of mass at O_2 and C respectively.

The potential energy U can be written as [27]:

$$U = \sum_{i=1}^2 \frac{1}{2} E_i I_i \int_0^{L_i} \left[\frac{\partial^2 f_i}{\partial x_i^2} \right]^2 dx_i \quad (9)$$

the term on the right-hand side in (9) describes the potential energy due to elastic deformation of the links. The term relative to the gravity is not present here as the manipulator moves on a horizontal plane. f_i is the deflection at the abscissa x_i of the link i . E_i is the Young's modulus and I_i the second area moment of inertia of the considered link.

Then the potential energy can be reduced to the more condensed form:

$$U = \frac{1}{2} \mathbf{q}^T \mathbf{K} \mathbf{q} \quad (10)$$

Where, \mathbf{K} is the symmetric stiffness matrix. The first two rows and columns of \mathbf{K} are zeros as U does not depend on \mathbf{q}_r ,

so it can be written $\begin{bmatrix} \mathbf{0} & \mathbf{0} \\ \mathbf{0} & \mathbf{K}_e \end{bmatrix}$.

The dynamic motion equation can be derived in terms of the Lagrange-Euler formulation:

$$\frac{d}{dt} \left[\frac{\partial \ell}{\partial \dot{q}_r(i)} \right] - \left[\frac{\partial \ell}{\partial q_r(i)} \right] = \Gamma_i \quad (i=1,2) \quad (11a)$$

$$\frac{d}{dt} \left[\frac{\partial \ell}{\partial \dot{q}_e(j)} \right] - \left[\frac{\partial \ell}{\partial q_e(j)} \right] = 0 \quad (j=1,2) \quad (11b)$$

Where ℓ is the Lagrangian function and $\ell = T - U$.

Substituting (8) and (10) into (11a) and (11b) yields to [20]:

$$\mathbf{L}_r \Gamma = \mathbf{A}(\mathbf{q}) \ddot{\mathbf{q}} + \mathbf{B}(\mathbf{q}, \dot{\mathbf{q}}) \dot{\mathbf{q}} + \mathbf{K} \mathbf{q} \quad (12)$$

where $\mathbf{L}_r \Gamma$ is the torque vector $[\Gamma_1, \Gamma_2, 0, 0]^T$ applied to the joint, $\mathbf{A}(\mathbf{q})$ is the (4×4) inertia matrix, $\mathbf{B}(\mathbf{q}, \dot{\mathbf{q}})$ is the (4×4) matrix of Coriolis and centrifugal terms, \mathbf{K} is the (4×4) stiffness matrix.

III. NONLINEAR CONTROL

This control law consists of a proportional and derivative (PD) part completed by a reduced model which contains only the rigid part of the whole nonlinear dynamic model of the flexible manipulator. It is a generalization of the classically known 'computed torque' used to control rigid manipulator [44, 45]. Dynamic equation of motion (12) can be rewritten as:

$$\begin{bmatrix} \Gamma \\ \mathbf{0} \end{bmatrix} = \begin{bmatrix} \mathbf{A}_r & \mathbf{A}_{re} \\ \mathbf{A}_{er} & \mathbf{A}_e \end{bmatrix} \begin{bmatrix} \ddot{\mathbf{q}}_r \\ \ddot{\mathbf{q}}_e \end{bmatrix} + \begin{bmatrix} \mathbf{B}_r & \mathbf{B}_{re} \\ \mathbf{B}_{er} & \mathbf{B}_e \end{bmatrix} \begin{bmatrix} \dot{\mathbf{q}}_r \\ \dot{\mathbf{q}}_e \end{bmatrix}$$

$$+ \begin{bmatrix} 0 & 0 \\ 0 & \mathbf{K}_e \end{bmatrix} \begin{bmatrix} \mathbf{q}_r \\ \mathbf{q}_e \end{bmatrix} \quad (13)$$

or,

$$\Gamma = \mathbf{A}_r \ddot{\mathbf{q}}_r + \mathbf{B}_r \dot{\mathbf{q}}_r + \mathbf{A}_{re} \ddot{\mathbf{q}}_e + \mathbf{B}_{re} \dot{\mathbf{q}}_e \quad (13a)$$

$$\mathbf{0} = \mathbf{A}_{er} \ddot{\mathbf{q}}_r + \mathbf{B}_{er} \dot{\mathbf{q}}_r + \mathbf{A}_e \ddot{\mathbf{q}}_e + \mathbf{B}_e \dot{\mathbf{q}}_e + \mathbf{K}_e \mathbf{q}_e \quad (13b)$$

Therefore, the following control law is proposed:

$$\Gamma_{NL} = \mathbf{A}_r \ddot{\mathbf{q}}_r^d + \mathbf{B}_r \dot{\mathbf{q}}_r^d + \mathbf{K}_{pr} \tilde{\mathbf{q}}_r + \mathbf{K}_{vr} \dot{\tilde{\mathbf{q}}}_r \quad (14)$$

Where \mathbf{q}_r^d , $\dot{\mathbf{q}}_r^d$ and $\ddot{\mathbf{q}}_r^d$ define the desired angular trajectory. $\tilde{\mathbf{q}}_r = \mathbf{q}_r^d - \mathbf{q}_r$, $\dot{\tilde{\mathbf{q}}}_r = \dot{\mathbf{q}}_r^d - \dot{\mathbf{q}}_r$ are angular position and velocity errors, respectively. \mathbf{K}_{pr} and \mathbf{K}_{vr} are positive gain constants.

IV. STABILOITY ANALYSIS

In this section a Lyapunov stability analysis of the nonlinear control given in (14) is presented.

By subtracting the control law (14) from the dynamic motion equation (13a), the error equation is written:

$$\mathbf{0} = \mathbf{A}_r \ddot{\tilde{\mathbf{q}}}_r + \mathbf{A}_{re} \ddot{\tilde{\mathbf{q}}}_e + \mathbf{B}_r \dot{\tilde{\mathbf{q}}}_r + \mathbf{B}_{re} \dot{\tilde{\mathbf{q}}}_e + \mathbf{K}_{pr} \tilde{\mathbf{q}}_r + \mathbf{K}_{vr} \dot{\tilde{\mathbf{q}}}_r \quad (15)$$

With $\tilde{\mathbf{q}}_e = \mathbf{0} - \mathbf{q}_e = -\mathbf{q}_e$ and $\dot{\tilde{\mathbf{q}}}_e = \mathbf{0} - \dot{\mathbf{q}}_e = -\dot{\mathbf{q}}_e$ representing the elastic stabilization errors.

In addition, rewriting the coupling equation (13b) according to the trajectory and the elastic stabilization error variables gives:

$$\mathbf{A}_{er} \ddot{\mathbf{q}}_r^d + \mathbf{B}_{er} \dot{\mathbf{q}}_r^d = \mathbf{A}_{er} \ddot{\tilde{\mathbf{q}}}_r + \mathbf{A}_e \ddot{\tilde{\mathbf{q}}}_e + \mathbf{B}_{er} \dot{\tilde{\mathbf{q}}}_r + \mathbf{B}_e \dot{\tilde{\mathbf{q}}}_e + \mathbf{K}_e \tilde{\mathbf{q}}_e \quad (16)$$

Equations (15) and (16) allow to write the global error equation:

$$\mathbf{A} \ddot{\tilde{\mathbf{q}}} + \mathbf{B} \dot{\tilde{\mathbf{q}}} + \mathbf{K}_p \tilde{\mathbf{q}} + \mathbf{K}_v \dot{\tilde{\mathbf{q}}} + \mathbf{s}_1 = \mathbf{0} \quad (17)$$

Where the positive constant matrices \mathbf{K}_p and \mathbf{K}_v are

$$\begin{bmatrix} \mathbf{K}_{pr} & \mathbf{0} \\ \mathbf{0} & \mathbf{K}_e \end{bmatrix} \quad \text{and} \quad \begin{bmatrix} \mathbf{K}_{vr} & \mathbf{0} \\ \mathbf{0} & \mathbf{0} \end{bmatrix}, \quad \text{respectively, and}$$

$$\mathbf{s}_1 = - \begin{bmatrix} \mathbf{0} \\ \mathbf{A}_{er} \ddot{\mathbf{q}}_r^d + \mathbf{B}_{er} \dot{\mathbf{q}}_r^d \end{bmatrix}.$$

The following Lyapunov function is considered to study the stability of the global system,

$$\mathbf{V} = \frac{1}{2} \dot{\tilde{\mathbf{q}}}^T \mathbf{A} \dot{\tilde{\mathbf{q}}} + \frac{1}{2} \tilde{\mathbf{q}}^T \mathbf{K}_p \tilde{\mathbf{q}} \quad (18)$$

Differentiating \mathbf{V} , using (17) and the fact that \mathbf{A} is symmetric positive-definite [46], gives:

$$\dot{\mathbf{V}} = \dot{\tilde{\mathbf{q}}}^T \left(\frac{1}{2} \dot{\mathbf{A}} - \mathbf{B} \right) \dot{\tilde{\mathbf{q}}} - \dot{\tilde{\mathbf{q}}}^T \left(\mathbf{K}_v \dot{\tilde{\mathbf{q}}} + \mathbf{s}_1 \right) \quad (19)$$

The property of passivity of the flexible manipulator provides that $(1/2)\dot{\mathbf{A}} - \mathbf{B}$ is skew symmetric [47], (19) becomes:

$$\dot{\mathbf{V}} = -\dot{\tilde{\mathbf{q}}}_r^T \mathbf{K}_{vr} \dot{\tilde{\mathbf{q}}}_r + \dot{\tilde{\mathbf{q}}}_e^T (\mathbf{A}_{er} \ddot{\mathbf{q}}_r^d + \mathbf{B}_{er} \dot{\mathbf{q}}_r^d) \quad (20)$$

The Lyapunov second method provides that the asymptotic stability of the control is borne out if the following conditions are met. \mathbf{V} is strictly positive everywhere except in $\tilde{\mathbf{q}} = \mathbf{0}$ where it is $\mathbf{0}$ and $\dot{\mathbf{V}}$ is strictly negative everywhere except in $\dot{\tilde{\mathbf{q}}} = \mathbf{0}$ where it is $\mathbf{0}$.

These conditions are always met if the desired angular velocities and accelerations are not too significant for a given tuning of \mathbf{K}_{vr} , so that $\dot{\mathbf{V}}$ remains essentially negative to ensure the control stability.

V. REDUCING THE COMPUTATIONAL BURDREN USING ARTIFICIAL NEURAL NETWORKS

The nonlinear law presented in (14) has some major advantages as it uses information extracted from the dynamic motion equations of the system to control the manipulator. Physical characteristics like the passivity of the system can then be used to elaborate a stable controller [27].

The drawback is that, using the model of the system in the construction of the controller can lead to a complex controller [1]. Computing such a controller can be time consuming. This is mainly the case with flexible manipulators as they are governed by complex equations which lead generally to a huge model. Using such a model can be incompatible with real time control. To avoid this problem, the proposed here is to approximate parts of the model (which will be used in the controller) with ANNs. The main feature that makes ANNs ideal technology for controller systems is that they are nonlinear regression algorithms that can model high-dimensional systems and have the extreme flexibility due to their learning ability. In addition their computation is very fast.

Hence, the functions \mathbf{A}_r and \mathbf{B}_r are approximated with the ANNs $\mathbf{A}_r\text{NN}$ and $\mathbf{B}_r\text{NN}$, respectively. Then, their outputs will be used in addition to the PD part of (14) to elaborate the first controller:

$$\Gamma_{NN} = \mathbf{A}_r\text{NN}(\mathbf{q}_r, \mathbf{q}_e) \ddot{\mathbf{q}}_r^d + \mathbf{B}_r\text{NN}(\mathbf{q}_r, \mathbf{q}_e, \dot{\mathbf{q}}_r, \dot{\mathbf{q}}_e) \dot{\mathbf{q}}_r^d + \mathbf{K}_{pr} \tilde{\mathbf{q}}_r + \mathbf{K}_{vr} \dot{\tilde{\mathbf{q}}}_r \quad (21)$$

Multi Layer Perceptron (MLP) model is adopted in the neural network design scheme of $\mathbf{A}_r\text{NN}$ and $\mathbf{B}_r\text{NN}$, with three-layered networks consisting of input, hidden and output layers. A sigmoid function is used in the hidden layer and a linear function in the output layer. There are 8 neurons in the hidden layer of $\mathbf{A}_r\text{NN}$ and 12 neurons in the hidden layer of $\mathbf{B}_r\text{NN}$.

Back-propagation algorithm is adopted to perform supervised learning. The two distinct phases to the operation of

back-propagation learning include the forward phase and the backward phase.

In the forward phase the input signal propagate through the network layer by layer, producing a response at the output of the network.

In this control scheme, the input signals of the input layer for A_r NN are the rigid and elastic position of the two links:

$[\theta_1, \theta_2, f_1, f_2]^T$. For B_r NN the inputs are rigid and elastic position and velocity of the two links: $[\theta_1, \theta_2, f_1, f_2, \dot{\theta}_1, \dot{\theta}_2, \dot{f}_1, \dot{f}_2]^T$.

The actual response of A_r NN and B_r NN so produced are then compared with the desired response of A_r and B_r respectively, generating error signals that are then propagated in a backward direction through the network.

In the backward phase, the delta rule learning makes the output error between the output value and the desired output value change the weights and reduce the error. The training is made off line so that it does not disturb the real time control.

VI. ADAPTIVE CMAC NEURAL CONTROL

To reduce the modeling error between the actual system and its representation, an adaptive Cerebellar Model Articulation Controller (CMAC) is added.

The overall robotic manipulator control system obtained is shown in Fig. 2. It can be written:

$$\Gamma = \Gamma_{NN} + \Gamma_{CMAC} \quad (22)$$

where Γ is the overall controller's output (torque), Γ_{NN} is the neural network controller's output as defined in (21) and Γ_{CMAC} is the CMAC adaptive neural controller's output.

The CMAC offers the potential of parallel computation with high flexibility; it can improve the controller's response time, important for robotic dynamic tracking. The fast convergence of its algorithm is essential for online adaptation. Therefore, the CMAC is used in the adaptive control part of the proposed control strategy.

CMAC is an associative memory neural network in that each of inputs maps to a subset of weights whose values are summed to produce outputs.

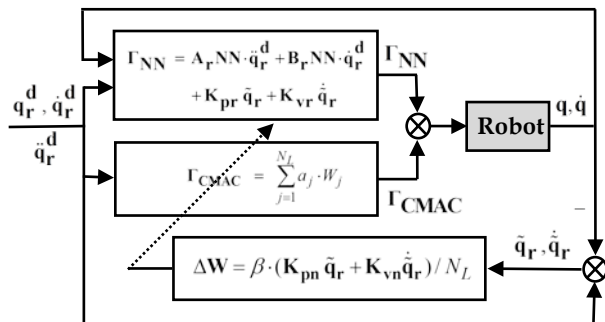


Fig. 2. The overall control system

An important concept used here is generalization that assumes that similar states require similar control efforts. The use of generalization speeds up learning because a group of memory cells that are close is updated in each control cycle. Generally, all the memory cells in a hypercubic region are updated in each control cycle. On the other hand, input vectors that are far away from each other will generate independent outputs [1].

The Cerebellar Model Articulation Controller (CMAC) network was first developed by Albus [48] to approximate the information processing characteristics of the human cerebellum. Miller [49] later developed a practical implementation of the CMAC neural network that could be applied to real-time control applications.

On a typical Albus CMAC neurons are called receptive fields and are organized as follows. The total collection of receptive fields is divided into N_L layers. The layers represent parallel N -dimensional hyperspaces for a network with N inputs. The receptive fields in each of the layers have rectangular boundaries and are organized so as to span the input space without overlap. Any input vector excites one receptive field from each layer, for a total of N_L excited receptive fields for any input. Each of the layers of receptive fields is identical in organization, but each layer is offset by a quantity Q_L relative to the others in the input hyperspace.

The width of the receptive fields produces input generalization, while the offset of the adjacent layers input quantization. The ratio of the width of each receptive field to the offset between adjacent layers must be equal to N_L for all dimensions of the input space. The integer parameter N_L is referred to as the generalization parameter.

A two-dimensional CMAC (2D CMAC), is presented as an example. The structure of a 2D CMAC is shown in Fig. 3. The two-dimensional input vector is defined by two input variables, x_1 and x_2 , quantized into three discrete regions, called receptive fields.

It is noted that the width of receptive fields affects the generalization capability of the CMAC network. For the first layer, the variable x_1 is divided into receptive fields A, B, and C and the variable x_2 is divided into receptive fields a, b, and c. The areas Aa, Ab, Ac, Ba, Bb, Bc, Ca, Cb, and Cc formed by quantized regions are called hypercubes.

By shifting each block a small interval Q_L , different hypercubes can be obtained. In Fig. 3, there are 27 hypercubes used to distinguish 49 different states in the 2D CMAC. For example, let the hypercubes Bb, Fe, and Hh be addressed by the state $(x_1, x_2) = (4, 3)$, only these three hypercubes are one, and the others are zero. The trained data are stored into these regions.

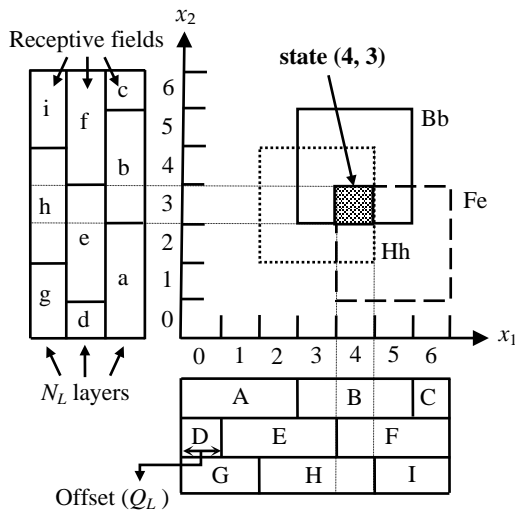


Fig. 3. Structure of a two-dimensional CMAC

The CMAC network is a local network. For a given input vector, only a few of the networks nodes (or hypercube cells) will be active and will effectively contribute to the corresponding network output. The general architecture of the CMAC network is shown in Fig. 4.

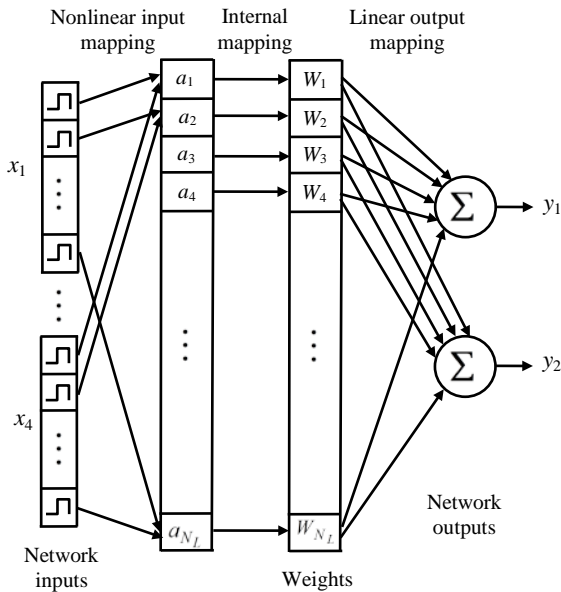


Fig. 4. Architecture of the CMAC network

The basic idea of the CMAC network is to store learned data into overlapping regions in a way that the data can easily be recalled but use less storage space. Furthermore, the action of storing weight information in the CMAC network is similar to that of the cerebellum in humans.

In the CMAC network used, each nonlinear output function y_i of the CMAC network corresponds to one CMAC controller's output (torque) Γ_{CMAC_i} (with $i=1,2$).

The vector of nonlinear output functions $y = f(x)$ is approximated using two primary mappings, $S: Q \rightarrow A$ and $P: A \rightarrow D$. Here, Q is a 4-dimensional input space corresponding to the angular position and velocity of joint 1 and joint 2, A is a N_A -dimensional association space, and D is a 2-dimensional output space corresponding to the CMAC adaptive neural controller's torque Γ_{CMAC} . The function $S(x)$ maps each point x in the input space onto an association vector $a = S(x) \in A$ that has N_L nonzero elements, with $N_L < N_A$.

For a conventional CMAC, the association matrix contains only binary elements, either zero or one. The function $P(a)$ computes an output vector y by projecting the association vector onto a matrix $W = [W_1, W_2]^T$ of adjustable weights so that each output y_i can be obtained by evaluating the inner product of the two vectors a and W_i with $i=1,2$.

Finally, each actual output y_i is derived as follows:

$$y_i = \Gamma_{CMAC_i} = a^T W_i = \sum_{j=1}^{N_L} a_j \cdot W_{i,j} \quad (23)$$

where a_j represents the j^{th} element of the association vector a and $W_{i,j}$ the j^{th} element of the weight vector W_i with $i=1,2$.

The basic concept of the adaptive CMAC neural network used in the second controller is to produce an output that forms a part of the overall control torque that is used to drive the manipulator joint to track the desired trajectory.

Errors between the joint's desired and actual position/velocity values are then used to train the CMAC neural controller. Training is made online and the weight adjustment ΔW is given by:

$$\Delta W = \frac{\beta \cdot (y^d - y)}{N_L} = \frac{\beta \cdot (K_{pn} \tilde{q}_r + K_{vn} \dot{\tilde{q}}_r)}{N_L} \quad (24)$$

where y^d and y are the vectors of the desired and actual outputs of the CMAC network, respectively. K_{pn} and K_{vn} are positive gain constants. β is the learning rate.

VII. SIMULATION ANALYSIS

Performance of the hybrid controller given in (22) is tested using a dynamic trajectory having a sinusoidal form:

$$\theta_1^d(t) = \theta_2^d(t) = \pi \sin\left(\frac{2\pi}{T}t\right) \quad (25)$$

T is chosen equal to 20 seconds to avoid the destabilization of the control law induced by fast dynamics.

Parameters of the first controller as follows:

$$K_{pr} = \begin{bmatrix} 1 & 0 \\ 0 & 0.5 \end{bmatrix} \text{ and } K_{vr} = \begin{bmatrix} 4 & 0 \\ 0 & 0.8 \end{bmatrix}.$$

Parameters of the second controller are as follows:

$$\mathbf{K}_{pn} = \begin{bmatrix} 0.8 & 0 \\ 0 & 0.4 \end{bmatrix}, \mathbf{K}_{vn} = \begin{bmatrix} 4.8 & 0 \\ 0 & 1.5 \end{bmatrix}, N_L = 32, Q_L = 0.5, \beta = 0.9.$$

Let suppose that the actual values of the parameters of the robot are such as specified in Table 1. To test the robustness of the control strategy proposed, the extreme case where the estimated values of the dynamic parameters of the robot: J_{A_i} , J_{B_i} , M_i , M_{C_i} , ρ_i , E_i and I_i (with $i=1,2$) are considered the tenth of their actual values given in Table 1.

These estimated values are then used to compute \mathbf{A}_r and \mathbf{B}_r and therefore $\mathbf{A}_r\text{NN}$ and $\mathbf{B}_r\text{NN}$.

This will drive the first controller to produce an incorrect torque. It is interesting to see how the second controller deals with this error and how it will correct it.

In order to better appreciate the effectiveness of the hybrid controller given in (6), its results will be compared with the nonlinear controller given in (4).

The goal here is to simulate an important error due to a bad estimation of the dynamic parameters (or ignorance of some of them). It is supposed that if the hybrid controller can handle this important error, it can *a fortiori* handle a small one.

TABLE I. MANIPULATOR CHARACTERISTICS

Physical parameters	Link 1	Link 2
Length (m)	$L_1 = 0.80$	$L_2 = 0.50$
Moment of inertia at the origin of the link (kg m ²)	$J_{A_1} = 1.80 \cdot 10^{-3}$	$J_{A_2} = 1.85 \cdot 10^{-4}$
Moment of inertia at the end of the link (kg m ²)	$J_{B_1} = 4.70 \cdot 10^{-2}$	$J_{B_2} = 0.62$
Mass of the link (kg)	$M_1 = 1.89$	$M_2 = 1.18$
Mass at the tip (kg)	$M_{C_1} = 1.0$	$M_{C_2} = 0.5$
Mass density (kg/m ³)	$\rho_1 = 7860$	$\rho_2 = 7860$
Young modulus	$E_1 = 1.98 \cdot 10^{11}$	$E_2 = 1.98 \cdot 10^{11}$
Second area moment of inertia (m ⁴)	$I_1 = 2.25 \cdot 10^{-10}$	$I_2 = 2.25 \cdot 10^{-10}$

For simplicity on the simulation tests, dynamic parameters are equally bad estimated. Even if it is not always the case on practice, this will not affect the adaptive CMAC controller, which is in charge of compensating these errors, because this controller considers the global error (the resultant of the sum of all errors).

Fig. 5 to Fig. 14 illustrate the results obtained with the hybrid controller applied to the two-link flexible manipulator. These figures describe the evolution of: angular position, error on position, deflection, angular velocity and error on the angular velocity, for the joints 1-2, respectively.

Table 2 and Table 3 present the maximum error and the root mean square (RMS) error on the angular position and velocity obtained with the two types of control strategy used for the joints 1-2, respectively.

Results of the nonlinear control are reported in dashed line for comparison. The desired trajectory (target) is reported on Fig. 5, Fig. 8, Fig. 10 and Fig. 13 in dotted line. For the position control (see Fig. 5 and Fig. 10), it is noticed that the angular trajectory obtained with the hybrid controller matches perfectly the target, with an error of no more than 0.012 rad (0.7 deg) for the first and the second links (see Fig. 6 and Fig. 11), whereas it reaches 1.2 rad (69 deg) with the nonlinear controller (see Tables 2-3).

Velocity tracking with the hybrid controller is good (see Fig. 8 and Fig. 13) and the error induced is acceptable, whereas the nonlinear controller strongly deviates from the target. Velocity error obtained with the hybrid controller is lower than 0.024 rad/s (1.35 deg/s) for the first and the second links (see Fig. 9 and Fig. 14), whereas it reaches 0.696 rad/s (20.9 deg/s) with the nonlinear controller (see Tables 2-3).

The proposed controller deals well with the elasticity of the links (see Fig. 7 and Fig. 12). As the manipulator is on the dynamic phase of the movement, with an imposed position and velocity, the deflection of the links can't be avoided. The purpose here is rather to dampen the vibrations. It is noticed from Fig. 7 and Fig. 12 that whenever vibrations appear they are lessened.

With the nonlinear controller, the presences of high frequency oscillations at the tip of the links and in the angular velocity tracking are noticed. Using the nonlinear controller with faster trajectory and/or more flexible links lead to the destabilization of the trajectory control due to the presence of these oscillations. Here a better behavior of the trajectory tracking system is achieved with the hybrid controller. Results obtained demonstrate the good performance of the proposed control strategy.

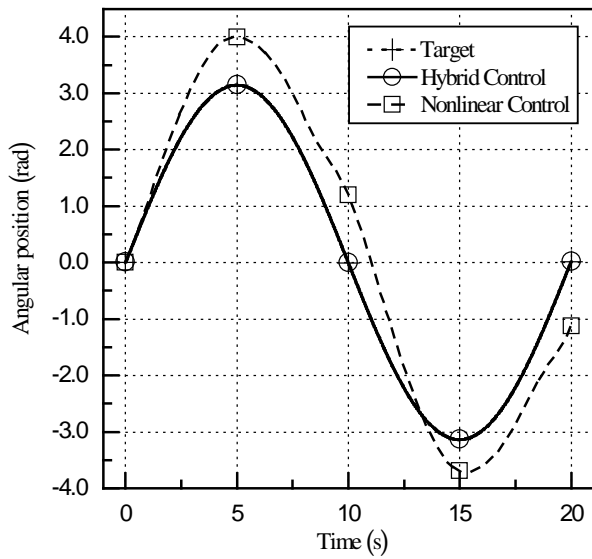


Fig. 5. Evolution of the angular position θ_1

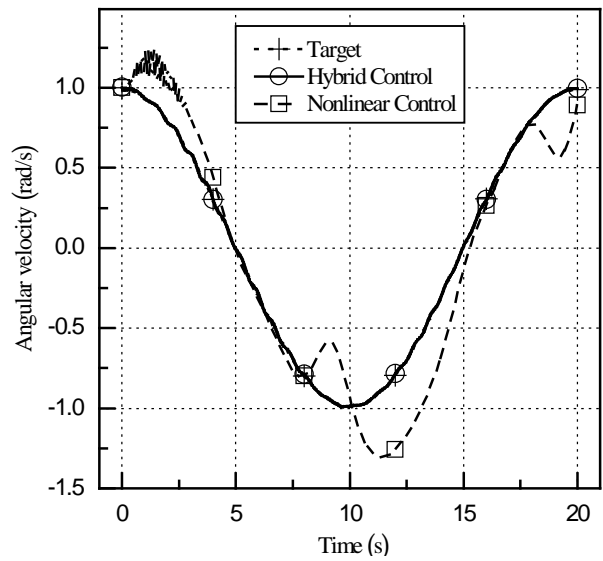


Fig. 8. Evolution of the angular velocity $\dot{\theta}_1$

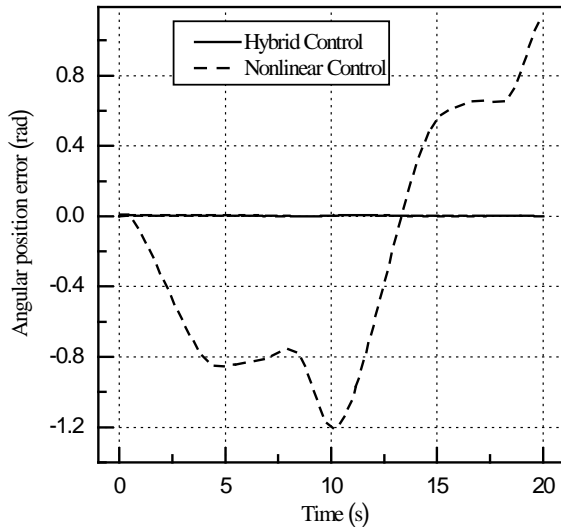


Fig. 6. Evolution of the angular position error $\tilde{\theta}_1$

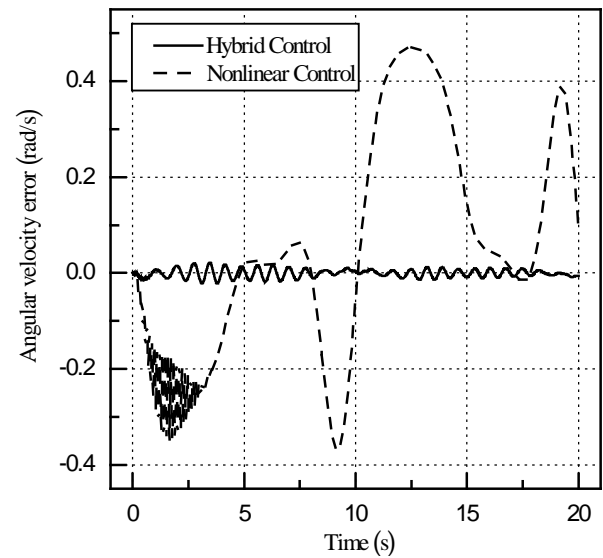


Fig. 9. Evolution of the angular velocity error $\dot{\tilde{\theta}}_1$

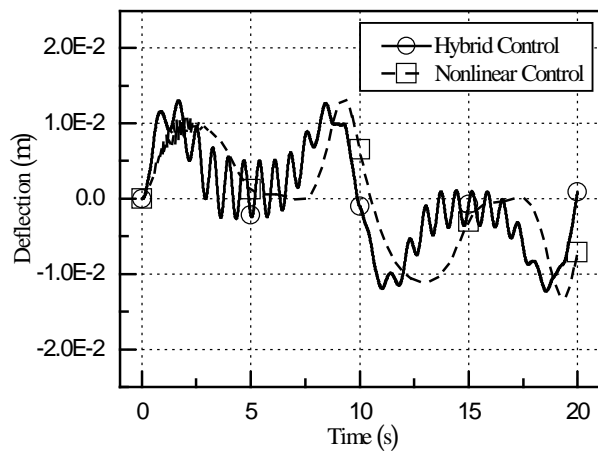


Fig. 7. Evolution of the deflection f_1

TABLE II. TRAJECTORY ERROR ON JOINT 1

Variable	Max. Error		Root Mean Square	
	θ_1 (rad)	$\dot{\theta}_1$ (rad/s)	θ_1 (rad)	$\dot{\theta}_1$ (rad/s)
Hybrid Control	9.86×10^{-3}	2.19×10^{-2}	3.98×10^{-3}	8.65×10^{-3}
Nonlinear Control	1.21	4.70×10^{-1}	7.25×10^{-1}	2.43×10^{-1}

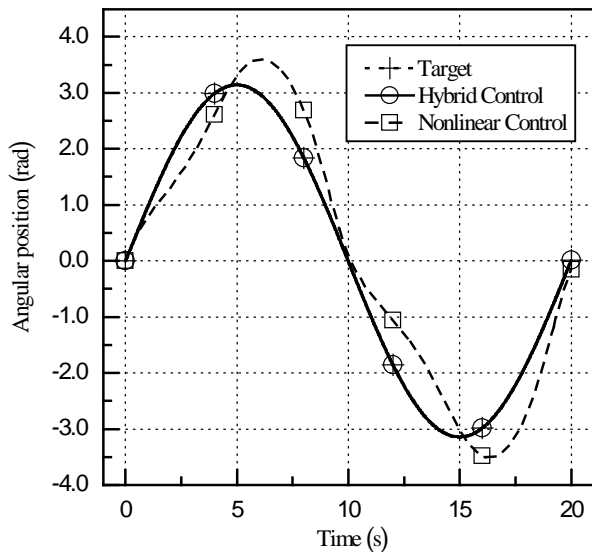


Fig. 10. Evolution of the angular position θ_2

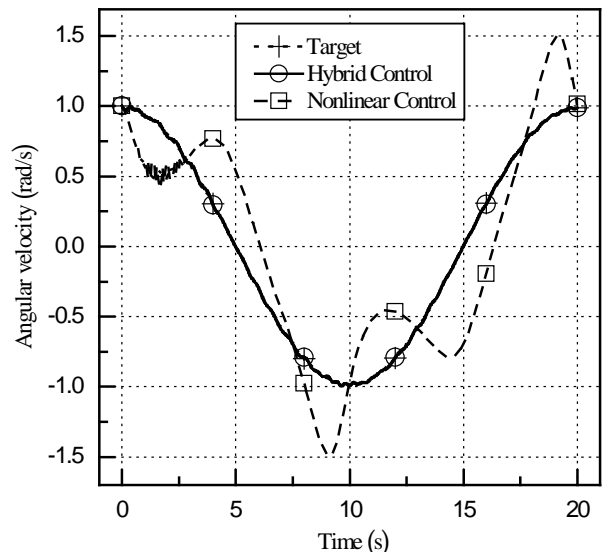


Fig. 13. Evolution of the angular velocity $\dot{\theta}_2$

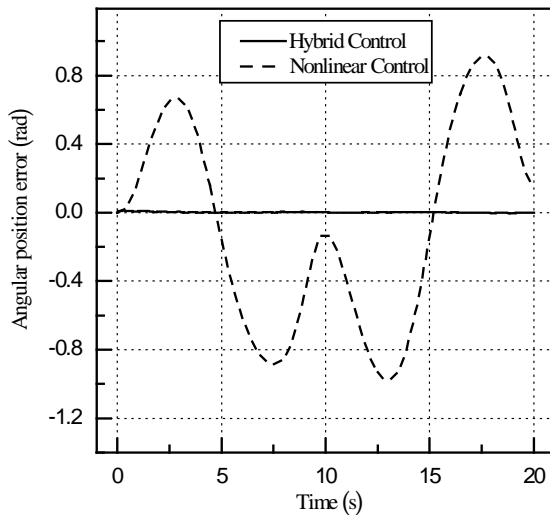


Fig. 11. Evolution of the angular position error $\tilde{\theta}_2$

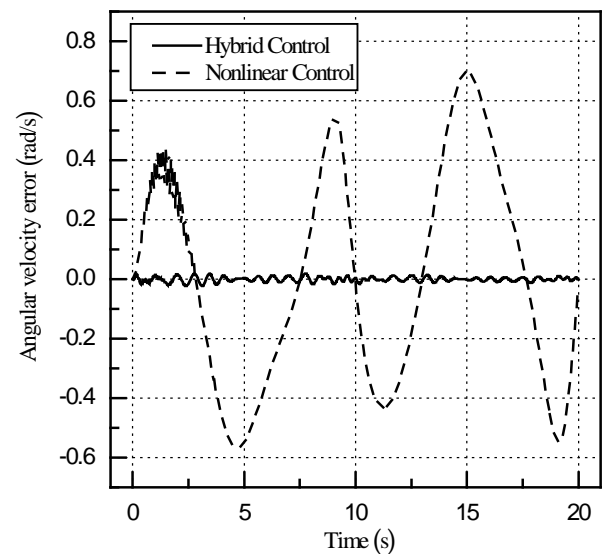


Fig. 14. Evolution of the angular velocity error $\dot{\tilde{\theta}}_2$

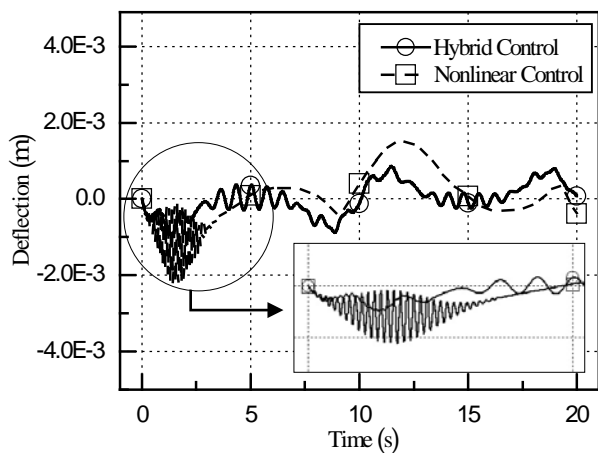


Fig. 12. Evolution of the deflection f_2

TABLE III. TRAJECTORY ERROR ON JOINT 2

Variable	Max. Error		Root Mean Square	
	θ_2 (rad)	$\dot{\theta}_2$ (rad/s)	θ_2 (rad)	$\dot{\theta}_2$ (rad/s)
Hybrid Control	1.17 $\times 10^{-2}$	2.35 $\times 10^{-2}$	3.43 $\times 10^{-3}$	8.40 $\times 10^{-3}$
Nonlinear Control	9.77 $\times 10^{-1}$	6.96 $\times 10^{-1}$	6.07 $\times 10^{-1}$	3.71 $\times 10^{-1}$

VIII. CONCLUSION

The goal here is to search for control techniques that improve the performance of flexible manipulators on the dynamic phase of the trajectory and reduce the computational burden.

The main idea is to combine two control techniques, nonlinear control and neural network control.

The new control strategy presented is composed of two controllers. A static feed forward artificial neural network controller and an adaptive CMAC neural network controller. The first controller is elaborated by approximating the robot's dynamic equation of motion with an artificial neural network and adding a proportional and derivative term. The aim of the first controller is to provide a stable and fast control, based on the dynamic model of the system. While the first controller provides the main of the control, the adaptive CMAC neural network strategy ensure that the real trajectory matches the desired one by compensating errors due to structured and unstructured uncertainties, increasing the precision of the control.

Using ANNs in the place of the nonlinear model allowed simplifying the structure of the controller reducing its computation time and enhancing its reactivity.

Simulation results on a dynamic trajectory with a sinusoidal form show the effectiveness of the proposed control strategy.

Despite obtaining an efficient control structure, current effort does not include a stability analysis of the overall control scheme. Future work includes exploiting sliding mode control to improve stability and performance of the control strategy proposed.

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Performance Testing, and Evaluation for the Voipv6 Network Related Functions, (Sendto and Receivefrom) (An Applied Networking Research Approach)

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Abstract—(The network related functions (Sendto, and Receivefrom) in VoIPv6, are needed to obtain the communication socket in both UDP, and TCP before the communication can take place between the sending and receiving ends. The intent of testing and evaluating the network related functions in Voice over Internet Protocol (VoIPv6) in this research work is not to provide a comprehensive benchmark, but rather to test how well TCP (Transmission Control Protocol), and UDP (User Datagram Protocol) perform in sending and receiving VoIPv6 traffic and bulk data transfer, Part of this, due to the cumulative nature of VoIPv6 performance, can be achieved by testing the network related functions which are the Sendto and Receivefrom socket calls. This is because the sending concept in UDP IPv6 is based on the best effort sending of packets not a guaranteed sending as in TCP IPv6. In this context, performance enhancement techniques are needed to be applied in VoIPv6 due to the fact that there is no dedicated line between the sending and receiving ends. This is actually the putty and the drawback at the same time of VoIP. This is also the reason for IPv6 to take longer time yet to reach its full maturity (Recommendation G.711 of the ITU expectation is by the year 2050) when fully deploying real time applications due to their time sensitivity.

Keywords—(VoIPv6 (Voice over Internet Protocol V6) Performance; Voice over Internet Protocol V6 Performance testing; Voice over Internet Protocol V6 Performance analysis; VoIPv6 quality testing in the protocol and application layers; Internet Measurement Research; and VoIPv6 performance improvements

PROBLEM STATEMENT

The problem statement in this research work in a précised way is that obtaining the Sendto and Receivefrom Socket calls in VoIPv6 contribute to both UDP, and the round trip delay as in TCP, and to the overall delay in VoIPv6 traffic, which is of a cumulative nature. Before any communication session could be established between any (TCP or UDP) client and server, the communication socket must be obtained first then only the sending and receiving of VoIPv6 packets can take place. This research work objective clearly here is to obtain and quantify this amount of delay so that improvement methods can be suggested and applied accordingly based on this. Additionally also to specify the contribution of the network related functions delay amount in the overall less possible delay limit

in VoIPv6. This overall less possible amount of delay was stated by the International Telecommunication Union (ITU) and Internet Engineering Task Fours (IETF), as in the G.114 ITU recommendation.

G.114 is the International Telecommunication Union (ITU) standard recommendation that addressed acceptable delays for voice applications. It is oriented to national telecommunications and is more stringent than what is normally applied in private voice networks. One of the things it addressed is voice delay. The G.114 recommendation of the International Telecommunication Union (ITU), published in the year 1994, stated that < 150 milliseconds delay in voice is acceptable, > 400 milliseconds is unacceptable and in between people should be made aware of quality issue. (Habibullah Jamal, Kiran Sultan, 2008) suggested VoIPv6 improvement in the protocol level, as concurrently in this research work, specifying the contribution of the sendto and receivefrom socket calls is a step before any improvement can be made.

I. INTRODUCTION

In this paper the time needed to obtain the communication socket in VoIPv6 was quantified and studied. The G.711 was the codec technique in the VoIPv6 software that was used in both the VoIPv6 client and server couples in both the source IPv6 and destination IPv6 as (W Richard Stevens 1998) discussed voice transmission under IPv6 in detail. This research work added codes that were developed using socket programming FreeBSD to run the real time tests. The aim here is to run performance tests for VoIPv6 network related functions that are frequently used in both the TCP and UDP client and server programs or codes, functions like socket applications (Client/Server) with combination of functions such as Sendto and Receivefrom, Client/Server codes were developed as in (Asaad A. Abusin, M. D. Jahangir Alam, Junaidi Abdullah, 2012).

II. RESEARCH MOTIVATION

This research work motivation from literature review, can be clearly described and summarized as, there is a need to quantify, and analyze the delay in VoIPv6 caused by different factors contributing to the overall delay as highlighted also in

(Mark Allman of the Internet Measurement Research Group (IMRG) Australian National University, 06 August 2012), one of these factors is the time needed to obtain the communication socket.

III. LITERATURE REVIEW

In (W Richard Stevens 1998) UDP was defined as a transmission protocol in IPv6 applications, since it simply prepares its 8-bytes header and passes the Datagrams to IP, IP prepares its header, determines the outgoing interface by performing the routing functions, and then either adds the datagram to the Datalink output queue or fragment the datagram and adds each fragment to the Datalink output queue. UDP, as part of IPv6, was chosen to be the test environment focusing on testing and performance evaluation of the network related functions in VoIPv6 traffic. (Asaad A. Abusin, M. D. Jahangir Alam, Junaidi Abdullah, 2012) focused on testing and quality measurement and analysis of VoIPv6 performance. A client, server codes were developed using Free BSD, as a step before analyzing the Architectures of VoIPv6 in the current internet in order for it to cope with IPv6 traffic transmission requirements in general and specifically voice traffic, this is being attracting the efforts of research bodes currently.

According to (Kaushik Das, 2008), the quality of a voice-call can degrade significantly if VoIP voice packets are lost or delayed at any point in the network between VoIP users, and users can also notice this quality degradation more in highly congested networks or over long distances. In order to address this quality issue a set of service requirements to deliver performance guarantee while transporting voice traffic over the network are needed. A fundamental fact is that the delay in VoIPv6 is an accumulation of many factors one of which is the time needed to obtain the communication socket, and this is exactly the focus in this paper. (Krishna P. Gummadi, Stefan Sarotu, W.A. Steven D. Gribble, Seattle, and W.A. King, 2002) investigated the ability to estimate network latencies between arbitrary internet end hosts, such ability would enable new measurement studies and applications, such as investigating routing path inefficiencies on a wide-scale or constructing topologically sensitive overlay networks.

(Mark Allman of the Internet Measurement Research Group (IMRG) Australia, 06 August 2012) highlighted a considerable network measurement research work being conducted within the Internet community, both in standards bodies (such as, IETF IPPM WG) and in various research labs. (Geoff Huston of Telstra, GEOFF HUSTON Holds March 2003) razed a big question that is, how good is your network? Or how can the quality of the service that specific network is offering can be measured and quantified? And in this context this research work is an attempt contributing to answering this question which has been lurking behind many IP vendors for many years. With the increasing levels of deployment of various forms of high-speed (or broadband) services within today's Internet there is new impetus to find answers that allow both providers and users to place some objective benchmarks against the current state of the current service offerings, putting in mind the need for a functional description of network performance addressing the distortion of transactions that are carried across the network, one of which

is this research work focus. (Request for Comments (RFC 5181), 2008) provided a detailed description of IPv6 deployment and integration methods and scenarios in wireless broadband access networks in coexistence with deployed IPv4 services in which the main components of IPv6 IEEE 802.16 access networks and their differences from IPv4 IEEE 802.16 networks, were discussed and also how IPv6 was deployed and integrated in each of the IEEE 802.16 technologies.

With some new features introduced by IPv6 network, such as a wider address space, a different scope based on addressing scheme, a modified protocol field, according to (M-K. Shin Ed., Y-H. Han, S-E., Kim, and D. Premec, 30 July 2007) the IPv6 topology discovery faces some challenging problems, one of which is the ability to transmit real time applications including voice, and many of topology discovery algorithms developed to collect IPv4 network topology information cannot be readily applied to IPv6 network. Problems generated when adopting IPv6 network, and some methods and algorithms for IPv6 network topology discovery need to be covered in detail when considering network performance improvements discussing the advantages and disadvantages highlighting also some challenging problems faces IPv6 network infrastructure which affect the overall protocol performance focusing on the network level. (Cisco Support Community, Document ID: 5125, Feb 2, 2006) discussed the voice performance behavior in between transmission rates of 150-to-400 Milliseconds of delay limited by the 150 milliseconds hit value in VoIPv6 as stated by the ITU provided that network administrators are aware of the Echo cancellers are required when one-way delay exceeds a value of 25 milliseconds as in recommendation G.131 of the ITU. For private VoIPv6 networks a conditional value 200 milliseconds of delay is a reasonable goal and 250 milliseconds is a value of maximum conditional limit, provided priory that the maximum expected voice connection delay is known as 150Ms, Table 1 illustrate this.

TABLE I. ITU DELAY SPECIFICATIONS

Range in Milliseconds	Description
0-150	Acceptable for most user applications.
150-400	Acceptable provided that administrators are aware of the transmission time and the impact it has on the transmission quality of user applications.
Above 400	Unacceptable for general network planning purposes. However, it is recognized that in some exceptional cases this limit is exceeded.

ITU G.711, and (Cisco Support Community, Document ID: 5125, Feb 2, 2006) specified limits for the acceptable amount of delay under which the voice quality will be acceptable, and this paper discuss the impact and the percentage that the network related functions contribute to this amount. (Request for Comments: RFC 2460, R. Hinden, Published by the IETF, Category: Standards Track Nokia December 1998) specified version 6 of the Internet Protocol (IPv6), also sometimes referred to as IP Next Generation or IPNG. RFC 2460 specified an Internet standards track protocol for the Internet

community and requests discussion and suggestions for farther improvements. Reviewers, according to RFC 2460, and by referring to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol, IPv6 solves the problem imposed by the limited 32-bit address field in IPv4, and the definition of the new protocol (IPv6) results in longer header size, which affects services that used to be transmitted over IP especially real-time services such as voice. (El-Hennawy, Hadia M S, Ain Shams University, Cairo, Egypt, Gomaa, Haitham H.; Amin, Amany S. 07 April 2010) demonstrated the effectiveness of transmitting voice traffic over IPv6 convergence using simulation results obtained using NS2.

(W. Richard Stevens, 1998) offered guidance tips on making the most of sockets, which are the de facto standard for UNIX network programming as in this research work the socket programming was the performance testing tool which is FreeBSD. (W. Richard Stevens, 1998) begin his book by introducing virtually every basic capability of TCP and UDP sockets, including socket functions and options, I/O multiplexing, also the name and address conversions presenting detailed coverage of the Posix, for example standard for the sockets and the Posix threads and also introduced advanced techniques for establishing both the IPv4/IPv6 interoperability, implementing non-blocking Input /Output routing sockets broadcasting, in addition to the multicasting as IP options, multithreading, advanced name and address conversions, UNIX based and domain protocols, and also raw sockets, such techniques teaches researchers how to choose among today's leading client/server design approaches. This research work tackled this idea using FreeBSD, including both the TCP iterative, concurrent, and Pre-forked and in addition to Pre-threaded servers, in this context the Internet/intranet revolution has dramatically increased in the demand for researchers graduating with a sophisticated understanding of network programming APIs especially sockets helping those researchers and students to achieve that goal.

(Habibullah Jamal, Kiran Sultan, 2008) discussed a possible performance improvement in the protocol level of the dominant transport protocol of today, TCP, which does not meet the demand for high quality voice because it favours reliability Over timeliness and fails to fully utilize the IP network capacity, most certainly this is due to limitations of its conservative congestion control algorithm. This paper rather focus on testing how well UDP (User Datagram Protocol) and TCP (Transmission Control Protocol) performs in sending and receiving VoIPv6 traffic and bulk data transfer, and the slow response of TCP in fast long distance networks leaves sizeable unused bandwidth in such networks. A large variety of TCP variants have been proposed to improve the connection's throughput by adopting more aggressive congestion control algorithms. Some of the flavours of TCP congestion control are loss-based, high-speed TCP congestion control algorithms that uses packet losses as an indication of congestion; delay-based TCP congestion control that emphasizes on packet delay rather than packet loss as a signal to determine the rate at which to send packets. Some other efforts combined the features of loss-based and

delay-based algorithms to achieve fair bandwidth allocation and fairness among flows.

(W. Lei, W. Zhang, and S. Liu (North-eastern University), July 25, 2014) stated that in the year 2014 most multimedia applications utilized a combination of real-time transport protocol (RTP) and user datagram protocol (UDP). Application programs at the source end format payload data into RTP packets using RTP specifications and dispatch them using unreliable UDP along a single path multipath transport as an important way to improve the efficiency of data delivery. In order to apply the framework of multipath transport system based on application-level relay (MPTS-AR) to RTP-based multimedia applications defining multipath real-time transport protocol based on application-level relay (MPRTP-AR), which is a concrete application-specific multipath transport protocol (MPTP), and the packet formats and packet types of MPRTP-AR follow the common rules specified in MPTP profile.

IV. THE AIM OF THIS RESEARCH WORK

The aim of this research is to test and evaluate the VoIPv6 (Voice over Internet Protocol v6) network related functions that are frequently used in both the UDP, and TCP client and the server programs, functions like socket applications (Client/Server) with combination of functions such as Sendto and Receivefrom, this is as part of the accumulated nature of the VoIPv6 overall delay in the same track as in (Kaushik Das, 2008), and (Krishna P. Gummadi, Stefan Sarotu, W.A. Steven D. Gribble, of Seattle, and W.A. King, 2002), and according to (Mark Allman of the Internet Measurement Research Group (IMRG) Australia, 06 August 2012), the Internet is subjective to continuous measurements, testing, and evaluations, and accordingly improvements for its ability to convey and cope with real time traffic.

V. EXPERIMENTS

There is a continuous need to test how good VoIPv6 network is as (Geoff Huston of Telstra, GEOFF HUSTON Holds March 2003) also investigated this issue, his approach was the broad spectrum of the network compared to this research work focus on a specific part of the network which is the network related functions. The first test in this research work was run using developed Free BSD codes for VoIPv6 network; see also (Asaad A. Abusin, M. D. Jahangir Alam, Junaidi Abdullah, 2012). These tests did not measure the whole application performance but only the selected function calls. The time was captured just before the function was called and immediately after the function return, as in the developed FreeBSD code below:

```
Struct timeval timestart, timeend;  
gettimeofday(&timestart, 0);  
int s = Socket(AF_INET, SOCK_STREAM, 0);  
gettimeofday(&timeend, 0);
```

Free BSD developed code to capture the time taken to process the calls.

All tests and experiments were conducted in Network and PCs with the following platform specification and parameters:

The test network platform was MMU Multimedia University Malaysia intranet.

Single Pentium II 450 MHz processor

64 Mbytes 100 MHz Random Access Memory

Free BSD 4.5-Released Operating system.

KAME version 20010528/Free BSD

A. Developing UDP IPv6 Client and Server:

IPv6 UDP source (Client) and sink (Server) source codes were developed, part of the codes are shown below in Code 1 and 2 respectively.

VI. CODES

A. Code 1:

IPv6 UDP Client:

Part of the developed code is shown in code 1 below:

```
#include <stdio.h>
#include <stdlib.h>
#include <unistd.h>
#include <errno.h>
#include <string.h>
#include <sys/types.h>
#include <sys/socket.h>
#include <netinet/in.h>
#include <arpa/inet.h>
#include <netdp.h>
#define MYPORT 4950 // the port users will be connecting
to
int main(intargc, char *argv[])
{
intsockfd;
struct sockaddr_in6 their_addr; // connector's address
information
structhostent *he.
intnumbytes;
char buf6(INET6_ADDRSTRLEN)
#define MYPORT 4950 // the port users will be connecting
to
int main(intargc, char *argv[])
{
intsockfd;
```

```
struct sockaddr_in6 their_addr; // connector's address
information
structhostent *he.
intnumbytes;
char buf6(INET6_ADDRSTRLEN)
if (argc != 3) {
fprintf(stderr,"usage: talker hostname message\n");
exit(1);
}
```

```
if ((he=gethostbyname2 argv[1], AF_INET6)) == NULL) { //
get the host info
perror("gethostbyname");
exit(1); }
```

B. Code 2:

IPv6 UDP Server:

Part of the developed code is shown in code 2 below:

```
#include <stdio.h>
#include <stdlib.h>
#include <unistd.h>
#include <errno.h>
#include <string.h>
#include <sys/types.h>
#include <sys/socket.h>
#include <netinet/in.h>
#include <arpa/inet.h>
#define MYPORT 4950 // the port users will be connecting
to
#define MAXBUFLEN 100
int main (void)
{
intsockfd;
struct sockaddr_in6 my_addr; // my address information
struct sockaddr_in6 their_addr // connector's address
information
intaddr_len, numbytes;
charbuf[MAXBUFLEN],buf6addr [INET6_ADDRSTRLEN];
perror("bind");
exit(1);
}
addr_len = sizeof(structsockaddr) ;
```

```

if ((numbytes=recvfrom(sockfd,buf, MAXBUFLen-1, 0,
(structsockaddr*)&their_addr,&addr_len)) == -1) {
perror("recvfrom");
exit(1);
}
close(sockfd);
return 0;
}

```

In the UDP client code, communication socket pair (Server IP address and port number) must be obtained first then only VoIPv6 stream could be sent to a specific UDP server.

Obtaining the communication socket is crucial to obtain first the pair (IP address and the port number) for the communication to take place between the client and the server. In the client code side obtaining the communication socket using FreeBSD developed code can be:

```

IntsockFd;
If ((sockfd=socket(AF_INET, SOCK_DGRAM, 0))==1) (
Perror("socket");
Exit(1);

```

VII. TEST RESULTS

A test to obtain the communication socket in IPv6 network:

A test was conducted to quantify the duration of time needed to obtain the socket of communication in the TCP and UDP client within Multimedia University (MMU) intranet: Table 2 Shows Different Trails and Tests to Obtain the Required Socket and the Average Time to obtain them.

TABLE II. TIME NEEDED TO OBTAIN THE COMMUNICATION SOCKET IN IPV6 UDP CLIENT IN MICROSECONDS

Test Number	1	2	3	4	5	6	7	Average
Time in Mmilliseconds	19	19	20	18	19	20	18	19

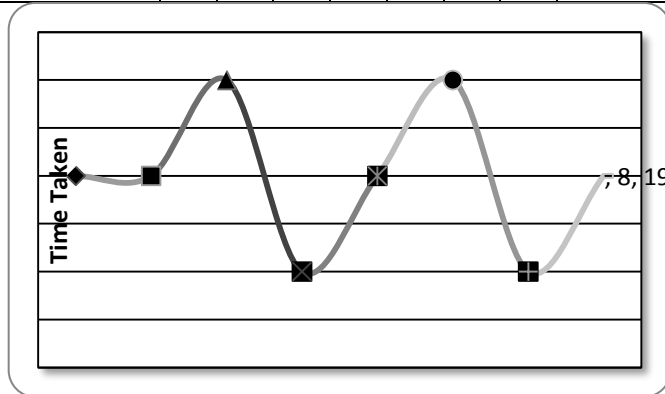


Fig. 1. Time taken to obtain the communication socket in IPv6 UDP client

A. Comparing the VoIPv6 maximum delay limit with the time needed to obtain the communication socket:

When comparing the time needed to obtain the communication socket with the VoIPv6 voice less limit which is 150 Milliseconds according to (Request for Comments (RFC 5181), 2008), the time needed to obtain the communication socket in IPv6 UDP client according to this research work above tests is around 19 Microseconds with contribution of 6% to the less possible amount of delay stated by the Request for Comments (RFC 5181), 2008) above which the voice quality will not be acceptable.

This is the main contribution in this research work which is evaluating and quantifying the time needed to obtain the communication socket and comparing it with the maximum possible delay limit stated by (Request for Comments (RFC 5181), 2008).

The Sendto socket call is needed to send packets and information to the UDP server; this is the case in any UDP client/server application. The Sendto call in the IPv6 UDP client program is shown in Code 1 above. The performance was tested to know the time duration needed to transmit different sizes of data in IPv6 UDP client; Table 3 shows this:

TABLE III. TIME NEEDED TO SEND DIFFERENT PACKET SIZES IN IPV6 UDP CLIENTS

Bytes Sent	100 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	49	48	49	49.7

Bytes Sent	200 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	51	50	51	50.7

Bytes Sent	300 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	53	53	52	52.70

Bytes Sent	400 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	53	54	53	53.3

Bytes Sent	500 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	55	55	55	55

B. Comparing the VoIPv6 maximum delay limit with the time needed to obtain the communication socket in IPv6 UDP Client (Via Native IPv6):

The Sendto socket call of the UDP client was tested under IPv6 native environment with one network switch separating the IPv6 TCP and UDP client and server, the IPv6 TCP and UDP servers were given a test IPv6 addresses, and the client was given a different test address. The tested IPv6 addresses of the client and server were configured first using Inet6 function call as below:

Ipconfig x10 Inet6 1::1, (Where x10 is the network driver card from 3Com)

TABLE IV. THE OBTAINED RESULTS. TIME NEEDED TO SEND DIFFERENT PACKET SIZES USING IPV6 UDP, VOIPV6 CLIENTS THROUGH NATIVE IPV6 TEST BED NETWORKS

Bytes Sent	100 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	70	70	69	69.67

Bytes Sent	200 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	73	75	88	78.67

Bytes Sent	300 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	76	76	76	76

Bytes Sent	400 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	78	89	77	81.33

Bytes Sent	500 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (Native)	76	78	75	76.33

The Performance of the Sendto UDP Socket call in Native IPv6 was tested and the results are shown in Figure 2.

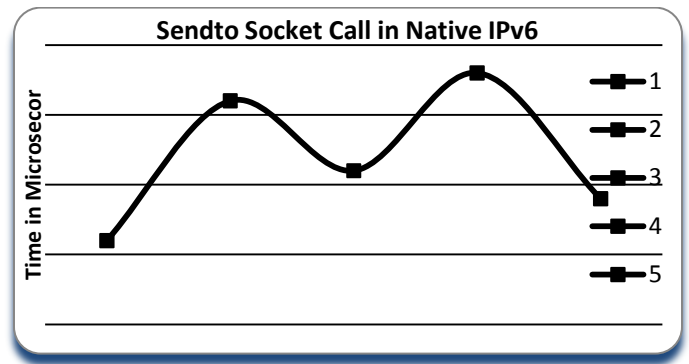


Fig. 2. Performance of the Sendto UDP Socket call in Native IPv6 Network

C. IPv6 UDP Clients (Via Tunneled IPv6 over IPv4):

In order to tunnel IPv6 over IPv4, it is needed first to configure an IPv6 address for the UDP client and the server using the following FreeBSD (UNIX based) line codes:

Ifconfig gif0 create

Gifconfig gif0 source-IPaddress(IPv4) destination IP address(IPv4)

Ifconfig gif0 Inet6 source-IP-address(IPv6) destination IP-address (IPv6) prifixlen/x

The command line Ifconfig here configures the physical address for the generic IP tunnel interface, such as gif0 which is a pseudo device for IPv4-IPv6 tunneling (see FreeBSD Hypertext Man Pages, 2003). The obtained results were shown in Table 5.

TABLE V. TIME NEEDED TO SEND DIFFERENT PACKET SIZES USING DEVELOPED IPV6 UDP CLIENT THROUGH: NATIVE IPV6 TEST BED NETWORK

Bytes Sent	100 bytes			Av.
	TS1	TS2	TS3	
Time in microseconds Ipv6 (tunneled over IPv4)	86	87	86	86.3

VIII. CONCLUSIONS AND FUTURE WORKS

The time taken to obtain the communication socket in IPv6 (TCP, UDP) clients was tested and quantified, and the contribution of this in the less possible amount of delay for VoIPv6 was specified. The VoIPv6 performance was tested in IPv6, in the TCP and UDP transmission protocols in a test bed IPv6 network confirming (M-K. Shin, Ed., Y-H Han, S-E Kim, D. Premec, 30 July 2007) expectation that the VoIP performance might not be the same in a tunneled IPv6 over IPv4 networking system, this is due to the simplicity of the IPv6 header which consists of seven fields compared to the IPv4 header which consist of 13 fields in the header affecting the voice test results in IPv6.

From all the tests and analysis done, one conclusion can be derived: That is the performance of the applications running over IPv6 are affected by the time needed to obtain the communication socket, this is due to the protocol nature, and accordingly any improvement in the protocol level or in the network level will contribute to improving the overall VoIPv6 performance as the long address space of IPv6 adds some overhead to the packet processing time, and the removal of checksum operation at the IP level in IPv6 is not enough to recover the long address processing time. Simpler IPv6 header does not help in this case because experiments in this research work were conducted in local loop environment however, from the theoretical point of view, it will enhance the end-to-end performance if there are many intermediate hosts in between since there will be less processing in the intermediate hosts and nodes. IPv6 uses smaller send and receive buffer compared to IPv4, this may affect the performance since small buffer become full faster. According to the IETF as in (Request for Comments (RFC 5181), 2008) IPv6 is still in its infancy stage experiencing continues renovation. The IPv6 KAME stack is still under development and frequently being changed and updated. The current updated release of IPv6 is not the optimum all the time.

Future work as a continuation to this research work can be carried out in investigating the performance of the implementation of the IPv6 stack in different platforms. According to (Request for Comments (RFC 5181), 2008), KAME IPv6 stack implementation needed continually to be improved to the most optimal level to give better performance results supporting different applications. These performance tests can be repeated in different network conditions highlighting the differences and similarities. The IPv6 features supporting real-time traffic can be clearly noticed as discussed in (Request for Comments: RFC 2460, R. Hinden, Published by the IETF, Category: Standards Track Nokia December 1998), and (El-Hennawy, Hadia M S, Ain Shams University, Cairo, Egypt. Gomaa, Haitham H.; Amin, Amany S. 07 April 2010). The evolving enhancement in IPv6 network with the introduction of Broadband internet service can contribute to better performance, and performance improvements in the real-time applications under IPv6 supported also by (Request for Comments (RFC 5181), 2008).

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fMRI Data Analysis Using Dempster-Shafer Method with Estimating Voxel Selectivity by Belief Measure

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Abstract—In the functional Magnetic Resonance Imaging (fMRI) data analysis, detecting the activated voxels is a challenging research problem where the existing methods have shown some limits. We propose a new method wherein brain mapping is done based on Dempster-Shafer theory of evidence (DS) that is a useful method in uncertain representation analysis. Dempster-Shafer allows finding the activated regions by checking the activated voxels in fMRI data. The activated brain areas related to a given stimulus are detected by using a belief measure as a metric for evaluating activated voxels. To test the performance of the proposed method, artificial and real auditory data have been employed. The comparison of the introduced method with the t-test and GLM method has clearly shown that the proposed method can provide a higher correct detection of activated voxels.

Keywords—Dempster-Shafer theory; fMRI; GLM; t-test; HRF; OTSU method

I. INTRODUCTION

Understanding the way the brain works depends on studying the functional Magnetic Resonance Imaging (fMRI). The study of fMRI time series is related to the activity of neurons in response to an input stimulus during an experiment. However, it would be difficult to notice this activation because it occurs in milliseconds, and it is influenced by noise. To overcome this problem, a contrast method, known as Blood Oxygen Level Dependent, denoted as BOLD for short, is proposed by Ogawa et al. [1] given that the metabolic process increases blood flow and volume in the activated areas during brain activity that are characterized by hemodynamic response functions (HRF). In other words, a local increase in blood flow leads to a different amount of oxygen consumption. The increase in local blood flow permits to map the changes in the areas where oxygen concentrates on the entire brain.

Given this fact, several attempts have been made to gain a better precise classification of voxels regarding activation. Mainly, fMRI data analysis approaches can be classified into two main groups. The first one is the hypothesis-based methods, such as the General Linear Model (GLM) [2]. The other one is the data-driven technique including clustering, Principal Component Analysis (PCA) and Independent Component Analysis (ICA). This latter has proven to provide a practical method for the exploratory analysis of temporal [3] and spatial [4] fMRI data and it is becoming among the useful

methods over the last decade [5]. On another plan, the dimensionality reduction methods have shown their ability in fMRI data analysis [6] since they have been used to reduce some voxels surrounding the brain. Thus, to enhance the detection of activated voxels, it is recommended to perform dimensionality reduction before applying one of the methods mentioned above.

The hypothesis-based approaches have an essential role to play in the analysis of fMRI data because of their complex spatial and temporal correlation structure [3,4]. Recently, many developed approaches have relied on the general linear model (GLM). The latter has become the famous classical model-driven analysis and an excellent technique for analyzing fMRI data. In fact, GLM intends to spot functionally active brain regions and to characterize both functional anatomy and changes resulting from certain diseases [7, 8] because it needs prior knowledge and assumes a canonical hemodynamic response function (HRF) to model a BOLD signal. The main limitation that GLM method suffers from is that it ignores correlation between voxels in both time and space dimensions that are present in fMRI times series [9].

Clustering fMRI time series has emerged as a possible way to detect brain activity. It has been used as an exploratory data analysis technique in fMRI time series. In this case, several approaches based on c-means have emerged [10] such as spatiotemporal clustering analysis of fMRI data [11] and Fuzzy clustering analysis (FCA)[12]. Therefore, hierarchical clustering analysis (HCA) [13] has been gained its place with its ability to produce connectivity map in fMRI data.

Recently, clustering time series [14,15] has newly concerned a considerable amount of research. They have shown their efficiency because they form partitions based on similarity of voxel values in the fMRI time series where each partition is represented by the cluster centroid that is sufficient for the analysis and investigation of fMRI time series [16].

Therefore, clustering techniques can be used to separate activated voxels from non-activated ones. However, methods based on clustering for fMRI data analysis suffer from limitations and relevant problems that need being addressed. Their main drawback lies on their reliability where it is established only with the number of iterations and repeated runs [17].

Accordingly, several methods based on probability theory of Bayesian spatiotemporal model approaches that incorporate spatial correlation among brain responses have been recently found as successful applications in the analysis of fMRI data[18][19]. An alternative method has been recently proposed by *Bowman et al.* [20] Gaussian Markov random field priors on the regression coefficients of a general linear model. Another research has investigated spatiotemporal modalities by hierarchical Bayesian approaches and incorporated the estimation of the HRF [21]. Another work to parameterize the HRF with voxel-varying parameters used Gaussian Markov a random field prior on the activation characteristic parameters of the voxels has been introduced by *Quirós et al.* [22]. All these works assumed an independent structure for the error terms in their models.

In this study, a new approach based on Dempster-Shafer theory of evidence is suggested to improve extracting brain activity. Particularly, the emphasis of this work has been placed on developing a new analytical framework that can provide better detection of the brain regions that show signs of neuronal activity following a stimulus. Also, it aims to infer the association of spatially remote voxels that exhibit fMRI time series with similar characteristics. In the formulation that we adopt, the stimulus pattern is convolved with a hemodynamic response function (HRF). A voxel-dependent shape parameter characterizes the delay between the onset of the stimulus and the arrival of blood to the activated brain regions. Clustering the time course responses via a Dempster-Shafer process is a further feature of the proposed approach.

The rest of the paper is organized as follows: In section 2, a brief simplified description of the hemodynamic response model is given. Then, section three describes the basics of the Dempster-Shafer theory. Afterward, we put the proposed method in details. The conducted experiments on simulation and real data are clearly discussed with the findings of the research in section five. At last, a conclusion is given.

II. HEMODYNAMIC RESPONSE FUNCTION

It is worth to note that BOLD fMRI does not measure the activity of neurons directly. Instead, it measures the metabolic demands of active neurons. Given this fact, to gain a better understanding to neural activity is waiting for further study. Thus, we need to model the evoked fMRI response realized by the so-called hemodynamic response function that is a nonlinear function. In other words, we have to model the BOLD response into an impulse input. The box-car standard, the Gaussians and the canonical model proposed in [2, 23] are some of the several HRF models that have been developed. They have an essential role to play in characterizing the onset of the stimulus. This work focuses on the study of the canonical HRF model. As presented in “Fig. 1”, this model is divided into two parts. The first part describes the Peak whereas the second one is employed to model the Undershoot. A good model for the canonical HRF is obtained by the function whose peak is situated between 4-6 seconds [24]. The relationship between the stimulus and BOLD response, denoted by $y(t)$, is typically modeled as the convolution of the stimulus function with an impulse response (HRF) as presented in the following equation:

$$y(t) = s(t) \otimes h(\tau) = \int_0^T s(t - \tau)h(\tau)d\tau \quad (1)$$

Where $h(t)$, $y(t)$ and $s(t)$ denote HRF, the result and the unprocessed fMRI signal respectively

The convolution result is known as epochs in SPM (Statistical Parametric Mapping) [9]. The canonical HRF performs well in many experiments. However, some activated voxels are ignored because the real HRF varies in different people and in different brain regions of the same person as well [25]. To address this problem, this work introduces a new framework based on the Basic Belief Assignment probability as will be closely illustrated.

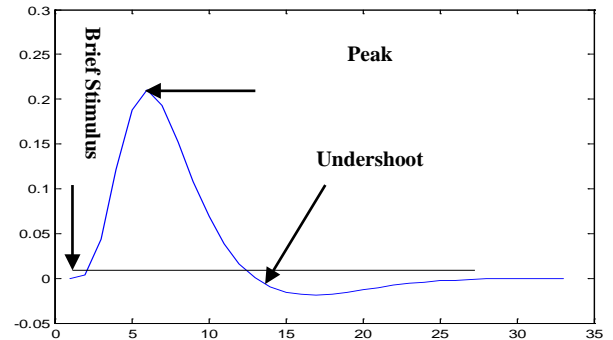


Fig. 1. Model of Canonical hemodynamic response

III. THE DEMPSTER-SHAFER THEORY OF EVIDENCE

In the following, we introduce the fundamentals of the Dempster-Shafer (DS) theory of belief function that has been proven to be an efficient tool in representing uncertain knowledge. This theory has paved the way for many researchers to study various aspects related to uncertainty and lack of knowledge and has shown its ability to solve real problems [26]. In fact, Dempster-Shafer theory can be considered as a generalization of the probability theory [27]. The references [28] [29], [30], [31] provide further information about this theory. In what follows closely, a brief introduction to the basic notions of the theory of evidence is given.

Let $\theta = \{\theta_1, \theta_2, \dots, \theta_k\}$ be a finite set of possible hypotheses. This set is referred to as the frame of discernment, and its power set is denoted by 2^θ where:

$$2^\theta = \{\emptyset, \{\theta_1\}, \{\theta_2\}, \dots, \{\theta_k\}, \{\theta_1 \cup \theta_2\}, \{\theta_1 \cup \theta_3\}, \dots, \theta\}$$

A key point of the evidence theory is known as Basic Belief Assignment (BBA)[19]. It is defined as:

A basic belief assignment m is a function that assigns a value in $[0, 1]$ to every subset A_i of Θ and satisfies the following:

$$m(\emptyset) = 0, \text{ and } \sum_{A_i \in \Theta} m(A_i) = 1 \quad (2)$$

The BBA (m) is associated with the belief function, denoted by $bel(\cdot)$. The definition of belief function is given as in [19]. A belief function assigns a value in $[0, 1]$ to every nonempty subset D of Θ . It is called degree of belief in D and is defined by

$$bel(A_i) = \sum_{A_j \in D} m(A_j) \quad (3)$$

The function, $pl(\cdot)$, associated with the BBA $m(\cdot)$ is a function that assigns a value in $[0, 1]$ to every nonempty subset D of Θ . It is called “degree of plausibility in D ” and is defined by

$$pl(A_i) = \sum_{A_i \cap D \neq \emptyset} m(A_i) \quad (4)$$

Furthermore, a BBA can also be viewed as determining a set of probability distributions over Θ so that $bel(A) \leq P(A) \leq pl(A)$. It can be easily seen that these two measures are related to each other as follows:

$$pl(A) = 1 - bel(\bar{A}) \quad (5)$$

Therefore, one needs to know only one of the three values of m , bel , or pl to derive the two other ones, where \bar{A} stands for the negation of a hypothesis A shown in “Fig. 2”

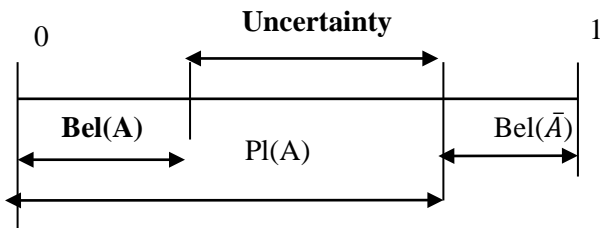


Fig. 2. Basics measures of Dempster-Shafer Theory of Evidence [19]

IV. PROPOSED METHOD

A. Overview

Figure “Fig. 3” illustrates the proposed scheme for fMRI data analysis and detection of activated area that is composed of five stages: i) data preprocessing and dimensionality reduction ii) HRF modeling iii) convolution with fMRI signal, iv) computation of the $m(\cdot)$, the belief function $bel(\cdot)$ for each voxel and v) separation of the activated voxels from non-activated ones using threshold by using OTSU thresholding method because it permits to get a threshold automatically .

B. Data preprocessing

Prior to analysis, fMRI data goes through a series of preprocessing steps to identify and remove the artifacts and to validate model assumptions as well. First, the fMRI slices have been spatially realigned. However, spatial smoothing may cause unforeseen changes to occur into the data. Thus, spatial smoothing has been avoided to ensure better performance. Then, the mean value has been subtracted from each of the time series and the variance has been normalized to a unit. The previous steps were realized via SPM tools[9].

C. Modeling HRF by Dempster-Shafer method

We model a peak and a subsequent undershoot of canonical hemodynamic response function by DS method using the sum of two gamma functions known by the density of probability function, as described above.

The modeling process of the HRF function has been performed as follows: HRF function has been partitioned into two hypotheses(θ_i, θ_j). The hypothesis θ_i corresponds to both detecting neural activation and determining a peak (on activation) while θ_j is assigned for modeling undershoots (off

activation). Each hypothesis is a sum of degrees of beliefs. In particular, the focus of this work lies on the first hypothesis. This latter is divided into two parts A and D , where A stands for degrees of belief included in D . $(D-A)$ denotes the uncertainty part. “Fig. 4” illustrates the proposed model.

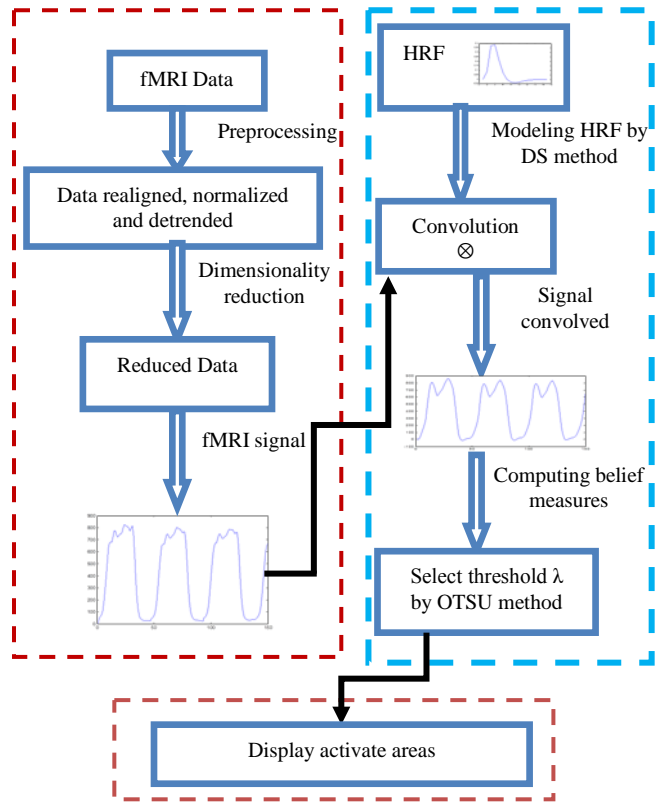


Fig. 3. Flow chart of the proposed model

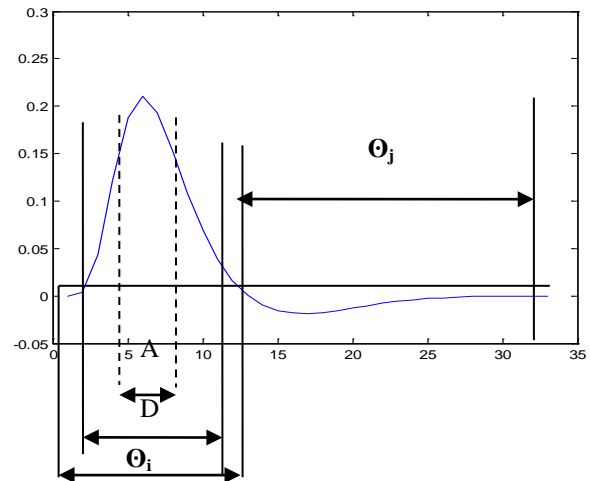


Fig. 4. Flow chart of the proposed model of HRF

At first, we localize the interval of stimulus. In the example where the Time response ($Tr=4$), the peak is in the first interval $[4..8]$ seconds[34]. To find a second stimulus in this example, 16sc have been added. In the second step, we determine the next interval and so on. The same process has been repeated till the end of fMRI signals. Finding these intervals is the focal aim

of this conception. “Fig. 5” illustrates the projection of the proposed model with fMRI signal to extract all intervals.

D. computing the basic belief assignments and the belief measure

After the convolution process, the $m()$ of each time (second) in fMRI times series must be first computed in order to compute the belief and the plausibility measures. Thus, the formula that consists in a transformation of each fMRI signal into a density probability function is described below. The used integral has the form:

$$\alpha = \int_0^{tn} |y(t)| dt \quad (6)$$

In the above equation, the global surface is denoted as α . So, $m()$ is calculated as follows :

$$m(t_i) = \frac{1}{\alpha} \int_{t_{i-1}}^{t_i} |y(t)| dt \text{ for } i = 1 \text{ to } n$$

$$m(t_i) = 0, \quad i = 0 \quad (7)$$

A vector of probability have been obtained where the sum of mass bribability function ($m()$) is 1 as mentioned above in section 3. To compute the belief and plausibility measures, the formulation described in (3) and (4) has been employed.

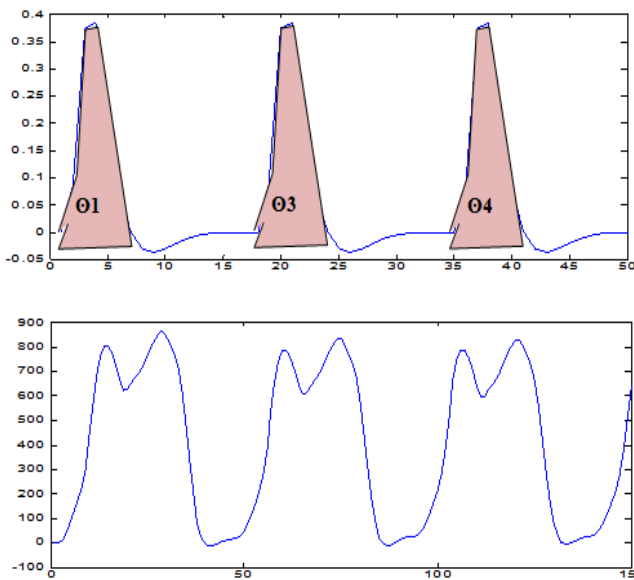


Fig. 5. the projection of the introduced model with fMRI data (where θ corresponds to a finite set of possible hypotheses)

E. Separating the activated voxels from the non-activated ones

To extract activated voxels, the belief measures have been employed in this stage. Each voxel of fMRI time series is presented by $bel()$ value. At first, the histogram of belief measures has been used and an appropriate threshold denoted as λ has been chosen. The OTSU method [33] was employed to choose (λ) threshold. It permits extracting an automatic threshold that minimizes the weighted within-class variance $\sigma_w^2(t)$. This turns out to be the same as maximizing the between-class variance $\sigma_B^2(t)$. The algorithm is as follows:

Step 1 compute the histogram of $bel()$ measure and the probability at each i level of histogram

Step 2 initialize the $\mu_i(0)$ and $q_i(0)$

Step 3 Browse all possible thresholds $t = 1$ to n

- Update $\mu_i(t)$ and $q_i(t)$
- Compute $\sigma_B^2(t)$

Step 4 $\lambda = \max(\sigma_B^2(t))$

where the weighted within-class variance is:

$$\sigma_w^2(t) = q_1(t)\sigma_1^2(t) + q_2(t)\sigma_2^2(t) \quad (8)$$

And the between-class variance is:

$$\sigma_B^2(t) = q_1(t)[1 - q_1(t)][\mu_1(t) - \mu_2(t)]^2 \quad (9)$$

The total variance is:

$$\sigma^2 = \sigma_w^2(t) + \sigma_B^2(t) \quad (10)$$

where the class probabilities are estimated as:

$$q_1(t) = \sum_{i=1}^t p(i) \quad (11)$$

$$q_2(t) = \sum_{i=t+1}^n p(i) \quad (12)$$

And the class means are given by:

$$\mu_1(t) = \frac{\sum_{i=1}^t ip(i)}{q_1(t)} \quad (13)$$

$$\mu_2(t) = \frac{\sum_{i=t+1}^n ip(i)}{q_2(t)} \quad (14)$$

The individual class variances are:

$$\sigma_1^2(t) = \sum_{i=1}^t [i - \mu_1(t)]^2 \frac{p(i)}{q_1(t)} \quad (15)$$

$$\sigma_2^2(t) = \sum_{i=t+1}^n [i - \mu_2(t)]^2 \frac{p(i)}{q_2(t)} \quad (16)$$

And $[0, n-1]$ is the range of intensity levels of the histogram.

F. evaluation metrics and proposed algorithm

This subsection describes the metrics of evaluating the proposed approach and the proposed algorithm based on DS theory.

1) Evaluation metrics

The threshold λ has been used to compute two metrics, the true and false activation rate. These two terms need to be defined herein: True activation rate (TAR) stands for the ratio between the number of time series correctly identified as activated and the total of truly activated time series. And the other one is false activation rate (FAR) referring to the ratio between the number of time series incorrectly identified as activated and the total number of truly non-activated time series. Also, these two ratios serve to analyze the performance of the proposed approach and to establish a comparison with the previous conducted studies like the GLM. It has been noticed in the presented work that the voxels with belief measure more than or equal to λ have been considered to be true active voxels. And the voxels that are less than the selected threshold have been considered as false active voxels. This process leads to obtain the activated regions.

2) Algorithm DS fMRI analysis

To sum it up the proposed algorithm is illustrated as follows:

INPUT: Tr : time response, h(t) : hemodynamic response function,

s(t): fmri signal with nb : size of signal

[a,b] : First_intrval and T : period

OUTPUT:

Y(t) : fMRI signal convolved with HRF

m(t) : Basic Belief Assignment

bel(n) : belief measures

λ : belief threshold for extracting activated voxels

Description:

For each signal fMRI s(t) do

{

$$y(t) = \text{conv}(s(t), h(t))$$

(*convolution fMRI signal with HRF*)

}

For every y(t) do

{Compute m(t) by using equations (6) and (7) }

(*compute belief measures*)

s = 0

For k=0 to nb do

{

$$s = s + m((a + k \times T) : (b + k \times T))$$

(*from (a+k×T) to (b+k×T)*)

}

$$\text{bel}(t) = s \quad (*\text{extract activated voxels} *)$$

{

- Show histogram of belief measure
- Choose λ OTSU method
- show the activated region

}

}

V. RESULTS AND DISCUSSION

This section describes fMRI data that have been used in the conducted experiments .Both artificial and real fMRI data have been employed to determine the identically activated areas. To test the performance of the presented approach, a comparison of the obtained results with the GLM and t-test results has been performed. It is worth mentioning that the tests have been conducted on the same benchmark. This comparison has been done by using the true activation rate and false activation rate as defined above. However, we illustrate plots of true and false activation rates at different belief thresholds.

A. Artificial data

This section describes a form of artificial data used by Francois et al. [11]. In general, fMRI signal is a stochastic process. So, a synthetic three-dimensional fMRI dataset (64, 64, 64) has been generated. The number of slices is 64 and each signal is generated by the following formula:

$$A(t) \times e^{i\odot(t)} + n_c(t) \quad (17)$$

The above function is a complex signal where A(t) stands for the amplitude. Let M be the levels of activation and let φ be the Gaussian random delay distributed with zero mean and unit variance. Let ω be the frequency of the signal and selected to be $\pi/10$ because the fMRI signal is relatively weak. The amplitude on such a basis is defined by a sinusoidal function as follows:

$$A(t) = M \times \sin(\omega t + \varphi) \quad (18)$$

We consider $\odot(t) = \pi/4$ where the real and imaginary channels play a symmetric role and $n_c(t)$ are the complex Gaussian white noise centered with unit variance. The phase of this signal is not used, and we only consider the magnitude:

$$s(t) = |A(t) \times e^{i\odot(t)} + n_c(t)| \quad (19)$$

We generate a set of signals in order to build sequences of fMRI time series as shown in “Fig. 6”.

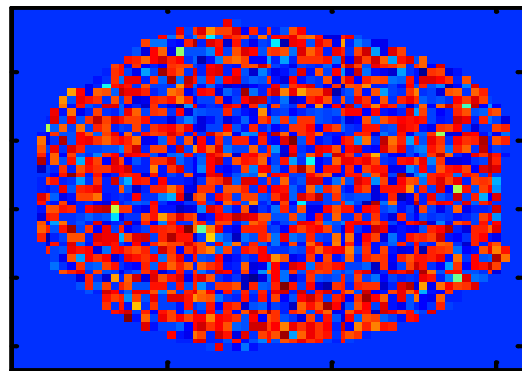


Fig. 6. The artificial data

After applying the proposed approach on this artificial data, the obtained results are presented in “Fig. 7” where brown areas stand for the activated voxels.

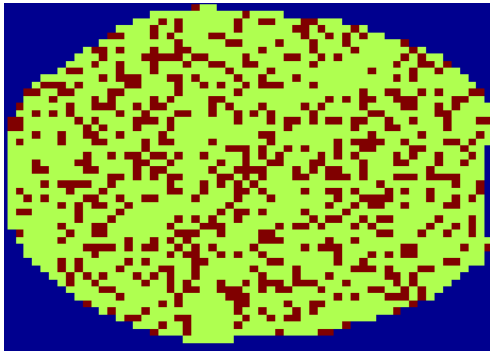


Fig. 7. The activate voxels with artificial data by the proposed approach

Figure “Fig. 8” presents some results using the simulated data described above. These results contain the TAR and FAR measures that have been obtained with $bel()$ threshold. The t-test statistic method has been used in order to compare the introduced method. Basically, t-test statistic method have been used to compute the TAR and FAR metrics at each p-value between 0.001 and 0.05. At first, the fMRI time series are divided in two groups i.e. determine the fMRI time series called on activation denoted as (XON) and the fMRI time series called off activation denoted as (XOFF). To determine these both groups for all fMRI signals the box-car hemodynamic response function have been employed as kernel. The plots in “Fig. 9”, present the TAR and FAR obtained results with t-test method

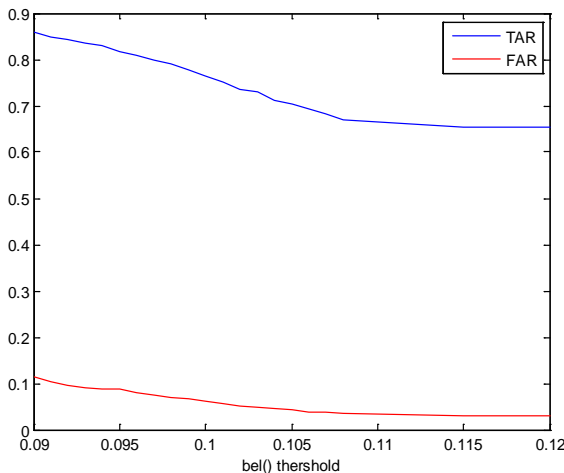


Fig. 8. The plots show the false activation rate and the positive activation rate using belief threshold

These results show that when the p-value is smaller, we get nearer to the activated areas and the number of false activation rate increases. Where p-value is near to 0.05, more precision is obtained in activation rate with less false activation rate. Table1 presents the mean of TAR and FAR for the proposed method and t-test method.

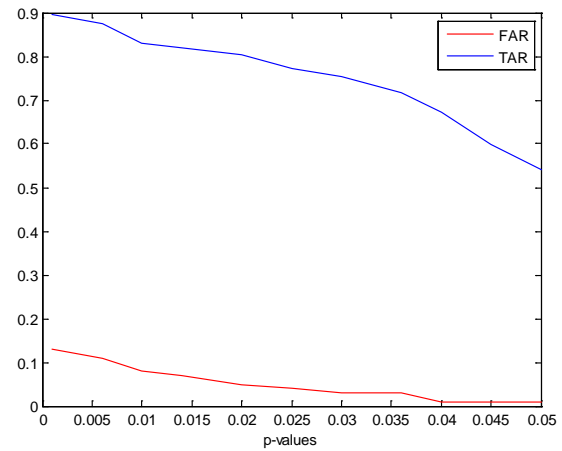


Fig. 9. true and false activation rates by t-test method

TABLE I. AVERAGE OF TRUE ACTIVATION RATE AND FALSE ACTIVATION RATE OF THE PROPOSED METHOD AND T-TEST METHOD TO THE ARTIFICIAL DATA SET

	AVG of TAR	AVG of FAR
DS method	0.766	0.063
t-test	0.753	0.082

Accordingly, these results obviously show the ability of the presented approach to detect true and false activation rate better than t-test method.

B. Real fMRI Dataset

This section reports the result of proposed method tested on a real fMRI dataset that concern an auditory stimulus. This data was collected by Geriant Rees et al. and are available in <http://www.fil.ion.ucl.ac.uk/spm/data>. These whole brain BOLD/EPI images were acquired on a modified 2T SIEMENS MAGNETOM Vision system. Each acquisition is composed of 64 contiguous slices (64x64x64 3mm x 3mm x 3mm voxels) where any acquisition occurs in 6.05s, with the scan to scan repeat time (TR) set arbitrarily to 7s. So that, 96 acquisitions were made (TR=7s), in blocks of 6, giving 42s blocks. Starting with rest, the condition for successive blocks alternated between rest and auditory stimulation that was bi-syllabic words presented binaurally at a rate of 60 per minute. In this experiment, the authors mentioned that the functional data starts at acquisition 4.

After modeling the HRF by Dempster-Shafer method (DS), a basic belief assignment, denoted as $m(v_i)$, has been calculated for all subset A_i of θ (where v_i stands for i th voxel). We have noticed that all fMRI signals have a similar pace with a difference in values of $m()$ which plays a primordial role in computing belief measures. This latter is used to characterize the voxel activity. Then, computing the belief measures enables to obtain the results at (TR = 7) which is in [0.2702, 0.3338]. To separate the activated voxels from non-activated voxels, the histogram described in Figure “Fig. 10” has been used and the threshold of belief measures (λ) by OTSU method has been selected automatically.

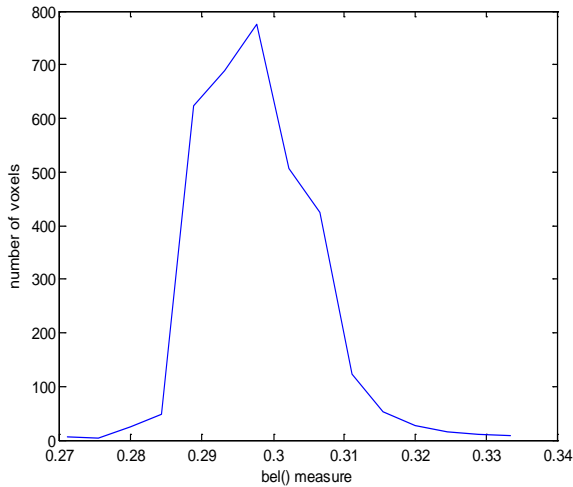


Fig. 10. The histogram of belief measures

To study the influence of the belief measure that presents the key parameters of proposed method, on the whole performance, we generate and plot activation regions (number of voxels) at different values of belief thresholds more than λ obtained by OTSU method. In addition, we measure TAR and FAR values given by belief threshold. Figure “Fig. 11” shows the number of true and false activation rate as a belief threshold. This experiment shows that the proposed method can identify more TAR with less FAR when $bel()$ near to λ . And the number of true and false activation rate tends to be lower when $bel()$ threshold between 0.292 and 0.3

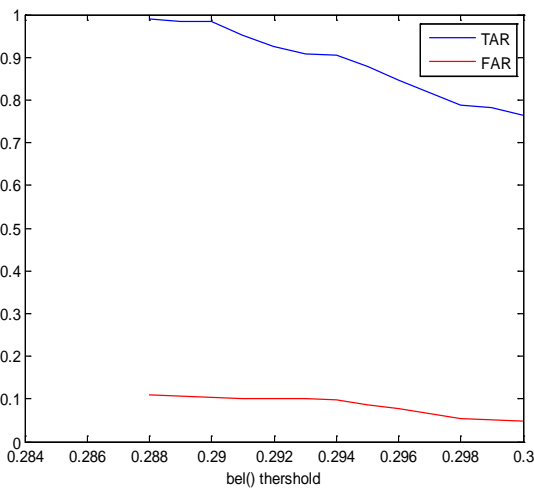


Fig. 11. false and true activation rate obtained by DS method

C. Comparison with GLM method

This section provides a comparison of the GLM results with the obtained results of the introduced framework. Firstly,

the GLM results realized by SPM tools assumes that the fMRI time series correspond to the realization of an identically independent stochastic process and divides data into two groups, obtained during on (activation) and off (no activation) periods. This separation is done by p-value (0.05) and (0.001). The GLM results have shown the different projection (axial, coronal and sagittal) as well as the design matrix (see Fig. 12). Another important feature that distinguishes the GLM method is the long time needed for completing the job. In contrast to GLM, the method based on Dempster-Shafer theory is not difficult to understand and it is easy to implement but it needs to have prior knowledge about the experiment conditions.

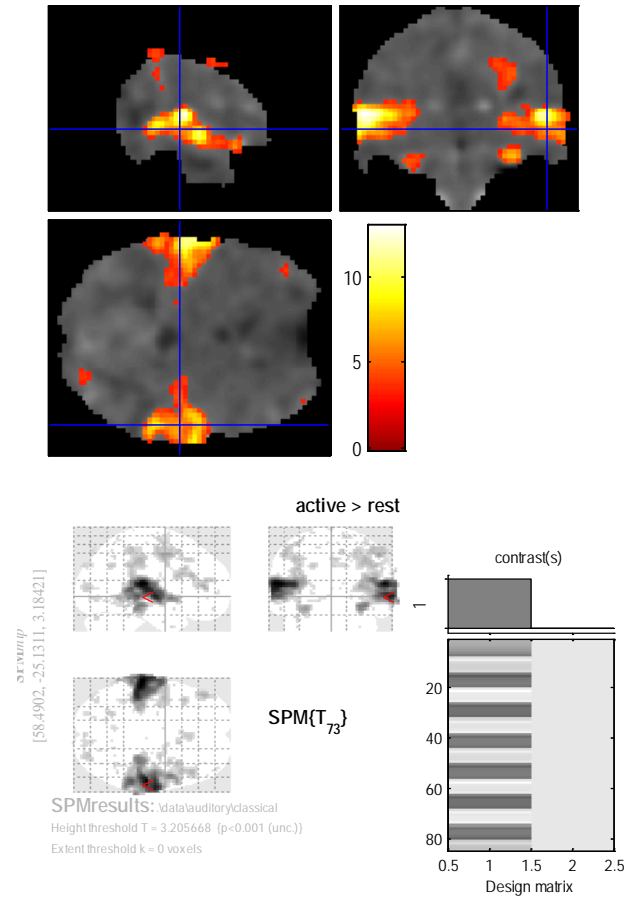


Fig. 12. The obtained result with GLM method

However SPM tools provide the metrics true activation rate and False Discovery Rate (FDR) that plays an important role as well as false activation rate. In other words, it is a proportion of activated voxels that are false positives [34]. However, we proceed to use p-value between 0.001 and 0.05 by using SPM tools that provides the results of FDR and the number of voxels detected activated in the regions used to compute the true activation rate measures. Figure “Fig. 13”, shows the result of this experiment.

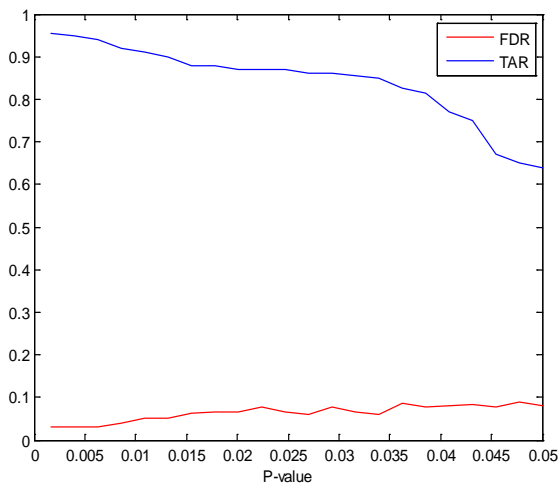


Fig. 13. false discovery rate and true activation rate obtained by GLM method

To sum it up, table 2 presents the average of TAR and FAR measures by the presented method and GLM method

TABLE II. AVERAGE OF TRUE ACTIVATION RATE AND FALSE ACTIVATION RATE OBTAINED BY APPLYING THE PROPOSED METHOD AND GLM TO THE REAL DATA SET

	AVG(TAR)	AVG(FAR)
DS method	0.896	0.056
GLM	0.887	0.071

This experiment shows that the proposed method based on DS theory outperforms the GLM method in identifying more true activation rate with low false activation rate. Figure “Fig. 14” presents some slices that show the activated areas by both method GLM and the proposed method

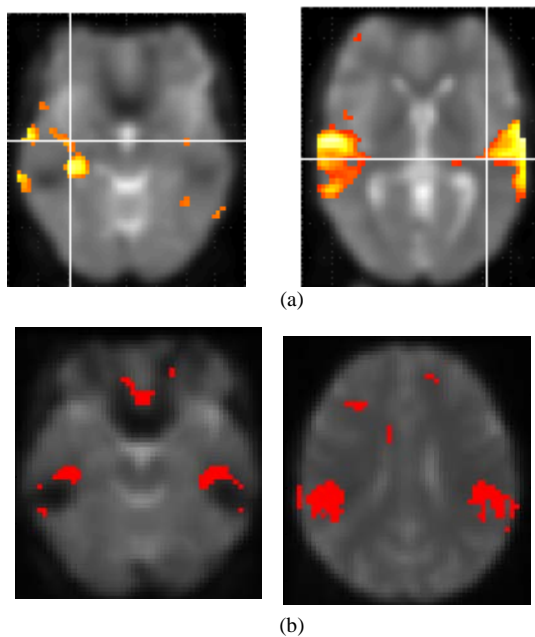


Fig. 14. some slices illustrate the activated regions: a) the result is generated by GLM method; b)the result is generated DS method

VI. CONCLUSION

This paper introduces a new analysis based on Dempster-Shafer theory (DS) that better separates activated voxels from fMRI time series by using basic belief assignment functions. The proposed approach aims to extract activated areas from fMRI data sets. Mainly, information background is required about the hemodynamic response model at the beginning. The introduced method has been validated on a real auditory fMRI dataset as well as on an artificial dataset and its performance has been compared with GLM method. The obtained results have clearly shown the ability of belief measures to yield a better clustering of activated voxels. From the outcome of this investigation, it is possible to conclude that the proposed framework can be employed in most fMRI data analysis methods. Also, the findings suggest that the theory of evidence can serve to understand the nature of data and to obtain relevant results that can be used and interpreted by neuroscientists. The future work aims to use DS theory in analyzing fMRI-EEG data fusion to take advantage of both modalities in order to better study the brain activity.

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Statistical Quality of Service to Increase Qos/Qoe of IP-Based Gateway for Integrating Heterogeneous Wireless Devices

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Abstract—In broadcast service area above communications supported cellular wireless networks, data is communicated to several addressees from a right of entry point/base station. Multicast significantly progresses the network effectiveness to dispense data to multiple addressees as associated to multiple unicast gatherings of the similar data to each receiver independently, by taking improvement of the communal nature of the wireless intermediate. These algorithms need to be intended to be responsible for the essential Quality of Service (QoS) towards an extensive assortment of applications while permitting seamless roaming between multitudes of access network technologies. This paper proposed a cellular-aided mobile ad hoc network (CAMA) structural design, in which a CAMA representative in the cellular network accomplishes the control data, while the data is transported over the mobile terminals (MTs). The routing and security info is substituted among MTs and the negotiator over cellular radio channels. A location centered routing protocol, the multi-selection greedy positioning routing (MSGPR) protocol, is projected. This novel feature makes it more appropriate in the actual world. In accumulation, dynamic new call blocking possibility is initially familiarized to make handoff decision for wireless networks. This paper proposes a novel technique to afford QoS sustenance by means of an assistant network to recuperate the failure of multicast data in the major network. A wireless device might misplace some of the multicast records send above the major network. The experiment results have exposed that the proposed algorithm outclasses traditional algorithms in bandwidth deployment, handoff dropping rate and handoff rate.

Keywords—*Heterogeneous Wireless Mobile Networks; Ad Hoc Networks; Cellular Networks; Quality of Service; Security; Wireless Networks*

I. INTRODUCTION

Upcoming wireless technology aims at providing a sunshade of services to its operators. Ad hoc systems have developed attractive for their potential for commercial relevance. Routing within ad hoc network is a challenge caused by the mobility of clients and the lack of fundamental control. Several routing protocols are proposed. These approaches agonize in network presentation that consists of large routing directly above, low throughput, and large end-to-end interruption. During ad hoc networks, the concerns of quality of service (QoS) and precautions are extra convoluted for the reason that lack of reliable approaches to

allocate information in the entire network. With the accumulative admiration and demand for wireless connection-based multimedia services and the erudite competences of mobile devices, mobile multimedia multicast/broadcast services have developed a significant element of wireless system. Individually the majority of significant features of wireless multicast are highly well-organized interactions as of the mutual environment of the wireless medium. At present 2.5G and 3G cellular networks are contribution multimedia services similar to Mobile TV. Broadcasting networks particularly intended for mobile communications like DVB-H [1], DMB [2] along with Media FLO [3] are presently further down deployment. In accumulation to these devoted mobile broadcasting networks, 3G cellular systems have been protracted to support Multimedia Broadcast Multicast Service (MBMS) [4]. One experiment in provided that such services be towards assurance the reception dependability of many multicast receivers since the wireless links are inaccuracy prone and numerous receivers knowledge heterogeneous channel circumstances.

The multicast service area in several networks for example Digital Video Broadcasting (DVB) and 3G multimedia broadcast/multicast facilities [5], [6] do not afford any opposite communication channel for the receivers to demand the retransmission of lost data. In several wireless multicast/broadcast systems, forward error correction codes (FEC) are cast-off to preserve multipath fading interference and decrease the package faults. Though the wireless channel circumstances are period varying and the multiple receivers in a multicast involvement heterogeneous channel environments. The FEC codes are frequently intended for the poorest channel conditions to confirm satisfactory reception quality for all the receivers in the anticipated service area. This results in a huge above in expressions of radio possessions in infrastructure-based multicast networks. Additional method to progress consistency and throughput is to use numerous projections. Conversely, this method suffers high cost and difficulty for wireless structures at the base station and the wireless devices. Consequently it is a key and stimulating task to sustenance good quality multicast provision to multiple mobile receivers while capably developing radio resources and refining the data and QoS of infrastructure-based cellular wireless networks.

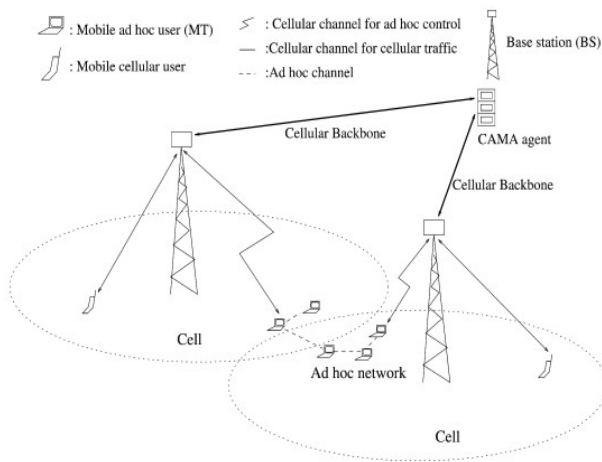


Fig. 1. Cellular-aided mobile ad hoc network

In this manuscript we are propositioning a solution through an integrated categorization uncontainable to packet loss, to provide high-quality multicast services over wireless networks.

The proposed structure contributions the major communications network (e.g., 3G) by enthusiastically conception an accommodating associate recuperation network (e.g., WiFi) between neighboring devices that make use of the identical service. For each device is fortified with two physical radio crossing point. One crossing point is associated to the major network and is answerable for receiving the downstream multicast statistics. The supplementary interface is cast-off to setup the subordinate recovery network. The devices recuperate missing multicast data packets in their foremost network by demanding them as of their recovery network. A device, by originating a supportive recovery technique in its subordinate recovery network, requirements and receive missing data from the devices that are in the locality. These adjacent devices use the identical service and have appropriately established the specific data packets. In this manner the devices recuperate lost data and therefore increase the QoS for the specific service.

II. RELATED WORK

Numerous methods have been proposed to develop the quality and productivity of the cellular network/infrastructure with the help of an ad hoc network. In the reporting system [7], and mobile devices with superior quality link to work as base station transmit for devices with poor link quality. In this classification, is used as an interface to a wireless one for each of the relay and infrastructure method. Bounded by the total cell productivity achieved in this hybrid network of display cellular bandwidth available. Recent reports in the system [8], and used two types of interfaces for wireless ad cellular networks, especially for the integration. In this scheme, and use wireless high bandwidth channels in ad hoc mode (IEEE 802.11) for the transfer of a single-traffic transmissions from the cellular network (3G) cellular to improve productivity and the extent of coverage.

System in the third reported [9] [10], and the transfer of multicast data to a node relay-wide and short inside the

cellular network (3G) and is routed to the remainder of the decade Subscribe relay node via a dedicated high-speed networks (IEEE 802.11). In [11] System publishes dedicated relay devices to communicate traffic from one cell to an additional movement, to evade the difficulty of congestion due to unstable movement in the cellular system.

Been integrated heterogeneous wireless networks are applied widely and studied. Example of the application of integrated methodology is AMPS / IS-95 cellular network, and Global Positioning System (GPS) apply in the cellular network to afford services point, and satellite / cellular network [15,42]. There is also a rising significance in cellular and wireless LAN Network Integration (WLAN). Universal Mobile Telecommunications System (UMTS) [12,20,41], otherwise known as cellular network 3G, is capable to afford dissimilar services (voice and data services) on their individual. Conversely, because of the bandwidth limited radio, and the network cannot contain a huge number of users at on one occasion, particularly for application that necessitate fast data transport rate. Additionally the high cost of the examine. As a complement to the cellular network, WLAN might afford services with high data transport rate of a moderately low cost. Combination of these two heterogeneous network be able to make available superior service by having mobile user handoff back and forth among the networks to attain the essential services [33,34] and [17]. Conversely, WLAN has a extremely small radio coverage (particularly in urban areas), and can simply afford services to users is very secure to its fixed access point. To be capable to serve most users in such an included network, this will be posted on the high density of access points for wireless networks. This leads to increased cost and reduced fixed infrastructure efficiency. To overcome this obstacle, a dedicated network can be used instead of the WLAN. In / integrated cellular network dedicated and multi-hop ad-private links stretching almost radio coverage.

It also can access the mobile phone client outer surface the radio coverage of the service access position (fixed or mobile) throughout an intermediary routing. Cannot peer-to-peer service is accomplish directly throughout a dedicated network without going throughout the cellular network. In addition, you can use a custom to redirect traffic between the cells to get the budget in the cellular network channels. This further improves the cellular network's capacity. Research/ cellular network integrated custom can be found in [17,45], and [46]. This effort spotlight on how a enthusiastic network may improve cellular services. Perhaps called the ad hoc advance with the assist of cellular networks. Global Positioning System (GPS) [1,35] has been used extensively to serve GPS. Based on top of satellite signals established, you may point to conclude its position throughout a built-in GPS chip. Among the help of GPS, which is the foundation of MT might recognize where the objective is the MT and decision making suitable assistance. And called such a point with the assist of GPS assistance for positioning technique. Studying greedy perimeter stateless routing

(GPSR) in [12]. In GPSR, the subsequently stage of the path is constantly closest to the intention MT. And MT requirements to be acquainted with the exact position of the multilateral trading system and all other parties. Authors make the assumption that there is a server position. In [22], presumably on the MT source to find out the destination MT position. Not flooded routing requests in the network, but only directed towards the destination to reduce the routing overhead. In [19], and GPS guidance similar to those in [22] are the study protocol. Other works on the help of GPS guidance can be found in [26,28]. Management is considering the situation in detail in [7, 25]. In [25], distributed server model site is described. MT multilateral trading system and uses the other parties within a certain area as its servers. MT position will be sent to these site servers periodically. Can multilateral trading system and the other parties know this position MT through access to any of its servers. In another approach to manage the site [7], all MT has a virtual home region (VHR) with a fixed center. MT and update their locations by sending ads stand for VHR her. In all the newspapers, the multilateral trading system must know the approximate coverage customized for the network. There are over a relatively large presence of site updates. In an environment where GPS is not available, such as an internal office, information can be used to determine the relative positions by custom to select the routing network. It utilize an algorithm to conclude the self-sites to compute the comparative positions for MTS in [7], so that the synchronize system network can be built on the site in sequence. In an additional paper [43], the authors recommend routing performance founded on the link among proposed for users more willingly than particular sites.

This routing protocol is known as association beaconing routing (ABR) protocol. In the majority obtainable ad allocated routing protocols, there is a message "hello", which might assist to attain information as regards the MT neighbors. Though, it is complicated to association condition routing in ad hoc networks due to the dynamic topology and distribution of information slow. In previous work, there is no central server specific provision of global information, so it can not apply the directive sites based on the information the global position of minutes.

III. PROPOSED WORK

In this research, we concentrate the significant problem of performing routing by means of end-to-end statistical QoS assurance in heterogeneous MANETs. As of previous discussion, we examine that QoS routing in MANETs engage two key in sub problems: (1) Generate a stable and connected routing structural propose that is robust in opposition to MANETs frequent topology modify and link failure, and (2) performing arts QoS routing on the consequential virtual spine. We concentrate on each of these two key sub problems individually, while at the equivalent time provided that an integrated clarification for the QoS routing difficulty in MANETs.

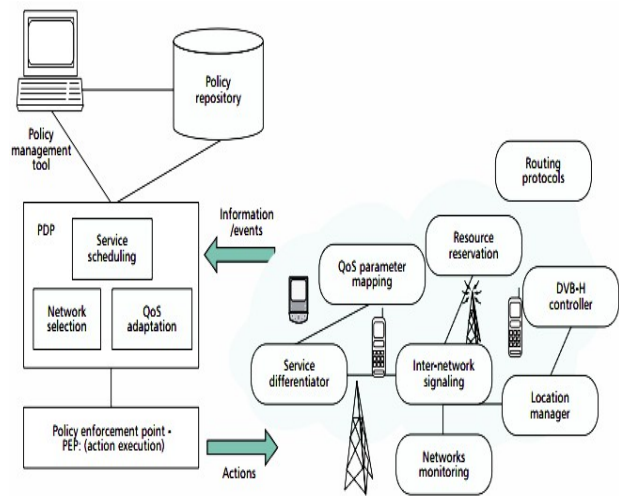


Fig. 2. System architecture of the Policy-based USG

USG is a compilation of the subsequent four types of entities:

- Policies: policies can be disseminated crosswise the UWD nodes depending on their implementation scopes;
- A policy assessment engine that motive over these strategy (known as the strategy assessment point or PDP in IETF expressions[5] to make decisions on network assortment and QoS adaptation;
- An accomplishment implementation engine (known as the strategy enforcement spot or PEP in IETF expressions) that afford the genuine execution of policy actions;
- Sequences of apparatus that execute the network procedure or pattern such as service discrimination, QoS parameter mapping, inter-network signaling, routing, and so on.

These apparatus are position across the UWD network. The adaptableness of this system structural design comes from its consciousness of the process surroundings (or context information).To makes an inclusive decision, the USG must reflect on the subsequent factors: service types, user necessities, user device individuality, and network status. These factors are all articulated by policies. Re-routing (including route repairing) or network reconfiguration transpire when there is a important modify in definite factors. This adjust is reported to the PDP throughout an occasion, and the PDP makes a appropriate decision by calculation above the policies. The way of thinking procedure might consist of variance resolution if the new background changes result in inappropriateness with user or system necessities. The events also can be generating by a instance phase circumstance or as a result of an action. Reinforcement the PEP is a variety of network administration and routing function. Those explicit to UWD service provisioning are in briefly converse as follows. For other common apparatus, pass on to the debate in [7].

Our involvement in this research paper is two-fold. Primary, we recognize that the majority of the routing problems in MANETs are partially due to the unbalanced topology of these networks. Consequently, having a predetermined, stable, and associated routing structural design (i.e. virtual wireless backbone) that is robust against node progress and recurrent link breakdown in MANETs can make simpler those routing tribulations. We examine provide such a constant topology with low transparency. For this principle, we develop an original zoning stratagem for heterogeneous MANETs. The zoning method maps the network objective topology on top of a virtual two-dimensional grid topology. The virtual grid consists of the occasional, but perhaps more commanding, mobile nodes called as Cluster Heads (CHs), which are designated sporadically. The proposed fasten effective topology provides stable routing backbone that is castoff to execute routing with the requisite end-to-end QoS assurance. Second, we proposition a QoS routing protocol for heterogeneous MANETs. The proposed protocol, called virtual grid architecture protocol (VGAP) makes use of cross-layer devise methodology with the intention of end-to-end QoS assurance. VGAP functions on the derivative predetermined virtual grid, where CHs ascertain various QoS routes by means of an comprehensive adaptation of the OSPF routing protocol, called Mobile OSPF (M-OSPF). To calculate QoS routes, M-OSPF utilize an comprehensive version of WFQ development policy, called Ad hoc WFQ (AWFQ). AWFQ takes into account the anecdotal time uniqueness of the wireless direct state into version, yet providing an higher vault on the arithmetical end-to-end delay guarantees of a definite flow. The prospect of using OSPF as a routing protocol in MANETs has been in current times discussed by the IETF MANET collection [24] but with no particular particulars.

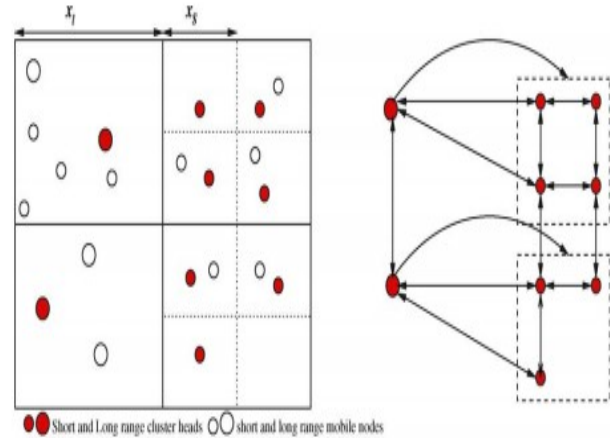
IV. NETWORK MODEL AND VIRTUAL NETWORK TOPOLOGY

We mull over a heterogeneous network that consists of N mobile nodes of dissimilar potential that are arbitrarily dispersed in the network area. To exploit location in sequence, we presume that each mobile node is operational with a global positioning system (GPS) card.

A. The zoning process

For the sake of effortlessness, but without loss of generalization, we presume that mobile nodes are of two dissimilar broadcast ranges: r_s for a short range (SR) nodes and r_l for long range (LR) nodes. The broadcast assortment in each case is a utility of the energy level at the mobile node. The network area is separated into fixed, disjoint, and standard shape zones (clusters). To generate a easy rectilinear virtual topology, we decide on the sectors to be square in form. A sector might have a combination of SR and LR nodes. At first, the network region is separated into large sectors where the sector side length x_1 is selected such that two LR mobile nodes in adjacent horizontal/vertical sectors and placed anywhere in the sector can correspond with each other straightforwardly. Consequently, x_1 is preferred as $x_1 = \eta / \sqrt{5}$. If the sector has only short range nodes, the sector is separated into subzones where each subsector side length (x_s) is calculated as $x_s = x_1/2$:

The numeral of layers in the fundamental topology is resolute by the number of broadcast series in the network. In Fig. 3 (b), the left side sectors have equally SR and LR nodes while the right side sectors have only SR nodes. Consequently, each of the right side zones is separated into four subzones to allocate straight communication among any two short range mobile nodes in adjoining horizontal/vertical subzones. As a consequence, the control visual projection is inadequate to the communication among CHs and in the linked vertical and horizontal directions only.



(a) Zoning with physical topology (b) Corresponding virtual topology

Fig. 3. Network zoning and the virtual topology

B. Cluster head election

Since mobile nodes be different in their potential, an eligibility criterion is considered necessary to establish which mobile node can develop into a CH. reminder that parameters like remaining battery power, node speed, and node position be supposed to count up in the direction of this decision. The CH role is revolved occasionally in the middle of nodes in each zone. The CH periodicity assists to equilibrium the nodes load allocation, attain fairness, and afford fault tolerance in opposition to single node failure. We describe the time flanked by two consecutive CH election instantaneous as the CH election interlude. each one mobile node make a decision whether it will become a CH for the present election period based on an eligibility factor (EF). The least mobile, most power proficient, most location suitable, and most lightly loaded node is the best applicant to act as CH. The eligibility of a node i to serve up as a CH at time t can be considered as follows:

$$EF_i(t) = a_1 e^{-v_i(t)} + a_2(1 - Q_i(t)) + a_3 B_i(t) + a_4(1 - s_i(t))$$

Wherever

$-v_i(t)$ is the standard speed of node i at time t.

$B_i(t)$ is the outstanding power in node i battery at time t,

$Q_i(t)$ is the tiny proportion of time node i remainder as CH for the duration of a time windowpane T (defined as a windowpane of time adequate to capture cluster dynamics) end at time t, $0 \leq Q_i(t) \leq 1$, and establish as follows

$$Q_i(t) = \frac{1}{W} \sum_{m=1}^W Y_i(t - m\delta t)$$

where $Y_i(t)$ is a binary indicator which is 1 if and only if Whenever $Y_i(t)$ is a binary pointer which is 1 if and only if node i was a CH at time t , and δt is a diminutive time percentage enhance such that $T=W\delta t$. $S_i(t)$ is the Euclidean detachment of node i regarding the midpoint of a zone at time t , and a_1, a_2, a_3, a_4 are scaling feature that replicate the significance of each parameter. Mobile nodes in each one zone will switch over the values of the intended EF's. The node that has the uppermost value of EF will decide on itself as the CH for the present election phase. That is, if a zone z has a nonempty set of n nodes, the CH is elected as pursue.

$$CH = \arg \max_{i \in n} (EF_i)$$

Each mobile node that has selected itself as a CH for the present phase will transmit a commercial message to situate the mobile nodes in its zone. If the CH foliage its zone or fails early, an early determination of the next CH obtain position after a handoff process in which the new CH succeed to the routing accumulation of the retreat CH. It is value declaring that the conception of CH determination is not a new conception. Conversely, our zoning method reduces to bare bones of its operation. Consequently, the routing utility will also be simplified.

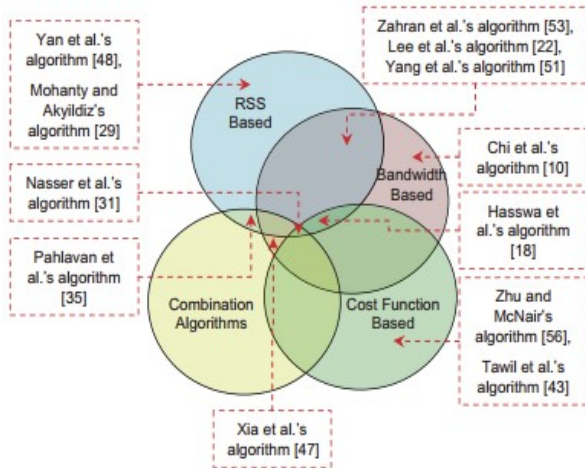


Fig. 4. The four categories of VHD algorithms and twelve selected representative schemes

C. Mathematical derivation of TBVH

In this section we begin by exploring the different scenarios for the time before a vertical handover (TBVH) component for upward vertical handovers, based on the MH's location and direction of movement. For the sake of simplicity, in this paper we only consider the UMTS-WLAN handover combination, although our model is easily applicable to other network combinations as well.

The first scenario to be considered is a MH that is roaming under the coverage of a BBS and is moving towards the boundary with velocity v as shown in Fig. 1. Here we consider

a circular cell of radius R . The inner dotted concentric circle shown in the above figure represents the handover threshold of radius r which is the distance from the BSS where a MH is expected to perform a handover. Location co-ordinates of the MH can be extracted as explained in references [3,11]. While x is the angle made by the BBS and direction of movement at the MH, the distance d between the BBS and MH and r can both be derived from the equation where the Received Signal Strength (RSS) is given by [14]:

$$RSS_{dB} = -10\gamma \log(l) \tag{1}$$

Where l is distance from the transmitter and γ is the propagation path-loss coefficient. The BBS passes the latest values of the required information to the MH which is responsible for performing the calculations. In order to find the TBVH in this scenario we need to calculate the distance z which is the point on the threshold circle where the MH is expected to vertically handover. As

$$r^2 = d^2 + z^2 - 2dz \cos x \tag{2}$$

The value of z calculated by solving the quadratic equation is

$$z = d \cos x \pm \sqrt{r^2 - d^2 \sin^2 x} \tag{3}$$

Therefore, the estimated TBVH for this scenario is:

$$TBVH = \frac{d \cos x + \sqrt{r^2 - d^2 \sin^2 x}}{v} \tag{4}$$

The parameter values in the above equation can be easily obtained, making it possible to calculate TBVH.

As WLANs may have specific points of exits such as doors in a building, the prediction accuracy of TBVH can be improved if the co-ordinates of these exit points are stored in the BBS and passed on the MH which then calculates the TBVH to the exit point.

In this case we adopt the concept of threshold distance TD [9] in the normal BS. This is a distance smaller than the cell's radius, which describes a smaller concentric circle located within the cell. The idea is that a MH moving inside the TD circle is more likely to change its direction, however on moving out of this circle, it is less likely to undergo a sudden change in its direction, thus enabling a correct prediction of the cell the MH is moving towards.

The goal in this case is to improve the prediction capability of the model by making it able to predict the TBVH of MH while it still moves in the coverage of A. As the MH is too far from the BBS to get a reasonably accurate value of b , we first need to find this distance and the angle β in order to calculate distance z .

$$c^2 = d^2 + b^2 - 2db \cos \theta$$

Therefore,

$$\theta = \cos^{-1}\left(\frac{c^2 - d^2 - b^2}{2db}\right) \quad (5)$$

Depending on which side of line AB point X lies,
Angle

$$\beta = |x - \theta| \quad (6)$$

Considering triangle BYC, we have

$$t = b \cos \beta \quad (7)$$

$$y = b \sin \beta \quad (8)$$

Therefore, in triangle BYX,

$$s = \sqrt{r^2 - b^2 \sin^2 \beta} \quad (9)$$

As $z = t + s$,

From (7), (8) and (9) we have

$$z = b \cos \beta + \sqrt{r^2 - b^2 \sin^2 \beta} \quad (10)$$

Thus the TBVH component for this scenario is,

$$TBVH = \frac{b \cos \beta + \sqrt{r^2 - b^2 \sin^2 \beta}}{v} \quad (11)$$

This is similar to the equation obtained in (4).

V. POSITIONING ROUTING IN CAMA

A. Centralized positioning routing

In CAMA, positioning routing is more reasonable because the CAMA representative may work as a centralized positioning in sequence attendant. An MT can locate its precise geographical point throughout GPS 1. The point in sequence is send to the CAMA representative throughout the BS. An MT's point can as well as establish by the cellular network by means of the current cellular position service. Separate from the positioning routing used in the wholesome ad hoc network, in CAMA routing, the present point of each MT can be fine known. An preliminary route from a resource to a objective can thus be indomitable moreover by CAMA agents or by MTs. If the routing is resolute by MTs, the BS will have to transmit the most efficient point in sequence. Founded on the conventional in sequence, each MT makes its hold routing verdict.

In this paper, CAMA agent is measured to be assembly the routing decision. Compared to MTs, the CAMA agent has further absolute global in sequence for the complete ad hoc network. This centralized routing method also transport advantages of routing optimizations, precautions, radio resource distribution and power savings. Furthermore, the centralized routing system doesn't require the episodic downlink dissemination of the positioning in sequence which generally devour large cellular radio bandwidth and MT power. Conversely, the centralized control has its

drawback. An MT might have to remain for a extensive time to obtain the routing decision from the CAMA agent if a lot of MTs send routing requirements at the similar instance. The delay is essentially caused by the retreat delay due to the uplink transmission conflict. In the case of a CAMA agent portion MTs in excess of one cell, the delay also comprise the uplink demand queuing delay and the downlink respond queuing delay. If the downlink delay is too elevated, the direction might lose its contemporariness. A new-fangled method has to be resolute by the simplified position in sequence. As of the standpoint of precautions, the centralized routing decision method might endure the attack of denial of service (DoS). In a actual wireless network, particularly in the inner-city area, two geographically secure MTs may not attain each other by the use of radio because of the composite radio proliferation environment (e.g., radio block). To develop the GPS routing exactness, MTs can send "hello" messages to their neighbors to confirm they are accessible to each other. This organization statistic is sent to the CAMA agent among MTs' particular positions, in order that the CAMA agent might know precisely what link exists, and make more exact routing assessments. This, conversely, enhance the overhead in mutually the cellular network as well as the ad hoc network. It must be illustrious that based on correlation between MTs, the CAMA agent might also make the routing decisions throughout link state routing techniques. The technique is not as instantly onward as GPS positioning routing and will be deliberate in our future work.

B. The procedure for making routing decisions

While an MT needs to send information to its objective, it will send a routing demand to the CAMA agent throughout the cellular radio channel. The channel know how to be the unsystematic access channel, the uplink regular packet channel, or else a pre-assigned traffic channel in UMTS. CSMA/CD how to be the unsystematic access method for the cellular uplink access. The MT will re-send the routing demand if it doesn't take delivery of the routing respond later than a time-out. The failure to receive a routing pronouncement is basis essentially by conflict with the concealed MTs. Conversely, the hiding workstation trouble here is not as severe as to facilitate in WLAN ever since the cellular radio reporting is huge enough measure up to the size of ad hoc network. The CAMA agent responds to the MT by means of a absolute route as well as every transitional MT throughout the frontward access channel, the downlink mutual channel, or else a pre-assigned traffic channel of the cellular network. Given that the point of all the MTs are recognized, the detachment for each hop is too recognized and the communication power of every MT can be predictable.

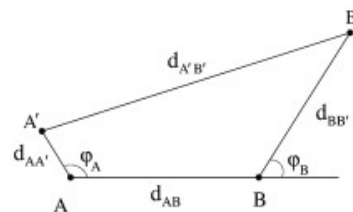


Fig. 5. An example of movements for ad hoc MTs

To supplementary accumulate power, MTs might “sleep” although listen to the cellular channel (e.g., the transmit channel or else the paging channel) every so often when they are not integrated in any active routes. When a original routing assessment is completed, the BS will page all the transitional MTs on the route by means of the objective MT by dissemination their IDs. These MTs will “awaken” to obtain and broadcast the data packets. In spite of everything the packages are received by the objective, the route will be free with all MTs on the route “sleep” over again. The routing in sequence is conceded in the description of each one data package, as is in DSR [8]. The midway MTs read the routing assessment to locate their next hop, as well as the suggested transmitting supremacy.

C. Position update

Location update is required when an MT progress on or after its earlier location. Intended for the GPS-aided positioning direction-finding, an MT has to launch its new-fangled position to the CAMA agent throughout the cellular channel. The new-fangled positions are modernized occasionally, by means of a time threshold value for the update phase. This value is founded on the specified prospect of wrong routing choice attributable to out-of-date position data being an end to than a value prt. It essentially depends on the network traffic, i.e., how frequently a original route has to be resolute and how frequently an MT is integrated in a new route.

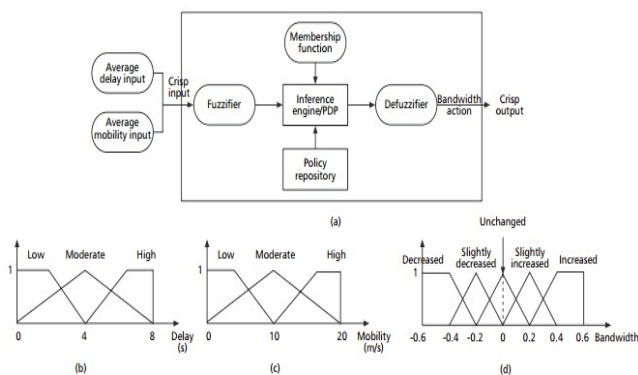


Fig. 6. The architecture of: a) the Fuzzy Control Module; and b-d) membership functions

Intended for the policy-based USG to endure the formless and indistinct data, fuzzy logic and fuzzy control [10] is make use of in group effort with policies and the PDP. The beginning of fuzzy control facilitate policies in the USG structure to utilize linguistic verbal communication to illustrate a network pattern tutoring, for model, “IF average delay is restrained AND standard mobility is low down, THEN bandwidth have to be improved.” Nevertheless, the delay assessment reported through the network is a hard value such as 500 ms, and the network can simply recognize hard values for bandwidth for example 1.5 Mb/s. For itself, there is a constraint for fuzzification and de-fuzzification as exemplify in the USG fuzzy control unit exposed in Fig. 5a. Intended for the present grade of our research, we choose three delegate network limitations to utilize for fuzzy control: delay, mobility, and bandwidth. The equivalent principle and

process are appropriate to further network parameters. The linguistic variables used to characterize the standard transmission delay of a data package are: low, moderate, and high. Those to characterize the standard node mobility as well as: low, moderate, and high. In the strategy point out earlier, both delay and mobility materialize in the provision part of the strategy. The fuzzy variables used to signify the strategy action, that is, bandwidth distribution, are separated into five levels: unchanged, a little increased, increased, a little decreased, and decreased. Intended for network pattern associated to delay, mobility, and bandwidth, the subsequent $3 \times 3 = 9$ policies or system, in the middle of several others, are necessary through the fuzzy control unit. Ever since triangular and trapezoidal formed membership task recommend more computational effortlessness than a Gaussian function [10], we select these to classify the membership task for the preceding three fuzzy variables worn in the policies. The realistic intend of these membership functions is fundamentally application definite. During our bandwidth allotment circumstances, we distinct the membership functions for delay, mobility, and bandwidth respectively as shown in Fig. 5b–d. The degree of membership is determined by plugging the monitored input parameter (e.g., delay or mobility) into the horizontal axis and projecting vertically to the upper boundary of the membership function(s). For example, if the input for delay is one second and the input for mobility is 9 m/s, then the membership functions show that the input linguistic value is low and moderate, respectively. The output value from the fuzzy control module activates the adaptation choice that is slightly decreased according to the corresponding linguistic policy. Then this fuzzy output response must be de-fuzzified by using the output membership function for bandwidth given in Fig. 5d. As a result, a crisp value for a decrease in bandwidth can be deducted, for example, decreased by 20 percent. An inference process is involved, especially for cases with more than one policy whose actions are satisfied. The MAX-MIN method [10], which tests the magnitude of each policy and selects the highest one, is adopted in our implementation. This method, though not combining the effects of all applicable policies, does produce a continuous output function and is easy to implement.

It is possible that when it is time for an MT to update its position, it remains close to the position in its previous update. A new position update is not necessary since there is no change in routing topology, and position update brings signaling and operating load to the cellular network. To determine whether a position update needs to be sent, another threshold value of the distance between an MT’s updated position and its position during last update should be defined. This threshold value is based on the requirement that the probability of a one-hop link break due to the non-updated position information should be no greater than prt. It should be adaptive in the networks with different MT mobility patterns. The threshold values can be estimated mathematically. To determine the time threshold value, we assume that λ is the mean of a Poisson packet arrival to each MT, and m is the average number of hops in each link. The time interval between any two cases in which an MT has to be active in a route is approximately negative exponentially distributed with a mean of $1/(\lambda \times m)$, so that the time threshold

value can be calculated by solving the equation $1 - e^{-\lambda_{mt}r} = P_{rt}$. For the distance threshold value, we assume that the original positions for two connected MTs are A and B. After a while these two MTs move to the new positions which are A' and B' respectively, as shown in figure 4. The new distance between these two MTs, $d_{A'B'}$, is:

$$d_{A'B'} = ((d_{BB'} \cos \varphi_A + d_{AB})^2 + (d_{BB'} \sin \varphi_B - d_{AA'} \sin \varphi_A)^2)^{1/2}$$

Assume d_{AB} , $d_{AA'}$, $d_{BB'}$, φ_A , φ_B are independent random variables with known distributions, the probability density function (PDF) for $d_{A'B'} \leq r$ under different distance threshold values can be numerically calculated, where r is the maximum ad hoc radio coverage. From the PDF function, we can find the threshold value d_{τ} for $d_{AA'}$ and $d_{BB'}$ so that $P \leq P_{rd}$. A numerical result of the percentage of a link-break against different distance threshold values is shown in figure 5.

Special updates may be needed when the radio environment for an MT changes significantly (e.g., when an MT turns a corner or goes into a building). These changes can only be measured by sending "hello" messages between MTs for reachable neighbors.

VI. METHODOLOGY

A. The First Step: Quick Evaluation Method for the Pre-Handoff Decision

Without a doubt, unusual classes of services necessitate different mixture of vertical handoff limitations (e.g., reliability, latency, and data rate). Consequently, user's traffic module (i.e., classes of service, service types) must be well thought-out in the handoff decision. Intended for the traffic classes, we track the four QoS classes of network applications distinct by UMTS [21]. They are informal class, streaming, interactive class and background class. For instance, according to the delay compassion distinctiveness, the primary two types are clustered as instantaneous service, Whereas the further two belong to non-real instant service. Given that real-time service is susceptible to delay, a assured broadcast rate is necessary. Certainly, the separation of traffic classes can be ended by users.

In vertical handoffs, several network limitations encompass an effect on whether otherwise not a handoff be supposed to take place. The significant parameters consist of quality of service (e.g., handoff delay, accessible bandwidth), security, power supplies, cost of service, and so forth.

During the primary step, the pre-handoff verdict estimates whether the bare minimum guarantee of a abuser is sustained for each network i, $i = 1, 2, \dots, N$. Further exclusively, the values of a little simple detected and critical parameters ought to be additional than the predefined thresholds, in that order.

$$M_i = F(b_i - b_{th}) \cdot F(RSS_i - RSS_{th}) \cdot F(V_i - V_{th}) \\ \times F(t_i - T_{th}) \cdot F(P_i - P_{th}) \cdot F(C_i - C_{th})$$

Equation (6) correspond to a minimum guarantee function, which point out the minimum guarantee of MN is maintain intended for each network i. It is aimed at assembly utilize few of the limitations referred in Eq. (6) with the intention of make a earlier and wiser handoff decision. At this point $b_i, RSS_i, V_i, T_i, P_i$ and C_i characterize the standards of obtainable bandwidth, received signal strength (RSS), velocity, duration, battery power and monetary cost of MN as of a exacting network i. The interval T_i represent the predictable point MN will continue in a specific network i, which is calculated by some comprehensive parameters. Into the bargain, $b_{th}, RSS_{th}, V_{th}, T_{th}, P_{th}$ and C_{th} be the predefined thresholds of obtainable bandwidth, acknowledged signal strength, velocity, duration, battery power and pecuniary cost to maintain the appealed traffic class, The demanded traffic class repeatedly is the major traffic class or the majority of important traffic category for MN, which takes up for the most part the whole time. The utility $F(.)$ is a unit step function. The unit step function is a irregular function whose significance is nothing for negative dispute and one for positive dispute. For this reason, previously there is not less than one parameter value of MN since a specific network i is lower than its threshold, the minimum guarantee function have nil value. In this casing, network i is not measured as a objective network to any further extent. If the minimum guarantee function value is individual for a specific network i, this network will be additional to the applicant network set S. Make a note of that the set S is set to be blank at the establishment of every handoff decision. Ever since Eq. (6) is effortless and the limitations as of this equation can be predictable rapidly, time utilization of the pre-handoff decision is extremely low. Certainly, various parameters can promote be misplaced according to the circumstance of exact relevance (e.g., the source of MN, the accessibility of a few parameters referred above). For instance, in a few circumstances, Ever since the interval of MN can't be predictable for a few reasons, this parameter has to be misplaced. Clearly, by establishing the interval into the minimum guarantee function, the severe ping pong effect can be proficiently removed. Thus, our advance condenses gratuitous handoffs while ever-increasing network throughput, lessening handoff delay, and keep away from connection dropping.

In addition, the velocity should be considered in the first step. For example, if MN's speed is over 100 km/h, WLAN cannot support its speed. Besides, battery power may be a crucial parameter for certain users. For example, when the battery level is low, the user may choose to switch to a network with lower power requirements (i.e., the threshold of battery power), such as an ad hoc Bluetooth network. After the pre-handoff decision is finished, according to the size of the candidate network set S, the propose decision algorithm generally falls into three cases. One is that the set S is empty, MN remains connected to the current network. Another is that there is only one member in set S. If the sole network is the current network, MN stays in the current network; otherwise, MN decides to perform vertical handoff procedure to be associated with the network. The other is that more than one network have been added into the set S. More specifically, there is more than one network, whose minimum guarantee

function value is one. If MN is a resource poor node and the current network is in the set S, MN will remain connected to the current network. If MN is a resource poor node and the current network is not in the set S, MN will handoff to any network included in the set S. If MN is a resource rich node, MN will proceed to the second step.

B. The Second Step: Vertical Handoff Decision Function

In this secondary step, an extended vertical handoff decision function (EVHDF) is existing, which is an extensive description of VHDF in [1]. EVHDF is worn to evaluate the development expanded by handing off to an exacting network j incorporated in the contender network set S. At this point we presume the dimension of the candidate network set S is m. It is noticeable that m is an numeral greater than one. According to Eq. (5), it can be seen with the intention of the calculation of DNCBP doesn't simply narrate to accessible bandwidth, but also network traffic load. Additionally, the DNCBP can be worn to point out network traffic load. Consequently, utilize the DNCBP in the EVHDF can be supplementary practical for network load balancing crosswise diverse networks than supplementary parameters (e.g., obtainable bandwidth). The network by means of the uppermost intended value for EVHDF is the majority optimal for MN based on particular predilection. The EVHDF for a meticulous network j, EQ_j is defined by:

$$EQ_j = \frac{\omega(1/c_j)}{\max((1/c_1), \dots, (1/C_m))} + \frac{\omega_s S_j}{\max(S_1, \dots, S_m)} + \frac{\omega_p P_j}{\max(P_1, \dots, P_m)} + \frac{\omega_D D_j}{\max(D_1, \dots, D_m)} + \frac{\omega_F F_j}{\max(F_1, \dots, F_m)}$$

At the same time as illustrate in [1], at this point ω_C , ω_S , ω_P , ω_D and ω_F are weights for every of the network parameters. The principles of these weights are tiny proportion (i.e., they range from 0 to 1). Furthermore, every part of five weights add together to 1. every weight is comparative to the consequence of a limitation to the vertical handoff algorithm. In adding together, C, S, P, D, F there the cost of service, security, power expenditure, network circumstances, network recital, respectively. The major differentiation among EVHDF at this point and that in [1] is that $D_j = b_j$ in [1] has been substituted with

$$D_j = \frac{b_j}{H_j}$$

The network by means of the uppermost EQ_j is the elected network. If the elected network is not the present network, vertical handoff takes places; or else, MN remainder associated to the present network.

VII. SIMULATION RESULTS

We simulate an ad hoc network by means of 100 MTs reside in an region of 1000 m × 1000 m. Every MT move about surrounded by the area, with a unsystematic direction and a unsystematic velocity consistently distributed between 0 and a greatest value. Without any measurement, this greatest value is 3 m/s, the momentum for ordinary users. The ad hoc direct has a predetermined data rate of 1 Mb/s.

The wireless crossing point workings like the 914 MHz semi-transparent Wave LAN, with a ostensible radio range of 250 m. MSGPR (multi-selection greedy positioning routing) underneath CAMA background is compared with two other ad hoc routing protocols, AODV and DSR. The penetrating set for MSGPR is set great adequate so that the preeminent route can always be establish. We presume that spot updates and routing requirements can constantly be sent effectively to the CAMA agent at their first effort. In this paper, the case that MTs send their links by means of adjoining MTs is not integrated.

TABLE I. SUMMARY FEATURES OF NP, QOS AND QOE

Quality of Experience	Quality of Service	Network Performance
User behaviour attribute	Service attribute	Connection/Flow element attribute
Focus on user expected effects	Focus on user observable effects	Focus on planning, development (design) operations and maintenance
User subject	Between (at) service Access points	End to End or network elements capabilities

A. Delivery ratio and cellular overhead

The delivery ratio and the equivalent routing visual projection for MSGPR, AODV, and DSR are exposed in figures 6 and 7. The routing overhead for MSGPR contains the routing requirements, routing reply, and point updates obtainable throughout the cellular radio channel. It is exposed that MSGPR has a much improved delivery ratio than AODV and DSR. The routing overhead in MSGPR is too much lesser. When the amount of active links enhance, the delivery ratio meant for MSGPR reduces, as is the case with AODV and DSR. This is appropriate to the improved conflict in the MAC coating. The overhead for MSGPR enhances slightly when the number of active links enhances because of the rising number of routing requirements and replies.

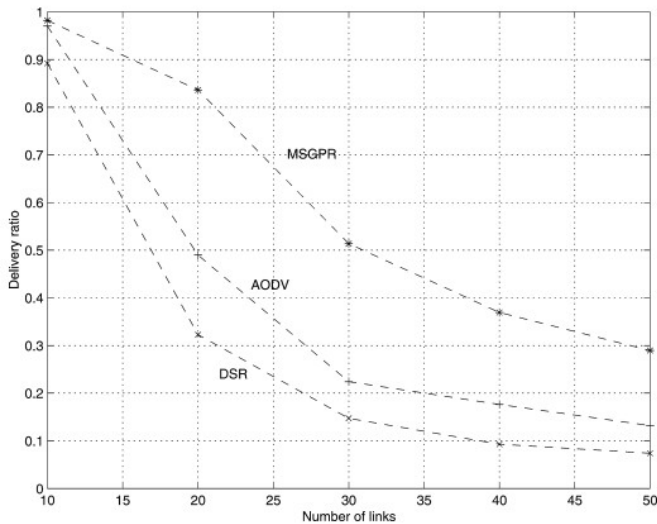


Fig. 7. Delivery ratio comparison among MSGPR, AODV, and DSR

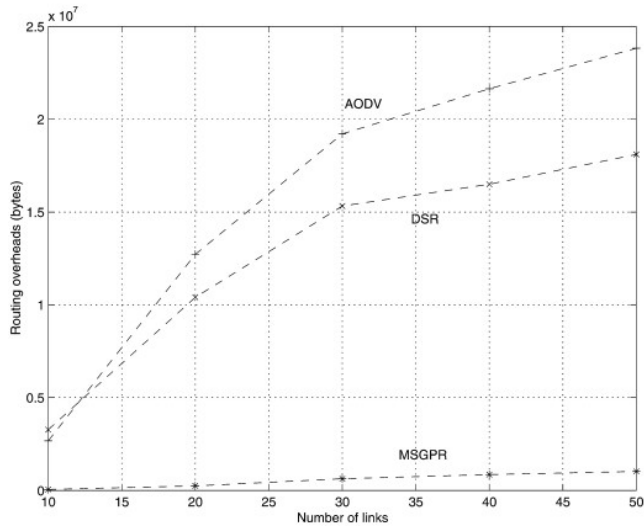


Fig. 8. Routing overhead comparison among MSGPR, AODV, and DSR

The routing visual projection within MSGPR is furthermore a cost in the cellular network. The expand by means of MSGPR over the AODV and DSR is exposed in figure 8. The expand in the ad hoc networks (the number of supplementary bytes distributed effectively than would be in AODV and DSR) is just about 10 times as great as the cellular overhead at the middling network load and high network load. Intended for networks with short load, the expand is level bigger. Intended for commercial wireless services, it is significance by means of CAMA if a byte in a cellular network is down with than 10 times the assessment of a byte in an ad hoc network.

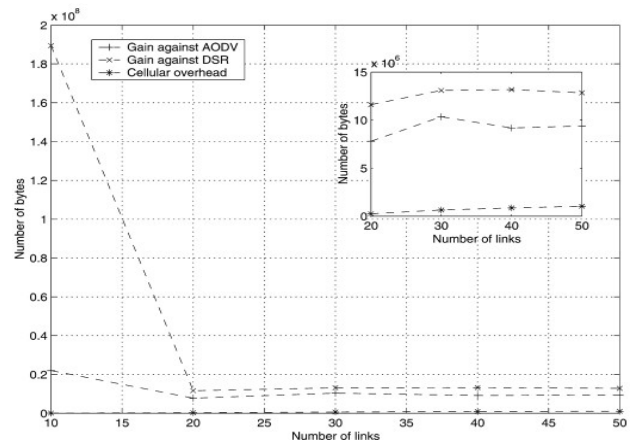


Fig. 9. Gains when using CAMA

B. Packet delivery ratio

We describe the packet deliverance ratio as the ratio among the number of packages established by the intention and the number of packages produced by the relevance layer sources of the conventional calls. Packet delivery ratio is significant as it illustrate the loss rate that will be observed by the transport protocols, which in revolve distress the maximum throughput that the network can sustain.

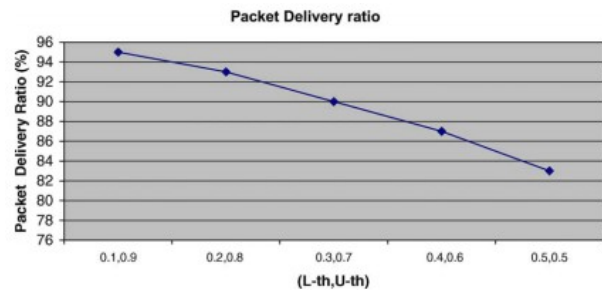


Fig. 10. Packet delivery ratio versus the link state update thresholds

The figure displays the integer of packets delivered effectively each second through the simulation phase. The probable successful deliverance rate of the sent package is five packets/ s. As of the throughput curvature, we can observe that network was capable to meet the potential at generally. From the figure, we capable also recognize or expect the packet loss in the network. Permanence of VGAP structural design (the virtual grid) permit it to accomplish elevated all through. Fig. 9 displays the packet delivery proportion by means of M-OSPF in VGAP and employing the link state announcement model that uses two threshold point. The thresholds (Lth, Uth) are distorted as of (0.1,0.9) to (0.5,0.5), correspondingly. VGAP is capable to attain high packet delivery ratio and continue good enough levels when the link updates enhance (enhance in control traffic).

C. End to End Delay

The end-to-end packet delay is also studied. When a link is undependable, the node fails to promote packets, basis packet plunges or longer delays. On low traffic load, nodes infrequently experience obstruction but frequently knowledge broken links.

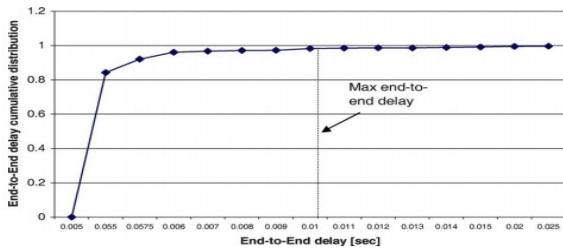


Fig. 11. Cumulative distribution of end-to-end packet delay

Consequently, packets that require to be re-routed will be queued and as a result encounter longer delays. The packet one after the other delay was situate to 10 ms. VGAP was capable to convince the delay necessities for mainly of these packets as shown in Fig. 10. The outline shows the increasing prospect allocation of the continuous delay of the engender packets in the network. We observe that the majority packets were delivered effectively within their required delay bounds.

VIII. DISCUSSION AND CONCLUSION

The convergence of more than two wireless access networks, though more complex, has a better chance of guaranteeing the QoS of users' services and offers better optimization to overall network resource utilization. A flexible (not rigid) management mechanism, as evidenced in this article, plays a critical role in the success of such a complex network. The proposed protocol makes use of the communication of the underneath three layers of MANETs to develop the network concert and sustain of quality of service requirements. The protocol superimpose a usual rectilinear virtual topology on the substantial topology using a effortless zoning process that takes into deliberation the transmission range differentiation in a heterogeneous MANETs. Our simulation outcomes illustrate enhanced concert in terms of bandwidth and uninterrupted delay guarantees and further network toughness to link failure and topology differences. The protocol as well utilizes a power manage algorithm at the physical layer to afford communication among nodes and sustain network connectivity in this heterogeneous setting.

IX. FUTURE WORKS

In future work we plan to extend the recovery model by using a hierarchical architecture with dedicated proxies in the assistant recovery network. Furthermore, we will investigate possible security implications that are introduced in the network due to the heterogeneity of two different networks. Finally we plan to implement the proposed protocol using socket programming in order to study its efficiency in a real environment.

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Design and Simulation of a Low-Voltage Low-Offset Operational Amplifier

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Abstract—In many applications, offset of the OP-AMPs should be canceled to high accuracy be accomplished. In this work, an asymmetrical differential input circuit with active DC offset rejection circuit was implemented to minimize the systematic offset of the amplifier. The proposed OP-AMPs show that the systematic offset voltages is less than $80 \mu\text{V}$.

Keywords—component; formatting; style; styling; insert

I. INTRODUCTION

The CMOS Op-Amp is an important building block of linear and switched-capacitor circuits. However, mismatch of the devices causes an offset voltage, which limits the high-precision application. Commonly, the offset sources of OP-AMPs are categorized as systematic offset and random offset. The systematic offset happens because of the channel length modulation of transistors and the value of the offset voltages are the functions of the input and output common mode voltages [1, 2, 3]. For example, in the formal OP-AMPs, the channel length modulation of transistor shows the systematic offset. However, the channel length modulation is unimportant, with no feedback, the output common mode voltage ever shows the fixed voltage level and doesn't follow the change of the input common mode voltage level. This difference between the input and output common mode level shows very small systematic offset voltages. The systematic offset can be minimized by controlling bias current of input stage to sustain the input and output common mode in same level.

The general method of offset cancelation of OP-AMPs is the feedback-capacitor circuit as shown in Fig.1 [7]. At first, as the switch 1 and 2 are turned on, the offset voltage is stored in C offset. Then the offset of V_{OUT} is omitted when the switch 1 and 2 are turned off and the switch 3 and 4 are turned on. But this circuit has some disadvantages of large capacitor, and many CMOS switches which is the source of the switching error. In this work, a continuous time asymmetrical differential input circuit with common mode feedback circuit which can minimize the offset of OP-AMPs is presented.

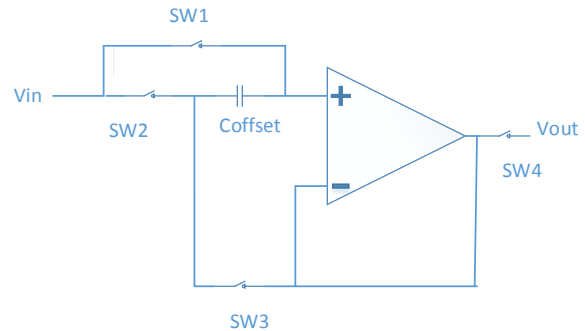


Fig. 1. Typical Offset Compensation Circuit

II. CIRCUIT DESCRIPTION

The proposed OP-AMP is composed of three parts: Asymmetrical differential input stage, Common Mode Feedback (CMFB) stage and output stage. In input stage, there are cross-coupled input devices, M_1 , M_2 , M_3 and M_4 and tail current sources, M_5 and M_6 . Input devices have asymmetrical differential structure. It means that W/L of M_1 and M_3 is larger than that of M_2 and M_4 , so the transconductance (g_m) of M_1 and M_3 is larger than that of M_2 and M_4 . The proposed CMFB circuit is shown in figure 3. The first stage is the combination of M_9 - M_{12} and current sources M_{14} - M_{17} [10, 8].

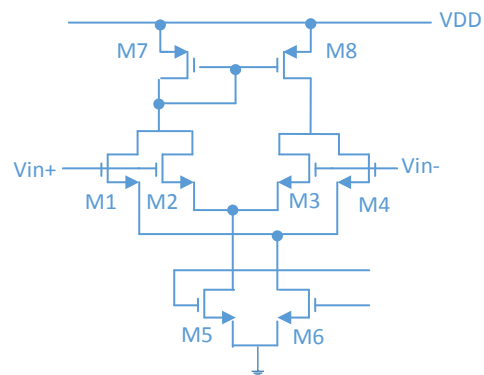


Fig. 2. Input Stage

The second stage is the combination of M_{14} - M_{19} . Long channel NMOS transistors are used for input stage to minimize the differential pair nonlinearity and to insert more input voltage swing. They also minimize the V_{bias} caused by the transistor mismatch among M_9 - M_{12} . The common-mode level of the input and the output could be detected and amplified by the DC offset rejection circuit, and changed to the feedback signal for current sinks of the amplifier. This is a negative feedback network.

By adjusting the current of the current sinks, the input and the output common-mode voltage would be maintained in same level to minimize the systematic offset. The difference between input and output common-mode level will be amplified. In the proposed circuit, the sum of drain currents, I_{M5} and I_{M6} is constant. Therefore, small signal differential mode open loop voltage gain of input stage is given by:

$$A_{vd,input-stage} = -(g_m(M3) + g_m(M4)) \cdot (r_o(M8) \parallel r_o(M3) \parallel r_o(M4))$$

Where g_m is the transconductance and r_o is the output resistance of transistor. The total small signal open loop differential mode voltage gains are given by:

$$A_{vd} = (A_{vd,CMFB-stage}) \cdot (A_{vd,input-stage}) \cdot (A_{vd,output-stage})$$

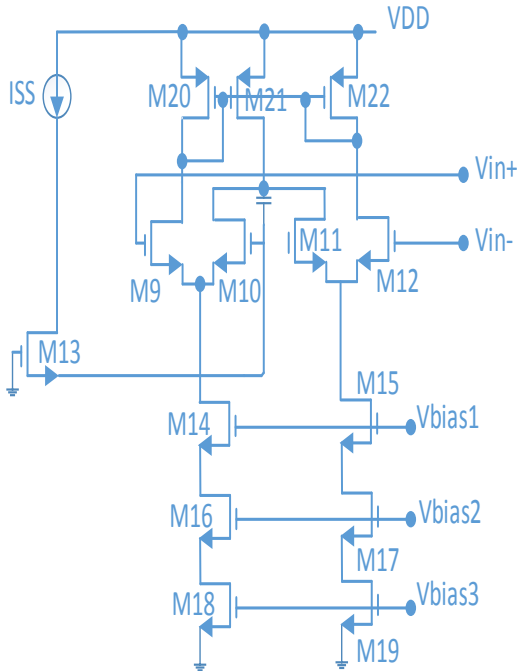


Fig. 3. DC Offset Rejection Circuit

III. OUTPUT STAGE

The class-AB output stage shown in Figure 4, is composed of an output buffer. Transistor M_{23} - M_{26} form two floating current sources to provide bias current of branches, which confirm the transistor M_{33} , M_{34} , M_{35} and M_{36} work in saturation region. Their gate voltages are provided by two biasing branches respectively. The current signal I_{IN1} and I_{IN2} are subtracted through current mirror and amplified by push-pull stage [10, 2].

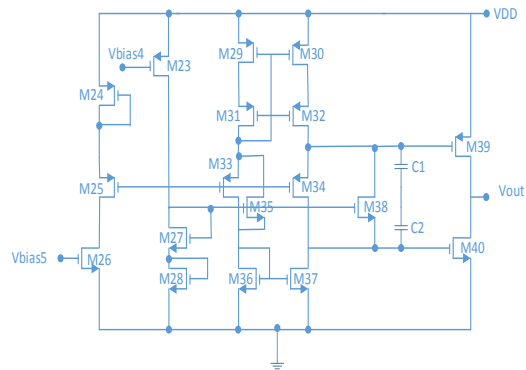


Fig. 4. Output Circuit

IV. SIMULATION CIRCUIT

Figure 5 shows the AC responses of the proposed OP-AMP while driving a 2 pF load. It shows 60 dB open-loop gain, 63.5° phase margin, and 2.82 MHz unity gain bandwidth. DC sweep analysis of the amplifier connected in an inverting unity-gain configuration is shown in figure 6. The simulation results showed good following characteristic between V_{in} and V_{out} , and the offset voltage less than 80 μ V by averaging.

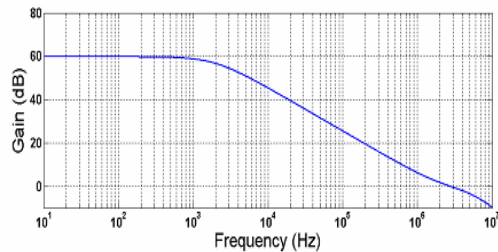


Fig. 5(a)

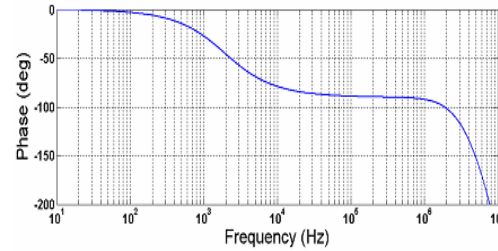


Fig. 5(b)

Fig. 5. Simulation Result of Frequency Response: (a) Magnitude, (b) Phase

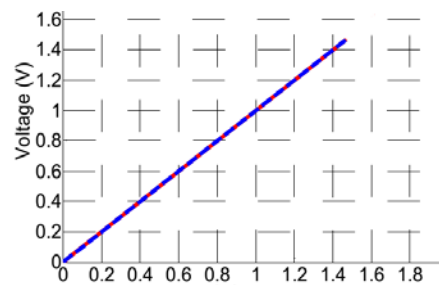


Fig. 6. The Simulation Result of the Following Characteristic

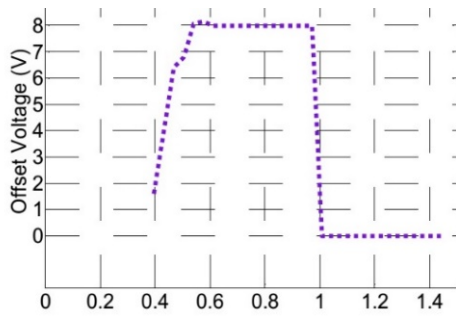


Fig. 7. The Simulation Result of the Offset Tuning Range

TABLE I. PERFORMANCE SUMMARY

CMOS Technology	0.18 μ m
Supply	1.5V
Gain	60dB
Phase Margin	62°
Unity Gain Bandwidth	2.73 MHz
Input Offset Voltage	80 μ V

V. CONCLUSION

An offset cancellation technique that uses an asymmetrical differential input circuit with active DC offset rejection circuit has been presented. Simulation results show that Op-Amp offset voltage is less than 80 μ V in entire operating voltage range.

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Enhanced Audio LSB Steganography for Secure Communication

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Abstract—The ease with which data can be remitted across the globe via Internet has made it an obvious (as medium) choice for on line data transmission and communication. This salient trait, however, is constraint with akin issues of privacy, veracity of the information being exchanged over it, and legitimacy of its sender together with its availability when needed. Although cryptography is being used to confront confidentiality concern yet for many is slightly limited in scope because of discernibility of encrypted information. Further, due to restrictions imposed on the use of cryptography by its citizens for personal doings, various Governments have also coxswained the research arena to explore another discipline of information hiding called steganography – whose sole purpose is to make the information being exchanged inaudible. This research is focused on evolution of model based secure LSB Steganographic scheme for digital audio wave file format to withstand passive attack by Warden Wendy.

Keywords—Conceal; Human Auditory System (HAS); Imperceptible Communication; Internet as a Secure Communication Medium; LSB Based Audio Steganography; Modeling Security of Steganographic System; WAV File Steganography

I. INTRODUCTION

Steganography written as $\sigma\tau\epsilon\gamma\alpha\nu\acute{o}\varsigma$ $\gamma\rho\alpha\phi\acute{\eta}$ means “Hidden/Covered/Concealed Writing”, has its roots in Greek [1] and is not something new as it dates back to 440 B.C. [2]. Digital steganography came into glare of publicity when [3] reported its unlawful usage. 9/11 event indirectly contributed towards a serious drift of academics towards exploration of this field where a blend of encryption and steganographic techniques has emerged as a recommended solution [4] to achieve two fold security.

A. Terminology used in Steganography

Steganography works by selecting some carrier/container e.g., image, audio or video etc. called cover together with information to be hidden inside the cover referred to as message. The process of hiding is known as embedding and that of information retrieval is termed extraction reigned by a share secret called stego key between the communicating parties. The outcome of bit embedding is known as stego object while the entire activity at sending and receiving end constitutes a complete steganographic system [5].

B. Terminology used in Steganography

Steganography works by selecting some carrier/container e.g., image, audio or video files etc. [6] describes three types of steganography as follows:

- **Pure Key Steganography:** The oldest form of Steganography employing schemes without any Stego key.
- **Secret/Symmetric Key Steganography:** The Steganographic schemes make use of pre-agreed shared key that reign in the algorithm.
- **Public/Asymmetric Key Steganography:** It makes use of Public and Private Stego Key pair on the analogy of Asymmetric Cryptography.

C. Model

Simmon [7] was first to apprehend steganographic model widely known as ‘Prisoner’s Problem’ where Alice and Bob held in imprisonment had to plan their scape on a pre-agreed protocol. Warden Wendy was there to monitor the message exchange between the two and could seize further communication in case of suspicion.

Type of constraints that Wendy may meet while examining communication between Alice and Bob include:

- **False Positive:** Wendy detects a cover with hidden message which in fact does not carry one.
- **False Negative:** Wendy lets go a cover that does carry a hidden message.

As ostensible, steganographic schemes must strive to force Wendy to commit ‘False Negative’ type of error.

D. Attacks on Steganographic System

Simmon [7] was first to apprehend steganographic model widely known Active and Passive type of attacks have been discussed in [8] where the former attempts to repeal the hidden information while the later is intended towards extracting the hidden contents without destroying original contents. However, [9] has discussed five type of attacks depicted in Fig. 1 followed by brief explanation of each.

- **Chosen Message attack.** This type of attack aims at analysing the stego object obtained from an input (message) of his / her desire i.e., to study the effect of embedding of known message on the cover.
- **Chosen Stego attack.** Here the attacker with known algorithm tries to extract hidden information from stego object of his/her choice.
- **Known Cover attack.** With original cover and the resultant stego object the attacker tries to contrast the

difference between the two to arrive at the hidden information.

- **Known Message attack.** Here the attacker tries to unfold the embedding algorithm while in possession of the actual message and the stego object.
- **Stego only attack.** The attacker tries to demystify the embedding algorithm in order to extract hidden information by having the Stego Object alone.

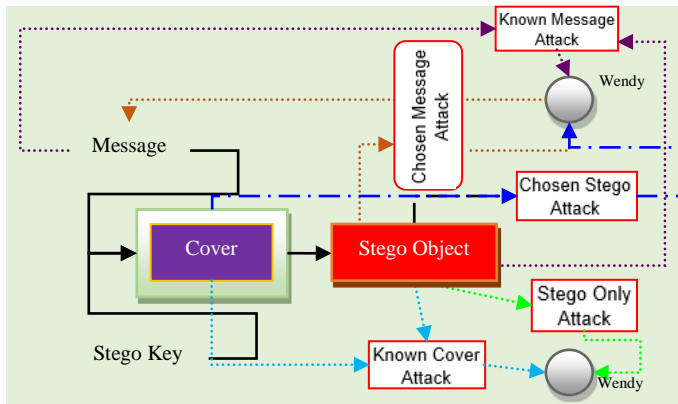


Fig. 1. Five Types of Attack on a Steganographic System

E. Perfect Security for Steganographic System

Kerckhoff's [10] opined that security of an algorithm in public domain resides in its key. In view of [1] "in a 'perfect' system, a normal cover should not be distinguishable from a stego-object, neither by a human nor by a computer looking for statistical patterns." Author in [11] discussed perfect security in context of following scenario: Amy creates a cover and embeds a random arrangement of secret message inside it under control of stego key. The stego object then reaches Bert through Crystal without suspicion. Bert may then retrieve the hidden message using the same (pre-agreed) stego key. Cachin [12] came up with equation (1) as a measure to estimate perfect security and asserted $\epsilon = 0$ as a requisite condition to achieve perfect security.

$$\epsilon \leftarrow D(P_c || P_s) \leftarrow \sum_{q \in Q} P_c(q) \log_2 \left(\frac{P_c(q)}{P_s(q)} \right) \dots \dots (1)$$

P_c & P_s signifies probability distribution of cover and stego object while Q is any finite set of alphabets.

[13] and [14] have already expounded at length on aforesaid discussion and in view of that we settle for imperceptibility and adherence to Kerckhoff's Principle as foremost bindings on any Steganographic scheme.

F. Audio Files and Formats

Pulse Code Modulation (PCM) is a technique for converting analog signals to digital format whose detailed discussion is beyond the scope of this paper, however, steps involved are depicted in Fig. 2 espoused from [15].

In general audio files are either flattened or uncompressed. Flattened audio may further be segregated into lossless (content preserving), and lossy that offers compact file size by persisting only prominent signals and includes MP3, AAC etc.

as shown in Fig. 3. Uncompressed PCM is referenced by its sample rate (that confines the peak frequency relevant to specific file format) and bit-depth (a measure of noise within a signal). For example: Uncompressed CD audio takes 44,100 samples per second where each sample is banked as a 16-bit number. Uncompressed "WAV" file format is analogous to CD audio and is the focal file format for this exertion. For details on WAV file format [16] serves as a good reference.

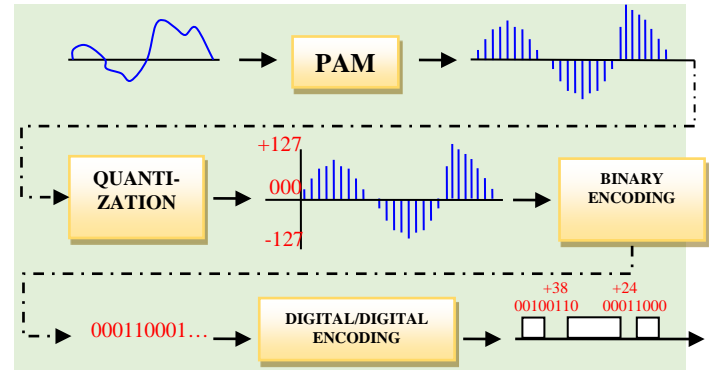


Fig. 2. Pulse Code Modulation (PCM). Analog-to-Digital Conversion

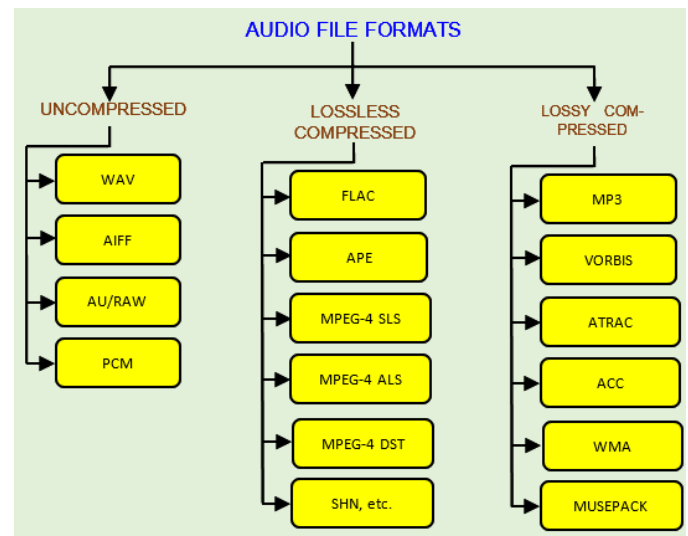


Fig. 3. Audio File Formats

By no means is audio steganography belittled in contrast to its counter parts like images and video etc. because according to [17] "...the human auditory system (HAS) operates over a wide dynamic range. The HAS perceives over a range of power greater than one billion to one and a range of frequencies greater than one thousand to one." This peculiar feature of HAS tends to undermine lower sound over louder ones and can be exploited either by acoustic or psychoacoustic models. Fig. 4 derived from [18] stretches a generalized view for data hiding inside audio files of which temporal domain (low-bit embedding) is the prima facie (prīmā faciē) of our research endeavor.

G. Evaluation Parameters

In [19] the impelling factors for audio quality includes (but are not limited only to) capacity of cover to hold secret data, and its imperceptibility of stego object in terms of Perceptual

Evaluation of Speech Quality (PESQ) computed via equation (2) whereas robustness against deliberate amputation of hidden data remains undisputed perilous trait of any auditory steganographic system.

$$SNR_{db} = 10 \log_{10} \left(\frac{\sigma_{signal}^2}{\sigma_{noise}^2} \right) \dots \dots (2)$$

where $\sigma_{signal}^2 = \frac{1}{N} \sum_{i=1}^N Signal_i^2$ is the variance and is directly linked to noise power.

Lateral discussion in said reference include criteria specifically relating to active attacks (if) initiated by Warden Wendy mentioned below only for reference:

- Amplification
- Filtering
- Re-quantization
- Re-sampling
- Noise addition
- Encoding/Decoding
- Transcoding

H. Low-bit Encoding

The simplest and easiest way to hide information is the Least Significant Bit (LSB) embedding where message bit gets substituted in place of LSB of cover audio. The advantage lies in high embedding capacity whereas the disadvantage is its vulnerability towards noise insertion, amplification, and filtration according to [17].

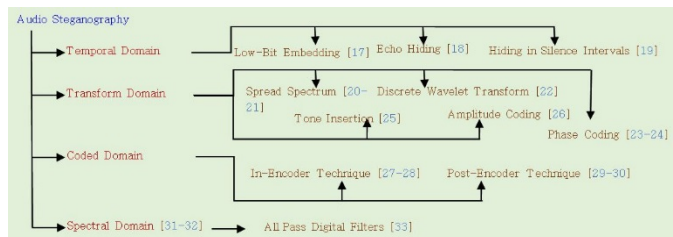


Fig. 4. Taxonomy of Data Hiding in Audio File

I. Paper Plot

Rest of the paper is organized as follows: Section II covers literature review of some of the most recent research on the subject followed by elaboration on our targeted LSB based steganographic scheme together with its limitations. Section III discusses modus operandi beginning with our statement of purpose followed by how we plan to achieve it (proposed logic/algorithm). Test results are shown in Section IV whereas Section V expound on theoretical aspect of proposed enhancement. Pros and cons come in Section VI. Section VII suggest future work while Section VIII concludes the discussion.

II. LITERATURE REVIEW

Following is a brief discussion on recent research focused on LSB steganography for audio files:

Authors in [20] have incorrectly stated SHA-1 as a PKE encryption algorithm. Their assertion “After receiving secret message’s binary file salt and cover audio file salt; public key encryption algorithm i.e. SHA-1 is applied to get the encrypted file. Finally, this encrypted data is hidden in cover audio.” renders their proposed algorithm as nothing more than a fabricated piece of work. Authors in [21] have used the basic LSB technique to hide encrypted data without deliberating on the encryption algorithm used. In [22] authors made use of Modulus 16 for selecting target locations within audio file to embed four bit chunks of secret information. Authors in [23] have come up with two steganographic schemes. The first scheme “...instead of directly replacing LSBs of digitized samples with the message bits, it first checks the parity of the samples and then carries out data embedding.” While the second flips/left un-changed the LSBs of the sample such that it’s XOR with 2nd LSB points to secret message bit. [24] appears identical to earlier work of [25] in its approach towards hiding information inside audio carrier grounded on Genetic Algorithm based LSB embedding except that it first encrypts the message and then performs bit embedding in high order bits of targeted samples. The core idea is to select a random bit point in original audio sample, replace that bit with secret message bit and alter the remaining bits of the sample for close approximation with that of the original sample. Author in [26] which is also our targeted scheme for enhancement simply used two files and termed those as “two intermediates” to transmit the hidden secret thereby asserting on its security. The first of the two files is the stego object that carries hidden data at three LSB positions of sampled data arranged in 2D mapping while the other conveys corresponding locations of the chunks carrying the hidden data via meaningful English sentences arrived at through the use of context free grammar (CFG). Noticeable limitations of targeted scheme include the following:

- Not in accordance with the definition of steganography.
- Not in accordance with Kerckhoff’s Principle.
- For the system to operate correctly the stego object and the file holding coordinates corresponding to audio samples containing hidden information must be handy at any given time – i.e., in the absence of any of the two referred files the system may not be available for use. Further, there must also exist some logical connectivity between the two files or else will complicate the extraction process (in case of mismatched files) that in turn poses a security risk.
- The contribution is silent over handling of repeated occurrences of same sequences (if any) of the randomly generated coordinates.
- By default, the cover need to be kept secret/destroyed and may not be reproduced/reused during the entire usage of the system in order to preserve system’s security.
- The information being exchanged defaults to “text message”.

III. MODUS OPERANDI

A. Statement of Purpose

To enhance security of LSB based audio file steganographic schemes using Wave file format.

B. Preliminaries

Confidentiality, integrity and availability (C.I.A.) of information merits security [27]. Cryptography is a preferred means to achieve confidentiality whereas Digital Signatures ensure integrity (of information) and non-repudiation (genuineness of sender). Availability, however, is linked with communication medium. To attain the said traits of Information Security we proceeded as follows:

- **Preferred Model:** As already stated we take in-audibility and adherence to Kerckhoff's principle as key traits to assure confidence in any audio related steganographic scheme, hence we have favored the model proposed by [28] which in turn is derived from [29] and [30] and is shown in Fig. 5. Our chosen model takes a preprocessed audio (cover) wave file and then performs key dependent varied (dynamic) 2-LSB bit embedding.

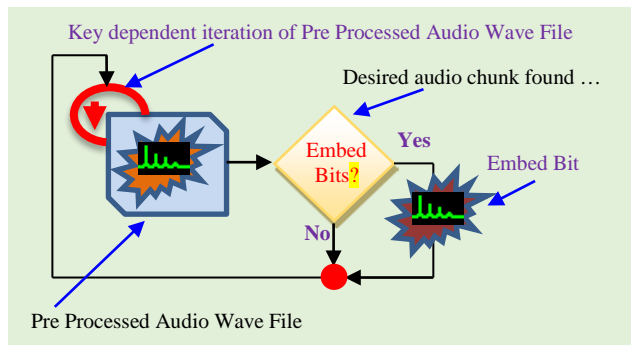


Fig. 5. Key Dependent Secret Bit Embedding

- **Message Header:** The noticeable facet of our conducted literature review is that apart from [31] remaining schemes are silent on issue regarding length of hidden information which defaults to processing of entire audio file and then to construct actual message from that which is extracted, though the logic used for its attainment is contentious and taken care of in our proposal. Further, all the schemes are silent over hidden data type which by convention seems to default as "textual information". Likewise, no concern shown towards integrity and non-repudiation of information being exchanged. Hence, to overcome the said impediments we constructed a twelve-byte header storing length of secret information, its type i.e., file extension, and its digital signature (three bytes for each) and appended it as a prefix to information being exchanged. Table 1 illustrates this distribution for visual comprehension. The choice of HASH and Digital Signature algorithm has been left at user discretion. We, however, have opted for SHA-2, 256-bit HASH algorithm [32] (using 32 leftmost bits) and have used exclusive-Or encryption just for the sake of presenting our concept. Public key encryption algorithm RSA [33]

is used to digitally sign message's HASH. Our proposed algorithm also make use of 256-bit Stego Key derived from [34] for test/experimental purposes.

TABLE I. MESSAGE HEADER WITH SCRAMBLED AND HIDDEN AUDIO DATA

- 32 bits -	-- 32 bits --	---- 32 bits ----	---(8 x N) bits---
Length of Secret File	Secret File's Extension	Digital Signature HASH	Audio Data

- **Pre Processing of Audio (Cover) Wave File:** A True Random Number Generator (TRNG) whose details are beyond the scope of this paper is used to generate random bit patterns that replaces 2-bit LSBs of every 8/16-bit audio sample (chunk) of wave file.

C. Information Hiding

Bit embedding process is illustrated in Fig. 6, and is explained as follows:

- Step 1: Iterate through every stego key byte and:
 - i. Reduce stego key byte value to modulo 2 i.e., numbers in range 0, and 1 that corresponds to number of bits to be inserted at the time of secret bit embedding.
 - ii. Store the result in sequence in a 32 byte array - say A1.
 - iii. Add the two adjacent stego key bytes starting from index 1 and reduce the result to (modulo 2) + 1. Store the result in another 32 byte array - say B1 in sequence. e.g., the adjacent byte for location B1 (31) will be B1 (0) and result will be computed as $B1(31) = (B1(31) + B1(0)) \text{ MOD } 2 + 1$.
- Step 2: Select audio (cover) wave file.
- Step 3: Preprocess selected cover.
- Step 4: Select secret file.
- Step 5: Calculate HASH of secret file.
- Step 6: Encrypt the computed HASH value with sender's private key.
- Step 7: Concatenate secret file's length, its extension along with encrypted HASH and affix it as pre-fix to selected audio wave file data.
- Step 8: Perform exclusive-Or operation between the output of step 7 and stego key bytes. Stego key is iterated continuously till the referred output gets completely processed.
- Step 9: Translate the outcome of Step 8 into bits which is the ready-to-embed secret information.
- Step 10: For key dependent dynamic point to start bit embedding equation (3) is used, where denotes total number of 8/16-bit audio samples constituting the wave cover file.
- Step 11: Iterate through preprocessed 8/16-bit audio samples starting from random position obtained vide Step 10 and proceed cyclically up to a point just before, by taking array A1 and B1 vide subsequent steps.
- Step 12: Iterate array A1 (i) | i=0 to 31, one at a time.
- Step 13: If A1(i) then take 'i' secret message bits and substitute these in place of one or two bits of 2-bit

LSBs of the selected sample starting at point indicated via $B1(j) | j=i$ in cyclic order.

Step 14: Terminate bit embedding process when all secret message bits have been got processed other-wise proceed to Step 12.

INSECURE COMMUNICATION CHANNEL

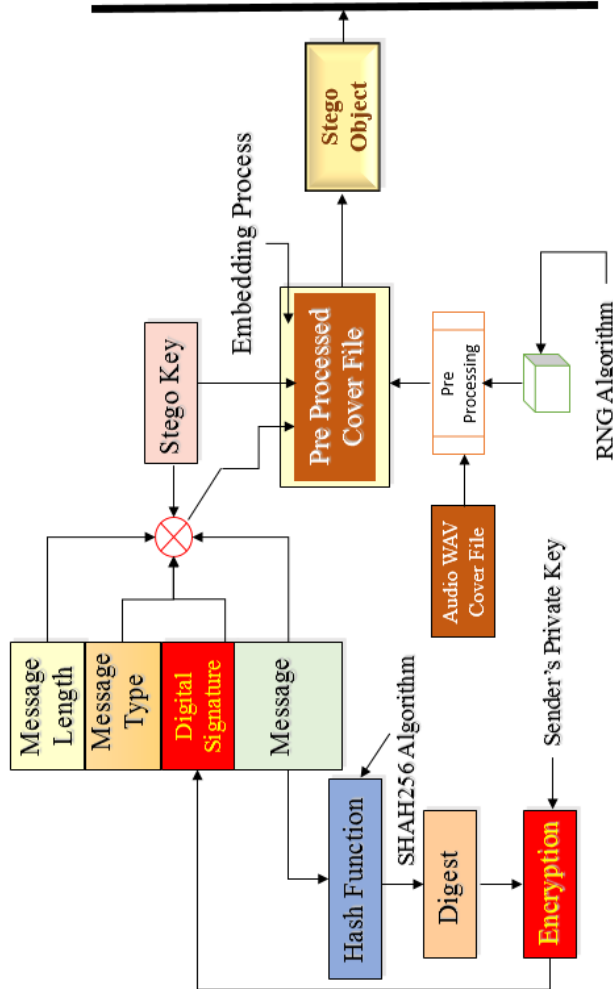


Fig. 6. Process of Bit Embedding

D. Information Extraction

Bit extraction process is shown in Fig. 7, and is explained as follows:

- Step 1: Same as Section III - C (Step 1).
- Step 2: Select audio (stego object) wave file.
- Step 3: For key dependent dynamic point to start hidden bit extraction use equation (3).
- Step 4: Iterate through 16-bit audio samples starting from random position obtained vide Step 3 ad proceed cyclically up to a point just before , by taking array A1 and B1 vide subsequent steps.

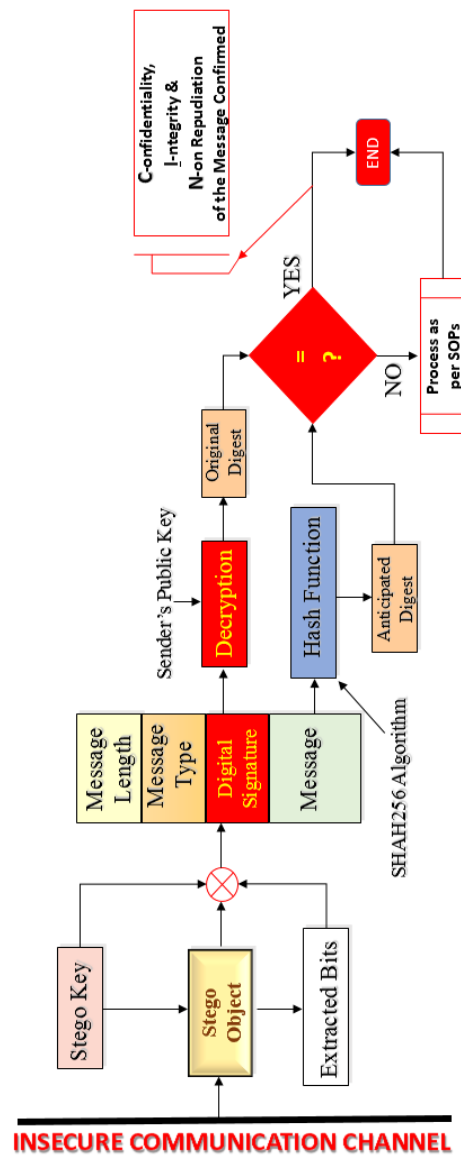


Fig. 7. Bit Extraction Process

- Step 5: Iterate array A1 (i) | i=0 to 31, one at a time.
- Step 6: If A1 (i) then extract and concatenate 'i' sample bits starting from point indicated via B1 (j) | j=i in cyclic order.
- Step 7: If first ninety-six have been extracted from samples being processed then advance to Step 8 otherwise proceed to Steps 5.
- Step 8: Perform exclusive-Or operation between the output of step 7 and stego key bytes. Stego key is iterated continuously till the referred output gets completely processed.
- Step 9: Translate the outcome of Step 8 into bytes which is the hidden header whose first 32 bits holds the length of information being hidden, next 32 bits gives its extension while the remaining 32 bits gives the digital signature.

- Step 10: Unfold the digital signature using sender’s public key and store the output for lateral usage.
- Step 11: Repeat Steps 6 and 7 till number of extracted bits equal to those contained in length of hidden message field of the header.
- Step 12: Translate the outcome of Step 8 into bytes which is the hidden information.
- Step 13: Calculate HASH of the hidden information.
- Step 14: Compare the computed HASH with the value stored vide Step 10. If the two values are identical then the received information is authentic and has arrived from authorized sender. If, however, the two values contradict then laid down Standard Operating Procedures (SOPs) be followed.

IV. TEST RESULTS

Audio files (16 bit, single channel: 1, 10 kHz and 100, 250 and 440 Hz of 30 seconds duration each) that we experimented with, have been downloaded from [35].

In addition to Signal-to-Noise Ratio (SNR) computation (discussed earlier) following have also been contrasted for an in depth analysis of our proposed bit embedding methodology using [36]:

- Mean Square Error (MSE)
- Root Mean Square (RMS)
- Peak Signal-to-Noise Ratio (PSNR)
- Equivalent Quantize Bits
- Maximum Deviation
- Arithmetic Mean
- Signal-noise Correlation
- Noise Autocorrelation
- Amplification

Table 2, Table 3 and Table 4 contrasts differences between source and preprocessed, source and stego object, and preprocessed and stego object respectively that speaks high of our proposed logic where the results out-shine those discussed in literature review.

TABLE II. CONTRASTING COVER AND PREPROCESSED AUDIO FILES

Parameters	1kHz	10kHz	100Hz	250Hz	440Hz
PSNR Square (dB)	86.37	86.38	86.37	86.37	86.34
PSNR Sine (dB)	83.36	83.37	83.36	83.36	83.33
SNR (dB)	80.16	80.17	80.16	80.16	80.13
Quantize Bits	13.553	13.554	13.554	13.554	13.548
RMSE (2 ¹⁶)	1.574	1.573	1.573	1.573	1.58
Max (2 ¹⁶)	3	3	3	3	3
DC (2 ¹⁶)	-0.0067	-0.0035	-0.0037	-0.005	0.0065
S/N Correlation	0.022294	0.023311	0.022879	0.022991	0.000227
Noise AC	0.014153	0.006709	0.002286	0.004856	0.012227
Amplification	1.000002	1.000002	1.000002	1.000002	1

TABLE III. CONTRASTING COVER AND STEGO OBJECT AUDIO FILES

Parameters	1kHz	10kHz	100Hz	250Hz	440Hz
PSNR Square (dB)	86.37	86.38	86.37	86.37	86.34
PSNR Sine (dB)	83.36	83.37	83.36	83.36	83.33
SNR (dB)	80.16	80.17	80.16	80.16	80.13
Quantize Bits	13.553	13.554	13.554	13.554	13.548
RMSE (2 ¹⁶)	1.574	1.573	1.574	1.573	1.58
Max (2 ¹⁶)	3	3	3	3	3
DC (2 ¹⁶)	-0.0067	-0.0035	-0.0037	-0.005	0.0065
S/N Correlation	0.02229	0.023307	0.02288	0.02299	0.000227
Noise AC	0.014153	0.006712	0.002284	0.004857	0.012225
Amplification	1.000002	1.000002	1.000002	1.000002	1

TABLE IV. CONTRASTING STEGO OBJECT AND PRE-PROCESSED AUDIO FILES

Parameters	1kHz	10kHz	100Hz	250Hz	440Hz
PSNR Square (dB)	131.52	131.19	132.03	131.52	129.54
PSNR Sine (dB)	128.51	128.18	129.02	128.51	126.53
SNR (dB)	125.31	124.98	125.82	125.31	123.33
Quantize Bits	21.053	20.998	21.137	21.053	20.723
RMSE (2 ¹⁶)	0.009	0.009	0.008	0.009	0.011
Max (2 ¹⁶)	3	3	3	3	3
DC (2 ¹⁶)	0	0	0	0	0
S/N Correlation	-0.00072	-0.00062	0.000181	-0.00025	-0.00001
Noise AC	0	0	0	0	0
Amplification	1	1	1	1	1

V. THEORATICAL SUBSTANCE

Let $A_1 = \{s_1, s_2, s_3, \dots, s_t\}$ be our cover audio file with ‘ t ’ samples, $M = \{m_1, m_2, m_3, \dots, m_i\}$ be the secret information of ‘ m ’ number of bits, and $K = \{k_1, k_2, k_3, \dots, k_n\}$ be the ‘ n ’-bit key that controls how $m_i \in M$ gets embedded in $s_t \in A_1$ and $C = \{c_1, c_2, c_3, \dots, c_t\}$ is the outcome (Stego Object) of operations where, through extended Van Diagram shown in Fig. 8, we may perceive the process of bit embedding as follows:

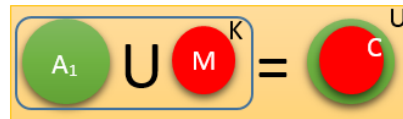


Fig. 8. Van Diagram Depicting Process of Bit Embedding

In principle A_1 Equals C only if outcome of encoding of M under control of K yields an exact pattern match as that of target bits of audio cover – Fig. 9 refers, chance of occurrences of which, however, are minimal. Hence, it is apparent from Fig. 10 that for known cover and stego object the hidden information whether encrypted or plain can easily be arrived at by searching for unrelated patterns even without the knowledge of secret key.

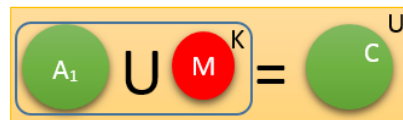


Fig. 9. Condition for Cover and Stego Object to remain Identical



Fig. 10. Extracting Hidden Information for Known Cover and Stego Object

The scenario for covert communication as perceived in [11] may be abridged as: For Alice and Bob to agree on their escape plan, Alice prepares a cover (e.g., - a painting) and sends it to Bob through Warden Wendy. Bob then alters the painting (e.g., - enhance its coloring etc. based on pre-agreed secret parameter) to implicitly convey details of escape plan and returns it to Alice that is more likely to get pass unnoticed by Wendy.

Foresaid above it is apparent that security of digital Steganographic system lies in deceiving Wendy to predict on actual cover used together with undetectable random diffusion of secret information inside its body, under control of some shared secret i.e., key, without the knowledge of which extraction of hidden information appear as hard-to-solve problem. Fig. 11 explicates on the concept (where preprocessed cover and extracted information are denoted by A_1' and M' respectively) to withstand Wendy's efforts in arriving at hidden information for a time sufficiently long enough where the significance associated with it loses its verve.

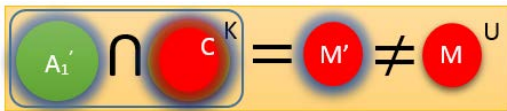


Fig. 11. Extrating Hidden Information from Preprocessed Cover and Stego Object

VI. PROS AND CONS

Following are the advantages and limitation of our proposed improvisation:

A. Advantages

- Abiding definition of steganography.
- In accordance with Kerckhoff's Principle.
- Availability of original cover at the time of secret message bit extraction not required at recipient's end.
- Procedure for concealing secret message bit does not permit over writing of previously written/concealed data.
- The bit embedding procedure does not (not) necessitate to keep original cover secret/destroyed in order to preserve system's security.
- Ant File type can be hidden inside audio (WAV) cover.

VII. FUTURE WORK

We are in process of extending our research to 2 or more LSB bit manipulation to accomplish high data embedding capacity without compromising on security.

VIII. CONCLUSION

This paper presented a secure LSB based audio steganographic scheme through revelation of limitations in existing ones arrived at via literature review of recently published studies on the subject. The author has first laid theoretical foundation of the concept of secure steganography followed by logical affirmation using mod-el based scenario. A

salient feature of the findings is that capacity of cover may not be attributed as an evaluation metric when it comes to evolution of secure steganographic scheme because the fewer the bits of secret information hidden in body of cover the more the chances are there for Wendy to commit false positive type of error and the more secure we are. An implicit trait of this research is to highlight the significance of Kerckhoff's Principle for steganographic systems whereas it has explicitly been shown that intelligence does not lie in embedding secret information in deeper sample layers of cover followed by alteration of its remaining bits but in inducing randomness (a technique which the Information Theory also endorse) during embedding process.

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Eliminating Broadcast Storming in Vehicular Ad-Hoc Networks

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Abstract—VANETs (Vehicular Ad-hoc Networks) offer diversity of appealing applications. A lot of applications offered by VANETs depend upon the propagation of messages from one vehicle to another vehicle in the network. Several algorithms for useful broadcasting of safety/ warning messages in the network have been presented by different researchers. Vehicles on roads are increasing day by day. Due to this increased number of vehicles and especially during the crest hours, when the networks become very dense, dissemination of the messages blindly in the network causes problems like packet collisions, data thrashing and broadcast storming. In this research, a relative speed based waiting time algorithm has been presented for avoiding broadcast storming problem in the VANETS especially in dense environment. This proposed algorithm calculates the waiting time for each vehicle after receiving the safety/ warning messages according to the relative speed of the vehicles, the distance between the vehicles and range of vehicles. The results show that the proposed relative speed based algorithm is better than already existing algorithms like blind flooding and dynamically broadcasting waiting time algorithm which uses number of neighbors and distance between the vehicles for calculating the waiting time.

Keywords—VANETs; Intelligent Transportation Systems; Broadcast Storming; Distance based flooding

I. INTRODUCTION

VANETs are Ad-hoc networks for vehicles on roads. These networks are temporary networks created for small amount of time since vehicles are moving at a rapid speed. Normally these networks are used to provide safety on road by transmitted information from other vehicles or infrastructure. VANETs fall under the broader category of ITS (Intelligent Transportation Systems). ITS are used for improving the efficiency of transportation systems using advanced technologies. The major objectives of ITS are to enhance road safety, provide convenience of information, reducing traffic congestions, improving user experience on roads and enforcing traffic laws [1]. Car navigation systems, railroad crossing alarms, speed checking cameras and electronic tolls gates are all examples of ITS.

In VANETs information transmission between vehicles should be reliable enough to ensure maximum safety on roads. This information transmission can be between 2 vehicles or between several vehicles at the same time. This concept of sending information to several vehicles is called information broadcasting. In VANETs, many techniques like Simple

Flooding, Probabilistic Flooding, Neighbor knowledge Flooding, Area based Flooding, Distance based Flooding, and Location based Flooding [2][3] are available for broadcasting information between vehicles and infrastructures. Increase in broadcasting these messages can increase packet collision and data loss. This increase in packet collision and data loss due to dense network is called Broadcast Storming.

Several techniques are available to avoid broadcast storming in VANETS. Some of these techniques are 1-persistence, slotted 1-persistence and weighted p-persistence techniques [4]. In order to minimize the broadcast storming problem, a new relative speed based waiting time calculation approach is adopted. This approach adjusts the waiting time of a vehicle for broadcasting a warning or safety message for other vehicles in the network dynamically with respect to the distance of the vehicle from the base vehicle and the relative speed. If the speed of the vehicle is slow, its waiting time is increased. The vehicle, which is moving with the highest speed and has more number of neighboring vehicles will have very little waiting time and will broadcast within no time.

In order to transmit information between vehicles a communication technology is required. However, since in VANETs, there are many vehicles on road and equipping all the vehicles with expensive modules would increase the overall cost of the system. There for the purpose of this research, a low cost and power efficient technology, ZigBee is used. ZigBee supports star, mesh and peer to peer topologies which are compatible with VANETs. It can provide high scalability and has better communication range as compared to Bluetooth and Wi-Fi devices. This technology can help in eliminating redundancy of safety messages being broadcasted, because it works in two different modes i.e. beacon mode and non-beacon mode that can facilitate the traffic in random directions. When the coordinator works with the help of batteries the beacon mode is used and in this way it proposes more power reduction and savings. On the other hand, non-beacon is more preferred and gets more favor when coordinator is running on the main supply of power. Furthermore, some researches have shown that ZigBee works well with moving objects like vehicles even if traffic is congested. Moreover, since these wireless modules work in outdoor environment, it is important to use a technology which can sustain extreme weather conditions like snowfall and torridity. Different researches have shown that ZigBee performs well in extreme weather conditions.

Rest of the paper is organized as follows: Section II deals with the related work in this field and analysis of already done research in this area, along with identifying the gaps for research. Section III proposes a model for eliminating the broadcast storming for four lanes scenario. Section IV explains the flow of the proposed system. Section V describes the proposed algorithm for eliminating the broadcast storming in VANETs. Finally the last section concludes the paper.

II. RELATED WORK

A lot of work has already been done to avoid the broadcast storming problem. This problem was firstly examined in the MANETs. Different authors have suggested different techniques to diminish the problem. These techniques are used to swiftly minimize the redundancy of the safety messages and are being used in VANETs for mitigation of the broadcast storming. This section describe literature review for the research.

NajafZadeh et al. [5] proposed a waiting time based algorithm named "Dynamic Broadcasting Algorithm" to reduce the number of rebroadcasts of same message time and again. In this method they describe that nodes prepare their neighbor's list by using hello beaconing. Their algorithm calculates the waiting time for a node to rebroadcast a message by number of neighbors and the distance between sender node and the other vehicles. By using simulation results, they have concluded that their presented dynamic broadcasting algorithm is better in performance than flooding technique and random waiting time protocol. This waiting time based algorithm helps to reduce the redundancy of safety messages and minimizes the broadcast storm.

Suriyapaibonwattana et al. [6] proposed an algorithm for minimizing broadcast storm in VANETs and called it "The Last One broadcasting method". In this method, when a vehicle receives information, it rebroadcasts this information by using random probability. This method is very useful to minimize the broadcasting storm in VANETs but cannot fully eliminate the broadcast storm.

Chiasserini et al. [7] discussed a scheme for achieving a smart broadcast analogy by using a channel access mechanism in vehicular ad hoc networks systems. This scheme states that the vehicles, which are going to rebroadcast the warning messages, will access channel with the different priority. This priority will depend on the distance between the last vehicles, which rebroadcasted warning message from the vehicles which will rebroadcast.

Francisco J. Ros et al. [8] explained a protocol for overcoming broadcast storm in vehicular ad hoc networks. This is modified form of PBSM (Parameter less Broadcasting in Static Mobile Wireless Ad-hoc) and is called AckPBSM (Acknowledged PBSM). PBSM can work with both 2-hop and 1-hop topological information. In this method every vehicle use a buffer to store the alert message and this message is discarded after a certain time. This AckPBSM works same like PBSM except that in AckPBSM broadcast alert messages are acknowledged and due to this acknowledgement the vehicle which has already received the same message will not

rebroadcast the same message. This phenomenon helps in reducing the broadcast storm.

Roberta Frachia et al. [9] proposed a broadcast storm avoiding algorithm which is based on distance between the vehicles. Their distance aware flooding technique depends on distance between the current node and closest neighboring node. This algorithm generates a waiting time for a message to rebroadcast. By using simulation results they have concluded that their proposed distance aware delayed flooding algorithm yields better performance to avoid broadcast storm in VANETs than other broadcast suppression techniques.

Koubek et al [10] proposed a scheme to minimize the number of vehicles, which broadcast the message regarding the identical event. Delays and redundancy of the data, which is being transmitted, have been removed significantly by using the proposed methodology. Due to erratic protocol used for broadcasting this ESSMD has decrementing probability for event response. This response rate has been enhanced by modified scheme, which is ESSMD + Rep. This scheme enhances the trustworthiness of the protocol by replicating broadcasts on base vehicles. ESSMD technique has been stimulated by the use of ideology of data aggregates, which is minimizing the redundancy of the transmission. This broadcast elimination technique seems to confine the base vehicles which are reporting on identical events in spite of cumulating information from numerous base vehicles at solitary point. This technique has been evaluated as much better technique than the already existing aggregation methods used for reducing the storm of broadcast messages. ESSMD + Rep uses the repetitions to enhance the consistency of booming broadcasts while the physical path is not burdened. By raising the repetitions enhanced reliability has been accomplished at charge of enhanced transmissions that can direct towards a flooded network. In both schemes i.e. Restricted-Mobility-Based protocol has been used in replacement of ordinary flooding protocols used for broadcasting.

These researches by the authors concerning the broadcast of safety/ warning messages between the vehicles on road has paid attention to the problems like medium contention, data packet collisions and broadcast storming etc. Their focus is mainly on medium access control, collision avoidance, message propagation protocols and broadcast storming avoidance algorithms and techniques. All these algorithms and techniques have benefits as well as their limitations. In this research, our focal point is the alleviation of broadcast storming in VANETs by using relative speed based technique to make the network efficient and reliable.

III. PROPOSED SYSTEM MODEL

This is a four lane road scenario. All of these vehicles are equipped with ZigBee devices as onboard equipment for communication with each other and with RSUs. In the given scenario, the vehicle "A" collects the information of the accident first and starts to broadcast this information of the accident to other vehicles in the network. A message "ACCIDENT AHEAD" is transmitted to all vehicles in the communication range of vehicle "A" and to the road side

units. All these vehicles, which receive this message from vehicle "A", calculate their waiting time according to the proposed relative speed based technique. After calculating the waiting time, these vehicles start to count down and when this waiting time reaches to zero, they start broadcasting the received information to other vehicles in the network. If the waiting time of the vehicle has not reached to zero and meanwhile, this vehicle receives the same information from any other vehicle in the network or from RSU, then it does not broadcast and discards this information. Vehicles also broadcast the information received from RSUs. Like vehicle with onboard equipment, RSUs are able to receive and broadcast the messages in the network. These RSUs can also save the received information and can send this information to the vehicles, which later enter in the communication range of the RSU.

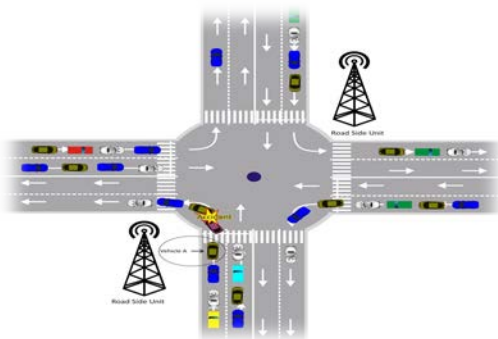


Fig. 1. Proposed system model

Fig. 2 shows a sparse network in which two vehicles are moving with the same speed. Due to moving with same speed, the relative speed of both the vehicles is zero. According to our proposed technique, when relative speed is zero, the wait time for rebroadcasting the warning/ safety message in the network becomes equal to maximum wait time for rebroadcasting. Due to this maximum wait time, the chances for receiving a duplicate packet of the same information are increased. As a result, rebroadcasting of the same message by both vehicles will not be ensured and hence broadcasting storm is minimized.

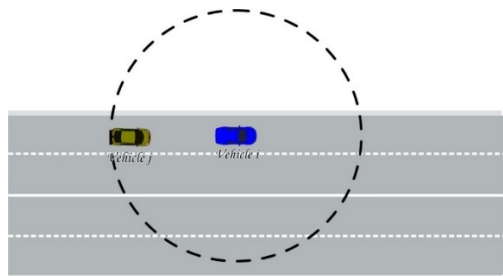


Fig. 2. A sparse network

Fig. 3 is a scenario of a traffic jam or traffic stuck on a signal. In this scenario, when all vehicles broadcast the warning/ safety messages according to the distance based technique or according to dynamic waiting time broadcasting algorithm, there are high chances of broadcast storming of the same information. In this scenario, almost every vehicle has

some neighbors. Here, if we consider the relative speed of the vehicles for calculating waiting time for each vehicles to broadcast the safety/ warning message in the vehicular Ad-hoc network, it will be much beneficial than the other existing algorithms. In this case the speed of each vehicle within the network is approximately zero and hence their relative speed is also zero. Here the waiting time for each vehicle comes out equal to maximum waiting time required for broadcasting.

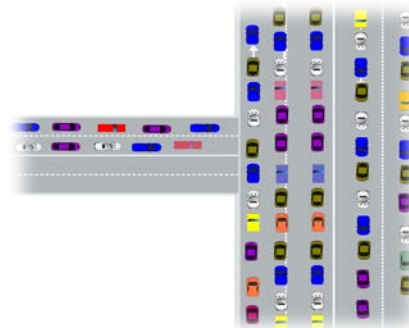


Fig. 3. Traffic Jam scenario

When each vehicle will wait for maximum time for rebroadcasting, there are more chances of getting the same information from the other vehicles. When a vehicle gets the same packet from other vehicle within the stipulated waiting time, it discards this packet rather than rebroadcasting. As a result of increased waiting time, a number of vehicles get the duplicate packet and discards it and hence broadcast storming problem is eliminated up to much extent as compared to other broadcasting techniques.

In Fig. 4, some vehicles in region "A" are stopped on a signal and other vehicles in the network are moving with some speed. Vehicle "V1" receives the information and starts broadcasting this information to other vehicles in the network which are in its communication range. Other vehicles within the broadcasting range of the vehicle "V1" calculate their waiting time according to proposed relative speed based waiting time algorithm using their relative speed with respect to vehicle "V1". When vehicle "V2" counts down its waiting time to zero and no duplicate packet of same information is received from any other source in the network, it starts rebroadcasting the safety / warning messages received from vehicle "V1" or Road Side Units. Other vehicles in the communication range of vehicle "V2" calculate their waiting time according to proposed algorithm by using their relative speed with respect to "V2".

The vehicles with more difference in their speed have much relative speed and have less waiting time for broadcasting the safety/ warning message. Similarly the vehicles which are moving with same speed or have no speed i.e. stopped, their relative speed is approximately zero and they have more waiting time before broadcasting. As waiting time before broadcasting is increased, the number of vehicles broadcasting the warning/ safety message in the vehicular network is reduced. As number of vehicles broadcasting the warning / safety messages is reduced, the broadcast storming problem is automatically be minimized.

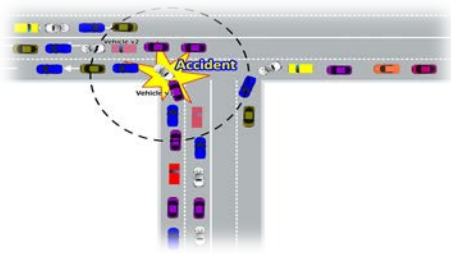


Fig. 4. Occurrence of an accident on traffic signal

IV. SYSTEM FLOW

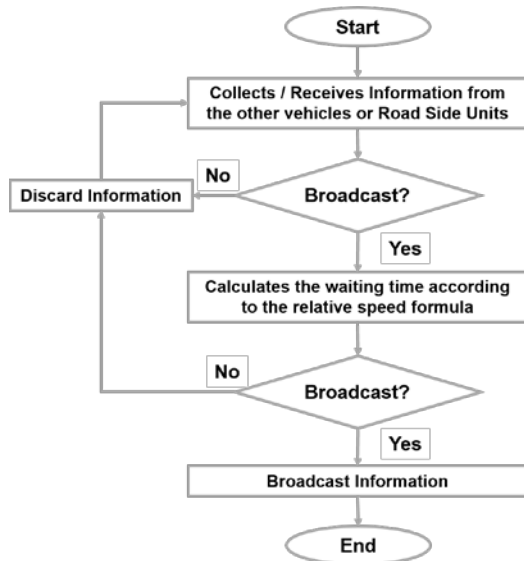


Fig. 5. Proposed System Flow of the model

The vehicle which collects information from environment or from Road Side Unit or any other source of the network, it checks whether it has received the same warning/ safety message already from any other source of the network. If it has already received the same packet it is discarded. If the vehicle has not already received the same packet, it calculates waiting time for broadcasting according to its relative speed with respect the sender vehicle. Now this vehicle waits until the waiting time counts down to zero. If the waiting time has not yet counts down to zero and in the meanwhile it receives the same information from any other vehicle in the network or from Road Side Unit, it discards the information and hence no rebroadcasting. If the waiting time has counts down to zero and the vehicle has not received the duplicate information from any source of the network it starts broadcasting the information in the network.

V. PROPOSED ALGORITHM

To tackle with the no broadcast and broadcast storming problems weighted p-persistence technique has been used. This technique works similar to p-persistence except that it uses probability for rebroadcasting the safety/ warning messages in the network which is calculates as under [11] :-

$$P_{ij} = \frac{D_{ij}}{R}$$

Where D_{ij} refers to the distance between the sending and receiving vehicles and R is maximum range of network.

In 1-persistence, the network has multiple numbers of slots with the constraint that the vehicles should be within any one of the slots otherwise the vehicle will not be able to send and receive the information. In this technique, when a vehicle receives a message from any other vehicle in the network, it starts to wait for some specific time and when this time counts down to zero then this vehicle starts to broadcast the message. Each vehicle has different time for waiting. This difference in time for wait is due to the different slots to which the vehicles belong. In this technique, the vehicles calculate the time for wait as under [12]:-

$$T_{S_{ij}} = S_{ij} * \tau$$

Where τ represents the wait time which is required for the multiple hop transmission and S_{ij} represents slots present within the broadcasting vehicle and receiving vehicle. We can calculate S_{ij} as under:-

$$S_{ij} = N_s \left(1 - \left\lfloor \frac{\min(D_{ij}, R)}{R} \right\rfloor \right)$$

Where N_s represent total slots, D_{ij} represents distance of broadcasting vehicle from the receiving vehicle and R is the range of the transmission. It is concluded that the vehicle which is at the most distance waits for less time.

A dynamic broadcasting technique has been proposed which is distance based technique, in which the base vehicle have knowledge of its neighboring vehicle with the help of hello beaconing. This technique considers not only the distance for calculating the wait time but also considers the number of neighboring vehicle. With the help of this technique the waiting time is calculated as under [13]:-

$$W_t = \left(1 - \frac{D_{ij}}{R} \right)^n W_{t_{max}}$$

Where n represents the neighboring vehicles, D_{ij} represents the distance of broadcasting vehicle from the receiving vehicle, R represents the range of the transmission and $W_{t_{max}}$ represents the maximum wait time. If there is present only a single neighbor the vehicle does not wait and broadcasts the message. The vehicle which is at the most distance from the base vehicle and more vehicles are its neighboring vehicles then it will have high priority to broadcast. This technique has better performance than random waiting time but this technique suffers in dense networks like when the vehicles are stopped at traffic signal or traffic is jam, then in such scenarios almost every vehicle has a lot of neighboring vehicles. There will be again a broadcasting storm in the network. As we know that in the sparse vehicular network, there is no problem of broadcasting storm. Its most threat is present in the dense networks. In the dense networks the vehicles often have slow speed as compared to the sparse network. We have optimized the results by modifying the above dynamic broadcast technique by adding the relative speed of the vehicles to calculate the wait time for initiating the broadcasting. By our technique the waiting time for each vehicle present in the network is calculated as under:-

$$W_t = \left(\frac{R - S_{ij}}{R}\right)^{RS_{ij}} W_{t_{max}}$$

Where, W_t represents the waiting time of vehicle for broadcasting the message, R represents the range of the forwarding/ broadcasting vehicles and S_{ij} represents the distance between the vehicle “i” and vehicle “j”, RS_{ij} represents the relative speed of vehicle “i” with respect to vehicle “j” and $W_{t_{max}}$ represents the maximum waiting time for a vehicle to wait before broadcasting the safety/ warning message in the network. With the addition of relative speed we have more optimized results for the reduction of the

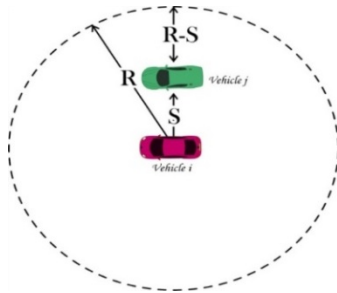


Fig. 6. Range of vehicle ‘i’ and its distance from vehicle ‘j’

broadcasting storm in the dense networks than above dynamic technique and random time technique. The use of relative speed of the vehicles to alleviate the broadcasting storm in the VANETs is unprecedented.

Waiting time for vehicles, the time for which the vehicles has to wait before broadcasting has been calculated for both Dynamic Waiting Time Algorithm and our proposed Relative Speed Based Waiting Time Algorithm. Then the resulting waiting time of the both algorithms has been compared.

In the following table we have used the data of such road scenario where traffic is very dense and all the vehicles are

moving with some speed. Some vehicles are moving with high speed and some are moving with low speed. Due to dense network, every vehicle has no. of neighbors and some relative speed with respect to other vehicles present in the network. Maximum waiting time has been chosen 20 milliseconds i.e. every vehicle will not wait more than 20 milliseconds before broadcasting if it would not get any duplicate information from any other source in the network.

The comparison of waiting time by using Dynamic Broadcasting Algorithm and proposed relative speed based waiting time algorithm shows that the waiting time for each vehicle by relative speed based algorithm is more or equal to the waiting time by dynamically broadcasting algorithm. It is clear that if each vehicle in the network has to wait for more, it has more chances of getting duplicate information from any other vehicle or road side unit in the network and has low probability for broadcasting the safety/ warning message in the network. The reduced probability for broadcasting results in less broadcasting nodes in the network as most of the vehicles receive duplicate packet of the information while waiting and discards this information instead of broadcasting. As a result, there will be few nodes broadcasting and forwarding the safety/ warning messages in the network. Table II shows the data and results for vehicles which are on a traffic signal, where some vehicles are moving with some speed and other vehicles are stopped on the signal and have zero relative speed. As it is a signal, there are more number of neighbors for each vehicle and by dynamic broadcasting algorithm waiting time will be less for each vehicle and these vehicle have more probability of broadcasting the safety/ warning messages in the network. But according to our proposed relative speed based algorithm, as it is traffic signal and relative speed of the vehicles is less or zero, each vehicle in the network has low probability of broadcasting the safety/ warning messages in the vehicular Ad-hoc network due to more waiting time.

TABLE I. EXPERIMENTAL DATA FOR RANDOMLY MOVING VEHICLES IN DENSE TRAFFIC AREA

V_i	V_j	R (m)	S_{ij} (m)	nn	Relative speed (km/h)	$W_{t_{max}}$ (ms)	W_t by Dynamic waiting time algo. (ms)	W_t by our proposed waiting time algo. (ms)	W_t of relative speed algorithm as compared to dynamic waiting time algorithm
V1	V2	100	50	10	5	20	0.195	0.625	More
V1	V3	100	60	20	20	20	2.19×10^{-7}	2.19×10^{-7}	Equal
V1	V4	100	45	25	15	20	6.45×10^{-6}	2.45×10^{-3}	More
V1	V5	100	90	50	10	20	0	2.19×10^{-9}	More
V2	V3	100	80	20	15	20	2.97×10^{-13}	6.55×10^{-10}	More
V2	V4	100	10	25	20	20	1.435	2.431	More
V3	V4	100	100	25	35	20	0	0	Equal
V4	V5	100	95	50	30	20	1.77×10^{-14}	1.86×10^{-8}	More
V3	V5	100	75	50	20	20	0	1.82×10^{-11}	More
V2	V6	100	35	40	10	20	6.56×10^{-7}	0.269	More

TABLE II. EXPERIMENTAL DATA FOR TRAFFIC ON A SIGNAL

V_i	V_j	R (m)	S_{ij} (m)	nn	Relative speed (km/h)	W_{max} (ms)	W_t by Dynamic waiting time algo. (ms)	W_t by our proposed waiting time algo. (ms)	W_t of relative speed algorithm as compared to dynamic waiting time algorithm
V1	V2	100	2	15	5	20	14.771	18.078	More
V1	V3	100	3	14	10	20	13.056	14.748	More
V1	V4	100	10	13	10	20	5.083	6.970	More
V1	V5	100	10	15	20	20	4.117	2.431	Less
V2	V3	100	20	14	10	20	0.879	2.147	More
V2	V4	100	15	13	0	20	2.419	20	More
V4	V3	100	10	14	0	20	4.575	20	More
V3	V4	100	15	13	0	20	2.419	20	More
V3	V5	100	5	15	0	20	9.265	20	More
V4	V5	100	4	15	0	20	10.841	20	More

In Table III, the data for a traffic jam scenario has been used and the results have been calculated by both the algorithms. As it is traffic jam, the vehicles are stuck on the road. Each vehicle has more number of neighbors and due to traffic jam the network is highly dense and speed of each vehicle in the network is zero and hence the their relative speed is also zero. Due to more number of neighbors the Dynamically Broadcasting algorithm yields less waiting time

for each vehicle as shown in the above table and have high probability of broadcasting the safety/ warning messages in the network. The proposed Relative Speed Based waiting time algorithm yields waiting time equal to maximum waiting time for each vehicle in the network due to zero relative speed. This means that each vehicle has to wait for maximum waiting time before broadcasting and has very low probability for broadcasting the warning/ safety messages in the network which results in less broadcasting nodes in the network.

TABLE III. EXPERIMENTAL DATA TRAFFIC JAM SCENARIO

V_i	V_j	R (m)	S_{ij} (m)	nn	Relative speed (km/h)	W_{max} (ms)	W_t by Dynamic waiting time algo. (ms)	W_t by our proposed waiting time algo. (ms)	W_t of relative speed algorithm as compared to dynamic waiting time algorithm
V1	V2	100	10	15	0	20	4.117	20	More
V1	V3	100	22	14	0	20	0.617	20	More
V1	V4	100	15	10	0	20	3.937	20	More
V1	V5	100	5	16	0	20	8.802	20	More
V2	V3	100	20	14	0	20	0.879	20	More
V2	V4	100	30	10	0	20	0.564	20	More
V2	V5	100	4	16	0	20	10.408	20	More
V3	V4	100	50	10	0	20	0.019	20	More
V4	V5	100	6	16	0	20	7.431	20	More
V3	V5	100	15	16	0	20	1.485	20	More

VI. CONCLUSION

To minimize the broadcast storming problem in Vehicular Ad-Hoc Networks efficiently, a relative speed based dynamically broadcasting waiting time algorithm has been proposed for broadcasting the safety/ warning messages in the Vehicular Ad-Hoc Networks. The experiments show that the relative speed of the vehicles has a great concern in broadcast storming problem. Therefore, the proposed relative speed based dynamically broadcasting waiting time algorithm calculates the waiting time for broadcasting the same message again in the network using distance between forwarding and receiving vehicles, broadcasting range of the vehicles and speed of the vehicles relative to each other.

The results of the proposed algorithm has been compared with the results of an already existing dynamically broadcasting algorithm which uses number of neighbors of the receiving vehicle and its distance from the propagating vehicle to calculate the waiting time for rebroadcasting the safety/ warning messages in the network.

The results of both broadcasting algorithms have been compared in different scenarios i.e. in dense networks, in sparse networks and in partially dense and sparse networks. In all scenarios, the results show that the proposed scheme yields more waiting time for broadcasting the messages in the network than already existing waiting time protocols. Particularly, in dense networks, the proposed relative speed based dynamically broadcasting waiting time algorithms performs much better than the already existing dynamically broadcasting waiting time algorithms, distance based algorithms and blind flooding etc.

Due to yielding more waiting time for each vehicle present in dense or sparse network for broadcasting messages in the network as compared to other waiting time protocols, the proposed algorithm is much better in terms of suppression of broadcast storm in the VANETs.

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Architectural and Qos Issues in Mobile Cloud Computing Environment for Real-Time Video Streaming

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Abstract—this paper state the issue related to mobile cloud computing for real-time video streaming. Recently the media streaming particularly video streaming becoming very popular among the users of hand held devices. Due to exponential growth in mobile device users, mobile network is facing large traffic of video signals. Due to this growth of users, video streaming especially real-time video streaming is now an interesting area of research. To increase the streaming speed and quality of real-time video a lot of work and progress is being done in hardware architecture of mobile devices. We have multi-core processors and enough memory space in modern mobile devices. The concept of GPU is also introduced in mobile devices to overcome the problem of speed in video data processing. But very less work is done to utilize the capability of these devices through parallel architecture development for real-time video streaming process. So it is the matter of discussion and research that how the computational and memory power of these devices can be utilized through parallel computations to speed up the process of video data up to ultra high speed. Mobile cloud computing environment can play an important role in this field. So In this paper, the review of architectural and QoS issues in mobile cloud computing environment for real time video steaming are being presented. There are many real-time video streaming techniques for mobile devices, some possible parallel architecture and QoS (Quality of Services) issue to leverage the architecture capability of the mobile cloud computing environment for these real-time video streaming is being discussed in this paper.

Keywords—QoS; Video streaming; Mobile cloud computing; A mobile device; GPU; Parallel algorithms

I. INTRODUCTION

In few next year's RTVS (real-time video streaming) in mobile devices will be a common and generalized application. RTVS in mobile devices may be improved by using MCC (mobile cloud computing) environment. MCC is the area in which mobile computing and cloud computing concepts and benefits are integrated to get better performance in mobile devices. Due to exponential growth in handheld device users and betterment in telecommunication services RTVS in handheld devices (Especially in Smartphone) is going to be a very common and usual application. Smartphone and tablets have cellular, wifi and Bluetooth interfaces facilities. Task offloading from Smartphone to the cloud is a promising

strategy to enhance the computing capability of Smartphone and prolong their battery life [14]. Utilizing multiple links parallel can improve video streaming in several aspects [15]. Now day's Smartphone are also equipped with multiprocessor systems; that can also be utilized for parallelization of streaming process.

In this progress, some parallel algorithms for processing of video streaming in different steps have been proposed. One of the techniques is motion estimation. Accurate motion estimation between frames is crucial for drastically reducing data redundancy in video coding. However, advanced motion estimation methods are computationally intensive and their execution in real-time usually requires parallel implementation [4]. Similarly, many techniques are used for RTVS process enhancement. But rare work is done via MCC in this field. Since RTVS play an effective role in education, public safety, health care, real estate industry, etc. so there is a need to have study about the improvement possibilities in this field.

This paper is aimed to address the issue related to RTVS over MCC environment. The issues discussed in this paper are the standardization of the architecture of MCC, operations over MCC, QoS of MCC, operations in MCC and economics of MCC services. The recent progress of digital media has stimulated the criterion, storage, and distribution of data requiring efficient technologies to increase the usability of these data. Video summarization methods generate concise summaries of video contents and enable faster browsing, indexing and accessing of the large video collection. However, these methods often perform slowly with large and high-quality video data. One way to reduce this long time of execution is to develop a parallel algorithm, using the advantage of the modern computer architectures that allow high parallelism [21].

MCC has some feasible solution to the inherited limitation of mobile computing. These limitations include battery lifetime, processing power and storage capacity. By using MCC, the processing and the storage of intensive mobile device job will take place in the cloud system, and the result will be given to the mobile device. This method of computation needs less power and time to compute intensive jobs like real-time videos streaming [28].

The remaining part of this paper has five more section. In section II, this paper elaborate MCC architecture which may support the RTVS. In section III, paper explores different video streaming algorithm and technique either based on simple or MCC architecture. This paper also discusses possible future work, findings, and improvements given in many recent research articles and papers in section IV. In section V, this article discusses the social and legal issue related to RTVS and how it is beneficial for health sector, education, social connectivity and security. Finally, this paper concluded the discussion in section VI.

II. MOBILE CLOUD COMPUTING ARCHITECTURE FOR REAL-TIME VIDEO STREAMING

MCC is born to leverage power full computing and storage resource in the cloud to provide abundant services in mobile environment conveniently and ubiquitously. The feature of MCC includes no upfront investment, lower operating cost, highly scalable and easy access etc [26].

The major benefit of cloud computing for mobile devices is ability to run applications between resource constraint devices and internet based cloud. Hence resource constrained device can outsource computation, communication and resource intensive operations to the cloud [24].

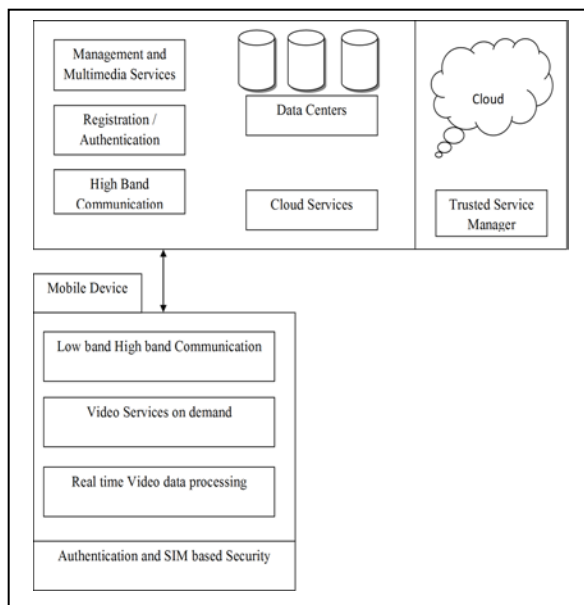


Fig. 1. Service Architecture for real-time video processing over mobile cloud computing environment

A. System Modeling

Here paper is presenting a generalized architecture for real-time video processing that relies on mobile and cloud computing models. By developer point of view following inherited requirement must be supported by architecture.

- Method of video data communication
- Method of security to have secure communication which control the cloud access

- Device/Resource registration used to access mobile cloud system by new users
- Environment constraints which effect the execution of video data processing that is performed in mobile cloud and mobile device

In 2014-2015 the use of Smartphone has changed dramatically. For example now days mobile user expect to have full access to their own data and also have desire to use all kind of application on their mobile devices specially Smartphone. Beside significant evolution in mobile devices, these are still and will always be limited in terms of processing power, storage, band width and energy. They also face the problem of less reliable connection less secure communication in compression of stationary devices.

In order to face these limitations the new generation of mobile applications already relies on, e.g. cloud augmentation as supported by current cellular network standards (3G, LTE) which allows overcoming such intrinsic restriction of mobile devices [9].

Since video processing application typically comprise computational and data intensive tasks, so MCC can be used to speed up the process. In fig 1, MCC architecture for real-time video streaming is given. Such type of cloud augmented application may be advertizing, video gaming, online classes and recognition assistant.

B. MCC Architecture

In general scenario mobile applications just rely on the mobile device capabilities, but some mobile applications may require the cloud resources to provide their full functionality. Such type of applications is called MCC-application. RTVS is one of those type of application, that may be executed entirely on the mobile device, but are able improve their performance by offloading certain parts onto cloud resources.

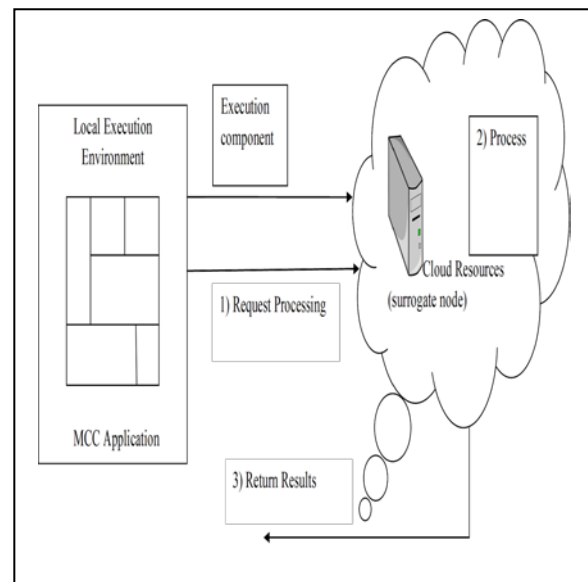


Fig. 2. Offloading execution component to cloud resources (surrogate nodes)

Fig 2 shows a generalized offloading task in which some part of the application's code are processed on a cloud resource which may be called surrogate node and the computed results are given back to the mobile device. There are following common component of MCC which enables this type of computation offloading.

- A partitioner
- Context monitor
- Solver
- Coordinator

Partitioner analyzes the application and determines the part of application's code which has candidature of offloading. Context monitor check the context information like available surrogate node, battery status and network connectivity then this information is used by solver which allocate the surrogate node to application's code . Coordinator perform additional task like discovery, authentication and synchronization.

C. Essential Requirement of MCC Architecture

From section II.A and II.B it is clear that MCC application development is not easy as simple mobile application. Thus it needs proper support to ease the development. Inter process communication, remote method invocation or service invocation models cannot be employed as it is in MCC.

Along with all the problem of distributed system like heterogeneity, security, scalability failure handling, concurrency, transparency and quality service some other problem are also exist in MCC, e.g. limited resource etc, these problem must be in consideration at the time of development of MCC applications. Additionally MCC applications should also support reliability, usability, efficiency, maintainability and portability as define in ISO/IEC 2501n. So here we are discussing combined general requirements of this category.

A good MCC application must support following criteria.

- 1) Availability
- 2) Scalability
- 3) Usability
- 4) Maintainability
- 5) Security

Availability: - To perform a computation in MCC environment a reliable connection must be available to all surrogate nodes. This availability must have low latency and high bandwidth.

Portability: - This is an important function in case of MCC. Since there is frequent transfer of application computation between mobile and surrogate. This shifting is dynamic in nature.

Scalability:- Application which run on heterogeneous and changing environment required dynamic partitioning. Due to remote execution of partitioned task and return of results to mobile need extra overhead in term of computation and bandwidth. MCC architecture assumes that cloud resource is infinite but this is not the case in practical. Since MCC need surrogate and task are being executed parallel so every

partitioned task is assigned a surrogate. This generally require a concurrency handling issue that affect the scalability of solution

Usability: - In MCC there are many issues which affect its usability, like energy issue, bandwidth issue, end user complexity etc. So after considering these issue MCC application should be easy and intuitive. Additionally an adequate abstraction should be there to hide the complexity of implementation of MCC. Adequate abstraction mean without compromising the full potential of MCC application.

Maintainability: - Software development for mobile devices now days is extremely agile and repetitive task. To support this task up to satisfactory level, it is required that the integration of MCC feature does not lead to increase complexity. Common development also suggests ease of bug fixing. Finally the architecture should be depending on open standards so vendor lock-in may be prevented with above facility and maintainability can be achieved up to adequate level.

Security: - Multi tenancy, concurrency and distribution are main feature of any cloud computing architecture. But due to these feature architecture compromises with security. So it is a never ending issue to have secure cloud architecture. Transferring confidential or private data to cloud node (surrogate node) always signals the risk of losing control over confidential or private data. Thus an easy to use solution/tool is requiring which may categorize and prevent the sensitive information to leave the trusted device (Mobile).

D. Some existing solutions for the essential requirement of MCC

MCC have special kind of requirement we focus on these requirement and here discussing some existed solution for these requirements. After doing the extensive survey we have classified existing solution according to their nature in terms of performing the computation offloading.

Existing solutions are categorized in six domains

- 1) Specific language
- 2) Middle ware and frame work
- 3) Distributed virtual machine
- 4) Universal computable solution
- 5) Native(non VM-base) MCC solution
- 6) Native(VM based) MCC solution

Specific language:- For the purpose of programming there are many programming languages for e.g. domain specific or for specific scenarios, MCC environment is different kind of environment which need some specific language i.e. developer of MCC application must know new language syntax and programming style and a mobile operating system to support the execution of languages.

Middleware and Framework: - Opposite to specific languages framework provide the facility of easy development. In this programmer/developer is given a framework where he/she just fit the application logic. The frame work provides the full control over execution and calls the proper application specific code.

Distributed VM:- Previously discussed solution categories need the partitioning of application. This must be done by developer explicitly. There are some methods to automate this task, one of the best example of this type of automation is distributed VM. These VM handle the application for their distribution, partitioning, concurrency, synchronization and also preserve global state of application. To do these tasks different VMs run on may node of designated network.

Universal computable solution:- These are some other solution in the area of universal applications. Some system level working models for this are like Vivendi/chrome and gaia. These approaches take care of related data discovery, resource discovery, data distribution, distributed execution and automatic partitioning of applications. With the help of these solutions, in design phase the distribution scenario of application can be dealt efficiently. By this developer is not requiring having panic about these issues in lower level implementation.

Native (Nov-VM Based) MCC solution:- these are basically two type of solutions, VM based and non-VM based, which target MCC applications. In non-VM model idea is to offload some of the application's tasks on to other devices so master can be unburdening from intensive computational tasks. To accomplish this task, applications are analyzed internally and some development strategies are created which can be applied upon the executing applications.

Native (VM based) MCC solution:- When we need higher degree of distribution transparency we use VM based approaches. Clone cloud is a simulation which used a complete image of a mobile device. This image runs in a VM on a server to execute a part of application. This may also be decided that which threads to offload using a profiler at run time.

E. Application models of MCC

The mobile cloud application models are designed to achieve a particular objective, such as executing applications that have in sufficient resources, for local execution, enhancing applications performance (in terms of computation time), or achieving energy efficiency on mobile devices [2].

Application models must be adopted considered the objectives and their effect on the counterparts [2].

Mobile cloud computing applications are classified in to four categories.

- 1) *Performance based application models.*
- 2) *Energy based application models.*
- 3) *Constrained based application models.*
- 4) *Multi Objective application models.*

Performance based application models:- Main objective of this model is to improve the performance of mobile device application using cloud resources. Computation which have the intensive requirement of resource are offloaded to the high speed clod, where within very less time the computation will be performed in compare of mobile devices. These applications execute in mobile devices with improved performance with the help of cloud resources.

Energy based application models:- This model is design to save the energy consumption in mobile devices. Intensive resource computation are offloaded in cloud resources so mobile device does not waste it power in computation of such applications. By these models we reduce the computational overhead of mobile device.

Constrain based application models:- Mobile device has some resource constraint environment (e.g. smart phones) like small memory, small battery, slow processor etc. Thus these models are designed to execute the application for which resources are not enough in mobile device. In these models light weight process/applications are run on mobile devices and resource intensive applications are offloaded to cloud where these are executed. Un doughtily these models are useful to execute high resource demanding application on resource constraint devices.

Multi Objective application models:- Some time we require achieving multiple objectives from same models. In this case these types of models are used, for example if we need performance and efficiency at same time then this model is most suitable. More than one objective can be achieved with a fair tradeoff between require objective.

F. Video streaming technique over MCC

In streaming procedure, video clip data file is sent to the end individual in a (more or less) continuous flow. It is simply a strategy for shifting information such that it can be prepared as stable and ongoing flow and it is known as streaming or encoded movie that is sent across information system is known as streaming.

Streaming movie is a series of "moving images" that are sent in compacted form over the internet and shown by the audience as they appear [22].

Video streaming can be elaborated in detail by following points.

- 1) *Streaming principle*
- 2) *Video streaming architecture*
- 3) *Video streaming technique*

Streaming principle:- in case of RTVS media packets should arrive in a timely manner, since delayed packets are treated as lost. In streaming process it is essential for the media packet to reach their location in continuous and regular basis, because the network blockage can be appeared due to extreme wait of the delayed packet. Quality of information may be compromised due to this extreme wait. Beside this synchronization between customer and host server packet may be damaged and mistakes to distribute in the provided movie.

There are two type of streaming

- a) *Real-time streaming*
- b) *Pre recorded streaming*

The protocol used for streaming purpose is UDP (User data gram protocol)

Video streaming architecture:- Movie streaming scheme based on MCC is represented here

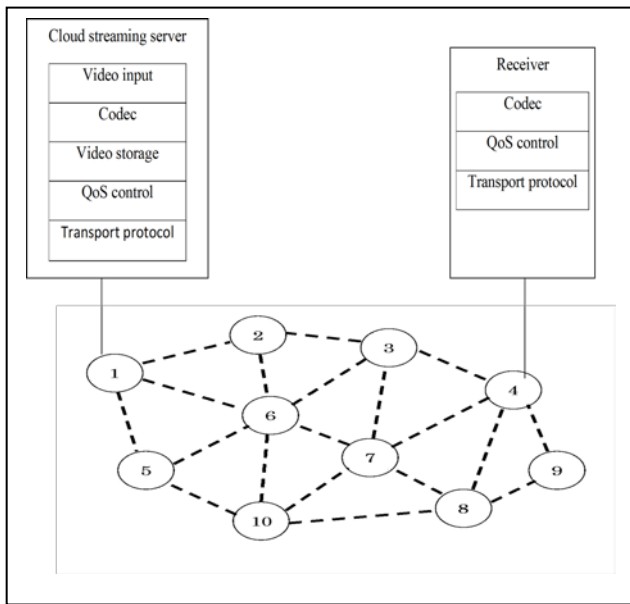


Fig. 3. Video streaming Architecture

In the given architecture there is a streaming host server implemented by cloud based source. This streaming host server is responsible for sending, retrieving and adapting video clip flow. Video clip may online protected for real-time broad casting or pre coded and stored for broad casting on demand, depend on application.

Interactive movie, mobile movie steaming, interactive online game and live broad cast require real-time encoding. But movie on demand like application require pre coding / pre encoded movie.

Video streaming technique: - There are various streaming techniques for differing mobiles devices, smart phones describe bellow [22].

- a) Progressive download
- b) HTTP live streaming

Progressive download- There are some options to customer to down load a pre coded press data file partitioned in the appropriate codec for the product to play for this frequently used method is HTTP or HTTPS. In this method play back starts just after gradual down load of preserved file. After this by the qualification player is constantly on the downloaded the rest of the material.

HTTP Live streaming: - HTTP live streaming (also known as HLS) is an HTTP based media steaming communication protocol implemented by apple Inc. as part of their QuikTime x and iPhone. Apple's HTTP live streaming protocol (HLS) is an adaptive streaming video delivery protocol for iOS devices. It utilizes the H.264 video codec, which is segmented and encapsulated in MPEG2 transport streams and .M3UP index files to deliver live and on demand video. The device automatically select the most appropriate stream given available bandwidth, CPU and platform constraints, downloads a manifest for that stream and then downloads segmented chunks to the buffer for the playback[22].

Users are provided best experience by HLS steaming, but inclusion of good IT practice and important business coordination is also a benefit of this method that can not be ignored.

Benefits of HLS

1) *The best user experience:* - Video clip server can maintain multiple version of video clip in different format. Due to this iPad user with a wifi connection are able to stream higher quality version of the video than iPhone user viewing over 3G connection

2) *Reach more viewer:* - video delivered with HTTP, support router, NAT and firewall setting in comparison of other protocol, by this more user may access your video.

3) *Save on data transfer:* in case of HLS only a few segments of video are downloaded at a time, opposed to a progressive download. Five minutes of steam video consumer only designated size, so publisher only pay for that data transfer.

4) *Secure video content:* - The HLS specification provide provision to ensure security of the stream,. This is great information for broad caster or publishers who want to stream licensed content. The entire HLS stream can be encrypted using AES-128.

G. Cloud front live streaming architecture (LSA)

Saurabh Goyal et all [22] have suggested cloud front LSA for live streaming with amzon web services allows to use the feature of adobe flash media server version 4.5, including live video streamlining where users live video is delivered by a service of HTTP requires from the player that is controlled by manifest files. Flash media server 4.5 supports two HTTP file format: HLS for iOS device and HTTP dynamic Streaming (HDS) for flash applications. User can stream high quality media using the free flash media live encoder desktop applications either for windows or for macOS [22]. Author also explore that cloud front delivery service would support on demand real-time media content streaming from flash media server 4.5. in practice this offers a new, flexible low cost contend delivery network (CDN) solution, particularly for users with relatively small or intermittent streaming delivery needs. AWS charges only for bit stored and bit transferred. There is no monthly minimum, no sign up fee or setup fee, and no ongoing costs unless you actually using the services. Author walks through the steps of setting up cloud front streaming and getting it working on user's site. There are following step for this process.

Step 1: Setup an Aws simple storage service (S3) account where content will live.

Step 2: create a bucket in S3 to store media files.

Step 3: Shift content to S3 bucket and set its permission to allow public access.

Step 4: Setup a cloud front streaming distribution that point at S3 storage bucket

Step 5: Now you are ready to stream.

Author in his architecture say cloud front usage Adobe flash media server 4.5 to stream an demand content with Adobe's Real-time messaging Protocol (RTMP). Cloud front accepts RTMP request over port 1935 and port 80.

Cloud front support the following variant of RTMP protocol:

- 1) RTMP:- Adobe's Real-time message protocol
- 2) RTMPT:- Adobe streaming tunneled over HTTP
- 3) RTMPE:- Adobe encrypted over HTTP
- 4) RTMPTE:- Adobe encrypted tunneled over HTTP

Author also suggested that to secure it, just use RTMPE protocol instead of the regular RTMP.

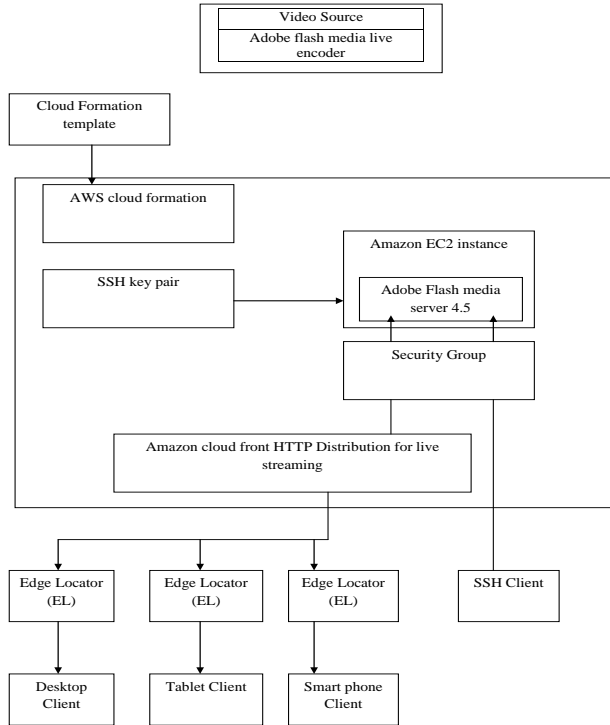


Fig. 4. Cloud front live streaming architecture

According to author statement many IT companies are using HTTP live streaming service to enhance the streaming power in their mobile domain infrastructure.

III. VIDEO STREAMING METHODS AND PARALLELIZATION

There are two basic methods of video delivery methods first video delivery via file download and second video delivery via streaming. In file down load method video down load is similar to a file we have down load, but due to its large size there are some drawback of file down load methods, down load approach requires long download method time and large storage space. If anyone wants to see the video then entire video must be downloaded first then it can be play for audience. To overcome above problem we have video delivery via streaming.

A. Video delivery via streaming

Video streaming is done to attempts to cope the problems of file downloading method of video delivery. This method

also provides significant amount of additional benefit/capability. In video streaming video is splits into parts, then these parts transmitted in succession, and receiver become enable to decode and play back the video as these parts are received, without waiting for entire video to download.

Conceptually video streaming consist of following steps:

Step 1: Partition the compressed video into packets

Step 2: start delivery of these packets

Step 3: Begin decoding and play back at the receiver while the video is still being delivered.

B. Basic problems in video streaming

There are number of basic problems those affect video streaming. In the following discussion, we focus on the case of video streaming over the internet since it is an important, concrete example that helps to illustrate these problems. Video streaming over the internet is difficult because the internet offers only best effort services. That is it provides no guarantees on bandwidth, delay jitter or less rate, specially these characteristic are unknown and dynamic. Therefore, a key goal of video streaming is to design a system for reliable delivery of the high quality video over the internet when dealing with unknown and dynamic bandwidth, delay jitter and loss rate [12].

The bandwidth which is available to use b/w two points in the internet is usually unknown and time changing/varying. To overcome the problem of congestion and suboptimal video quality which are the results of mismatch speed of transmitter and receiver, we estimate the available bandwidth and then match the transmitted video bit rate to the available bandwidth.

Accurate estimation of available bandwidth, matching of pre encoded video to estimated channel bandwidth, transmission at a rate which is fair so other concurrent flows in the internet and these entire situations in multicast scenario are the additional considerations that make the bandwidth problem very challenging.

In second problem every packet experience different en-to-end delay that may fluctuate also, at different time. This variation in end to end delay is referred to as the delay jitter. Delay jitter is a problem because the receiver must receive/decode/display frames at a constant rate, and any late frames resulting from the delay jitter can produce problems in the reconstructed video, e.g. jerk in the video [12]. Losses are third fundamental problem in video streaming. Losses those occur in video streaming are depends on particular network under construction.

C. Media streaming protocols and standards

In this section we have introduce the network protocol for media streaming over www/Internet. In addition some standards for media streaming are also discussed.

1) Protocols used for video streaming over Internet

We review some standards Internet protocol along with media delivery and control protocols

Internet protocols: TCP, UDP, IP.

Initially internet was developed to connect the heterogeneous device to share the information and functioning in which different packet switching technologies were employed. The internet protocol (IP) used to gain base line best effort network delivery to all hosts present in the network, by providing addressing, optimum/best routing, and a common format that can be interpreted/translated by everyone. On the top of the IP there are two transport protocols. First transmission control protocol (TCP) and second user data gram protocol. These protocols play vital role in transmission of packet/datagram.

Some of the difference between TCP and UDP that affects streaming applications are [12].

- TCP operate on a byte stream while UDDP is packet oriented.
- TCP guaranteed delivery via retransmissions, but because of the retransmission its delay is unbounded. UDP does not guarantee delivery, but for those packets which are delivered, their delay is predictable (i.e. one way delay) and smaller.
- TCP provides flow control and congestion control. UDP provides neither. This provides more flexibility for the application to determine the appropriate flow control and congestion control procedure.
- TCP requires a back channel for the acknowledgement. UDP does not require a back channel.

Since TCP/IP provides guaranteed delivery so web and data traffic are delivered with this protocol in which delivery is more important than delay or delay jitter. In contrast UDP/IP is used for media streaming where delay is not acceptable. Compressed media data is usually transmitted via UDP/IP despite control information is usually transmitted via TCP/IP.

Media delivery and Control protocol:

According to the specification given by IETF (internet Engineers Task Force) there are many protocols for media delivery control and description over the internet.

Media delivery:- there are two protocols designed by IETF to support streaming media, first Real-time transport protocol(RTP) and second Real-time control protocol(RTCP). RTP used for data transfer and RTCP for control messages. These protocols are not designed to enable real-time services. RTS are done by underline network only. RTP is not QoS guaranteed and reliable delivery protocol; it provides only support for applications which are time constraint. RTP make it possible by providing a standardized framework for common functionalities such as the stamps, sequence numbering, and payload specification. RTP enables detection of last process. RTCP provides QoS by providing feedback in terms of delay, jitters last packets etc. Feedback message provided by RTCP which is one in every second used by sender to adjust its operations, like, bit rate. Uses of RTP/UDP for media data and RTCP/TCP or RTCP/UDP for control message are conventional approach in media streaming.

Media control: - Real-time protocol (RTSP) and session initiation Protocol (SIP) are two protocols for media control; either can be used for media control. In video streaming RTSP is commonly used protocol. RTSP is used to establish a session and to support VCR functionalities like play, pause, seek and record. For voice over IP (VoIP) SIP is used. It is similar to RTSP, but in addition it can support user mobility and a number of additional functionalities.

Media description and announcement: - for this session description protocol is used. Information like video or audio format, codec, bit rate, duration etc are provided by SDP information description.

2) Video streaming standards and specification

Standard based media streaming systems, as specified by 3rd generation partition ship project (3GPP) for media over 3G cellular and the Internet streaming media alliance (ISMA) for streaming over the Internet, empty the following protocols [12].

Media encoding: - MPEG-4 video and audio (AMR for 3 GPP), H.263

Media transport:- RTP for data, usually over UDP/IP and RTCP for control messages, usually over UDDP/IP

Media session control:- RTSP

Media description and announcement:-SDP

Storage format for the compressed media are not specified by steaming standards but most widely used file format is MP4. MP4 file format have ability to include "hint track" that hints such as packetization boundaries, RTP header and transmission times etc. to simplify the streaming process.

D. Recent In the video streaming over Internet

During the 1919 and early phase of 2000s, attention of research focused on the design and implementation of new streaming protocols, such as the design of RTP specially for streaming media. This was the client server model for video streaming.

After client server model, peer to peer video streaming was introduced. P2P streaming protocol is based on the philosophy that end hosts, called peers. Peers works as both client and server that is not in the case of traditional client server design. Recently in video streaming HTTP video streaming over the cloud is a new paradigm. In P2P streaming protocols, it is required to download and install dedicated applications in user devices. In opposed to this HTTP video streaming allows user to stream videos directly over the web using a standard Internet web browser, without the need to download and install third party applications.

In new concept streaming server are hosted by cloud computing platform which give an ultimate experience to users, so as a result we are now in the experience of a migration to cloud computing and social media as the effective means to host and share video streams.

E. HTTP video streaming over the cloud

Though P2P has an exigency in being highly effective video delivery methods, but it is not convenient to regular

users. A stand alone application is require to be installed in user device to play back the streaming video as well as cache must have capacity to store the whole video. . In contrast HTTP video streaming using standards web browser is most convenient to watch video streams. After webRTC was developed [Bergkvist et al. 2012], these issues seemed to have been resolved; a solution to stream video over HTTP has browsers, a solution to stream video over HTTP has seen a substantial amount of industry support [3].

In this section we are presenting HTTP streaming which was developed even before P2P video streaming but now it is more widely used recently in comparison of P2P.

1) Dynamic adaptive streaming over HTTP

a) HTTP streaming from content distribution networks

Content distribution network (CDN) [Peng 2004] extensively support HTTP streaming, partially due to this, HTTP streaming gain rapid growth. In CDN method different server are deployed in different geographical location which is distributed over multiple ISPs. User can stream video from the server which are nearby their location or optimally closed to users. HTTP video streaming is a process of downloading video segments progressively from web servers via HTTP protocol. Clients who have support of HTTP protocol in their device can seek the arbitrary position in the media stream. This is done by byte range requests to the web server.

As a result, CDN can be effectively used for high quality TV content [cahil and sreenam 2004]. Adhikari et al.[2012] discovered that Netflix, the leading on demand internet video streaming provider, accounts for 29.7% of the peak downstream traffic in U.S. and it employs a mix of datacenters and CDNs for video content distribution. Watson [2011] has studied the dynamic adaptive streaming over HTTP (DASH) framework used by Netflix, which is the largest DASH based streaming provider in the world[3].

b) Research problem with DASH

DASH has been proposed to adapt the streaming rates from web server due to its best effort nature of streaming videos over the Internet. DASH was developed in 2010[MPEG 2010] and has become a new standard in 2011 [stockhammer 2011] to enable high quality streaming of media content over the internet, delivery from conventional HTTP web servers [3].

DASH has following feature

- Segmentation
- Media presentation description(MPD)
- Codec Independence
- Rate Adaptation Components
- Rate Adaptation strategies
- User quality experience

Video component is encoded and divided different segments in which initial segments contain the required information for initializing the media decoder, as well as the media segments containing the video data.

MDP tell about the segments from a video presentation. MDP used to get required segment for smooth playback and adjust bitrates or other attributes according to bandwidth estimates according to client request.

DASH is skeptical and its main container is the MP4 and MPEG-TS. It also allows smooth and flawless adoption of the upcoming improved HEVC video codec (i.e. H265).

Rate adaption component are leaved either for client side or server side implementation. There are different strategies for rate adaption to determine that how different version of segments are received by user to achieve the objective, including fairness, high quality and streaming stability. In terms of user quality experience, though rate adaption is highly correlated with the user's video quality experience in DASH streaming such as correlation may require a more detail study.

Cranley et al [2006] demonstrated the dynamic nature of user perception with adaptive video streaming. In the context of DASH, mock et al[2012] have studied the user experience and observe that users prefer a gradual quality change between the best and worst quality levels, instead of about switching. To better guide the design of rate of adaption strategies, a good metric for evaluation the user experience is still an open area of research[Song et al. 2011] [3].

2) Video streaming from the cloud

Major content providers are using HTTP streaming now days. Thus more and more video streaming is done via HTTP streaming. Due to this, sharing of user generated content are increased by many fold which changed the video streaming landscape. The new landscape provides the platform to the users to generate and upload video content dynamically in to the server.

The design of centralized data centers had increased the popularity of cloud computing since 2008. This design is exactly the opposite of peer-to-peer system design. Due to its "pay as you go" pricing model cloud computing platform for video streaming service are getting popular now. There are some examples like Netflix one of the leading video streaming service provider has been reported to restore the cloud services. This paradigm shift generated a demand of research in the area of cloud based video streaming networks, basically topologies in data centers and algorithm of streaming. There are many benefits of cloud based streaming some of those are inherited from cloud infrastructure like reliability, elasticity, cost effectiveness etc, and others are provided by service provider (e.g. Netflix) which include pay by GB for bandwidth resource which leads to long term cost saving. Cloud based video streaming service may also be able to handle busty demand very well.

Beside these benefits cloud based streaming have some shortcoming due to less research and implementation possibilities. There are number of theoretical and practical issue to be addressed while transferring video streaming services to the cloud. The very first and important problem is the heterogeneity and lease pricing of servers. The billing cycle cannot be arbitrary short, since service provider (e.g. Amazon EC2) provides VM on lease by a fix amount of time

like hourly basis. Second when VM is required it must be ordered in advance since it take time (few minutes) to instantiate it. So VoD must be predicted accurately by VoD providers.

A generic framework which provide the facility to migrate existing live media streaming service to a cloud based solution [Wang et al. 2012 a] is shown in fig 5. In this frame work there are two layer, layer1 cloud layer and layer2 user layer, both layer used to take and provide lease and adjusts cloud servers to accommodate temporal and spatial dynamic of user demands.

In the comparison of the Internet a decade ago, networked services in web 2.0 era focus more on the user experience, user participation, and interaction with rich media. The users are now actively engaged to be part of a social ecosystem, rather than passively receiving video stream [3]. If we consider streaming services, one most dominating example which reflects this change is youtube, established in 2005.

You tube is now providing services of 4 billion view or more a day, with most of the video data is generated by users. Now researchers are required to take attention about the huge amount of video content over Internet and also its streaming process.

3) Real-time Adaptive Algorithm for video streaming over mobile network

Protocols for streaming video over internet have existed for decades, and a large number of different protocols have been used in various degrees [11]. In the evolution of video streaming protocols adaptive bit rate streaming technique is most popular today.

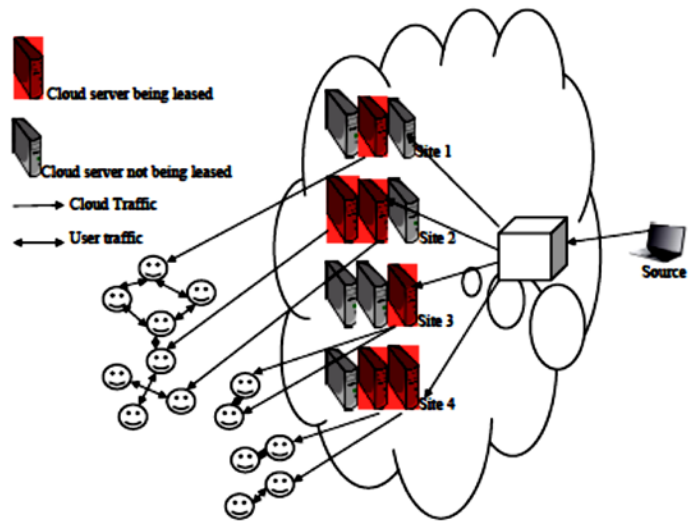


Fig. 5. A frame work for cloud video streaming

A brief history of video streaming is given in table 1.

The protocol used in adaptive streaming is HTTP. HTTP operates on top of TCP, which handle all the data [17].

[Min Xing et al] suggested the optimal video streaming process with multiple links is formulated as a markov decision process (MDP). In this work they have design a reward function to consider the quality of services (QoS) requirements for video traffic, such as the startup latency, playback fluency, average playback quality, play back smoothness and wireless service cost [15]. [Min Xing et al] has proposed an adaptive best action search algorithm to obtain a suboptimal solution, to solve MDP in real-time [15].

TABLE I. THE EVOLUTION OF VIDEO STREAMING TECHNIQUE

s.No.	Datagram Streaming	Progressive download Streaming	Adaptive HTTP Streaming
1	Uses UDP for transport <ul style="list-style-type: none"> Expensive for CDNs Often problem with firewall 	HTT for transport <ul style="list-style-type: none"> Uses existing infrastructure to reduce the cost Firewall friendly 	HTTP for transport <ul style="list-style-type: none"> Uses existing web infrastructure to reduce the cost Firewall friendly
2	Send rate close to stream's bit rate	Send rate is unlimited	Receiver limits its download rate when its buffer is full
3	Packet delivery is not guaranteed	Receiver input buffer is unlimited	Uses low bit rate when buffer is empty- > fast start
4	Fast start and little buffering	Play out interruption common	Bit rate adaption reduces play out interruption
5			
6	e.g. Protocol: MMS,RTMP,RTP,PNA	Used in most flash-based video sites	e.g. Protocol: smooth HLS, DASH, HDS
	Complexity of implementation is high	Complexity of implementation low	complexity of implementation medium

IV. POSSIBLE IMPROVEMENT IN RTVS OVER MCC

The emerging uses of smart phones have clearly indicated an industry trend to seamlessly integrate cloud services with mobile applications. By transferring a small amount of data from mobile device to cloud, user may be provided better experience after taking the advantage of computation power of cloud. A large number of research questions remain open in the area of mobile cloud computing like optimizing the use of cloud computing resources to scale up to millions of mobile users [27].

Here we are listing some possible improvement/findings/future work by which better RTVS can be provided to mobile users

- 1) Better utilization of inter data center network by fairly allocating and slicing network bandwidth using network virtualization [27].
- 2) Implementation of network virtualization at the application layer without changing traditional transport protocol [27].
- 3) Standardization of mobile cloud execution platform for the easier computation offloading [2].
- 4) New policies synchronization between smart phone and smart phone clone in the cloud [2].
- 5) Development of security mechanism to secure the clone from illegal access and protect the smart phone user from the malicious VMs executing in the cloud [2].
- 6) Automatic path prediction and path prediction that fails in video streaming over MCC [11].
- 7) Investigation of better load allocation between several links with finer granularity [15].
- 8) Improving the bandwidth estimation accuracy by the consideration of size of the video segment for variable bit rate (VBR) video [15].
- 9) Prediction of future bandwidth in RTVS over MCC environment [15].
- 10) Adjustment of mechanism of pre fetch window size to get optimum performance of online pre fetch algorithm in Adaptive video streaming over HTTP [13].
- 11) Implementation of video proxy in the ISP to mitigate the traffic redundancy of mobile video player in RTVS over MCC environment [1].
- 12) Observing the behavior of adaptive video player by large scale study of traffic redundancy in adaptive video players in RTVS over MCC environment [1].
- 13) Understanding and standardization of the economics between ISPs and video providers to provide good quality video for mobile/end users [1].
- 14) Virtualization and immigrate task from terminal to cloud to achieve better result /QoS in RTVS over MCC environment [10].
- 15) Standardization of MCC services billing for RTVS [30].
- 16) Optimization of performance metrics of MCC data center by non-linear function and multi-dimensional optimization algorithm [20].

17) Investigating the issue with flickering effect on link aware reconfigurable point to point video streaming for mobile devices [19].

18) Utilization of weight based optimized routing for better RTVS experience over MCC environment [5].

19) Development of new system architecture and standards that seamlessly integrate MC, IOT (Internet of things) and protocols that facilitate big video data streaming from IOT to MC [18].

20) Employing optimal multiple stopping method to get better RTVS over MCC environment [23].

21) Integration of MCC application development with agile development methodologies [8].

22) Utilization of QoS control mechanisms to balance between the rapid response of the abnormal network and the avoidance of the rate oscillation in RTVS [6].

23) Providing energy efficient green mobile cloud computing using the cloud Exptool²³ [7].

24) An efficient algorithm to determine the redistribution of system resources in a slice after a single node fails in RTVS over MCC environment [16].

25) Real-time scheduling of resources (surrogate node) in RTVS over MCC environment [31].

26) New scheduling scheme for energy efficient cloud offloading for multi-core mobile devices, while considering downloading the cloud execution output in the model [29].

27) Scheduling of the surrogate node and finding the availability of surrogate nodes is a NP complete problem, to compute the schedule, a computational grid of desktop can be used instead of having a large machine [25].

V. SOCIAL AND LEGAL ISSUE OF RTVS

Mobile live streaming brings the new way of showing video content to the users. In this field two major apps are launched recently meerkat²² in feb 2015 and periscope in march 26 2015. This time these two are the main competitor in the market of mobile live streaming (RTVS). The person who is having smart phone, the Internet and an app can now broadcast live video to the world. This is great for users but is a nightmare for content right holders.

RTVs can be used for business, education, healthcare and training and development. So one side of this technology shows a very positive impact on society, but in another side it has a question of different intellectual property right issue, as well as copy right and cyber security concern. These live streaming apps lets users simply point their smart phones at whatever is happening in front of them, whether they own the right or not, and broadcast it to a potential audience of hundreds of thousands now , millions in future.

Beside many benefits of RTVS, concern has been raised in particular over large right holder such as entertainment studios, sport broadcaster etc using periscope for users who film content direct from TV or events via their smart phones.

In this violation of IPR and copyright matter one of the most famous examples is Mayweather-Pacquiao boxing fight in 2015.

HBO saw an estimated 10,000 people watching this fight via periscope without paying any amount of money. This raises the alarm to have some standards and rules for RTVS among users, to protect the user's right, social benefits and IPR with CR issue.

VI. CONCLUSION

After the exhaustive study of different research papers and related article we found that real-time video streaming in mobile devices is now going to be next challenging field for the consideration of new research.

This paper tells that use of mobile cloud computing environment for the RTVS may provides better RTVS experience. Beside this in this study, paper also presented that there are many architectural and QoS issue in this area. E.g. standardization of streaming process for MCC environment, standardization of billing process of MCC resources, scheduling of resources for video segments in different surrogate nodes, prediction of required bandwidth for streaming, and IPR and Copyright issue as well as user's rights, etc.

This paper listed many issues along with possible future improvement and work in the field of RTVS over MCC environment. In future work any of the above explained possible improvement in the area of RTVS process over MCC environment may be implemented.

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Applications of Some Topological Near Open Sets to Knowledge Discovery

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Abstract—In this paper, we use some topological near open sets to introduce the rough set concepts such as near open lower and near open upper approximations. Also, we study the concept of near open, rough set and some of their basic properties. We will study the comparison among near open concepts and rough set concepts. We also studied the effect of this concept to motivate the knowledge discovery processing.

Keywords—Topological spaces; Rough sets; Knowledge discovery; open sets; Accuracy measure

I. INTRODUCTION

Information technology is huge and need more accurate measures to discover their valuable knowledge. An important field of this technology is the information discovery via available information. Rough set theory and their topological generalizations using some topological near open sets [1, 2, 4, 5, 7, 12, 13, 15, 17, 18, 19, 20] are a recent, an accurate and applicable approach for reasoning about data. Many researchers generalized rough sets, but the concept of a topological rough set given by Wiweger [16] in 1989 is the basic starting point. This generalization is basically starting with a topological space and defined the approximation via the interior and the closure operators on topological spaces. The concept of β -open set was introduced in 1983 by M. E. Abd El-Monsef and others [3].

In this paper, we introduce and investigate the concept of near open approximation space. These spaces help to get a new classification for the universe. Also, we investigate the concept of near open lower and near open upper approximations. We study near open, rough sets, the comparison between this concept and rough sets is also studied. Also, we give some counter examples.

II. ROUGH SET BASIC CONCEPTS

Classical rough set theory has come from the need to represent subsets of a universe in terms of equivalence classes of a partition of that universe. The partition characterizes a topological space, called approximation space $K = (X, R)$, where X is a set called the universe and R is an equivalence relation [8, 14]. The equivalence classes of R are also known as the granules, elementary sets or blocks; we will use $R_x \subseteq X$ to denote the equivalence class containing $x \in X$. In the approximation space, we consider two sets, namely

$\overline{R}(A) = \{x \in X : R_x \cap A \neq \emptyset\}$ and
 $\underline{R}(A) = \{x \in X : R_x \subseteq A\}$ that is called the lower and the upper approximation of $A \subseteq X$ respectively. Also, let $POS_R(A) = \underline{R}(A)$ denote the positive region of A , $NEG_R(A) = X - \overline{R}(A)$ denotes the negative region of A and $BN_R(A) = \overline{R}(A) - \underline{R}(A)$ denote the boundary region of X .

The degree of completeness can also be characterized by the accuracy measure, in which $|R|$ represents the cardinality of set R as follows:

$$\alpha_R(A) = \frac{|\underline{R}(A)|}{|\overline{R}(A)|} \quad \text{where } A \neq \emptyset.$$

Accuracy measures try to express the degree of completeness of knowledge. $\alpha_R(A)$ is able to imprisonment, how large the boundary region of the data sets is; however, we cannot easily capture the structure of the knowledge. A fundamental advantage of rough set theory is the ability to handle a category that cannot be sharply defined given a knowledge base. Characteristics of the potential data sets can be measured through the rough sets framework. We can measure inexactness and express topological characterization of imprecision with:

- (1) If $\underline{R}(A) \neq \emptyset$ and $\overline{R}(A) \neq X$, then A is roughly R -definable.
- (2) If $\underline{R}(A) = \emptyset$ and $\overline{R}(A) \neq X$, then A is internally R -undefinable
- (3) If $\underline{R}(A) \neq \emptyset$ and $\overline{R}(A) = X$, then A is externally R -undefinable.
- (4) If $\underline{R}(A) = \emptyset$ and $\overline{R}(A) = X$, then A is totally R -undefinable.

We denote the set of all, roughly R -definable (resp. Internally R -undefinable, externally R -undefinable and totally R -undefinable) sets by $RD(X)$ (resp. $IUD(X)$, $EUD(X)$ and $TUD(X)$).

With $\alpha_R(A)$ and classifications above, we can characterize rough sets by their size of the boundary region.

A topological space [6] is a pair (X, τ) consisting of a set X and family τ of subsets of X satisfying the following conditions:

- (1) $\phi, X \in \tau$
- (2) τ is closed under arbitrary union.
- (3) τ is closed under finite intersection.

The pair (X, τ) is called a topological space, the elements of X are called points of the space, the subsets of X belonging to τ are called open set in the space, and the complement of the subsets of X belonging to τ are called closed set in the space; the family τ of open subsets of X is also called a topology for X .

$\bar{A} = \bigcap \{F \subseteq X : A \subseteq F \text{ and } F \text{ is closed}\}$ is called τ -closure of a subset $A \subseteq X$.

Evidently, \bar{A} is the smallest closed subset of X which contains A . Note that A is closed iff $A = \bar{A}$.

$A^\circ = \bigcup \{G \subseteq X : G \subseteq A \text{ and } G \text{ is open}\}$ is called the τ -interior of a subset $A \subseteq X$.

Evidently, A° is the union of all open subsets of X which containing in A . Note that A is open iff $A = A^\circ$. And $b(A) = \bar{A} - A^\circ$ is called the τ -boundary of a subset $A \subseteq X$.

Let A be a subset of a topological space (X, τ) . Let \bar{A} , A° and $b(A)$ be closure, interior, and boundary of A respectively. A is exact if $b(A) = \phi$, otherwise A is rough.

It is clear A is exact iff $\bar{A} = A^\circ$. In Pawlak space a subset $A \subseteq X$ has two possibilities rough or exact.

Definition 2.1. [3] A subset A of a topological space (X, τ) is called β -open

$$\text{if } A \subseteq cl(int(cl(A))).$$

The complement of β -open set is β -closed set. We denote that the set of all β -open and β -closed sets by $\beta O(X)$ and $\beta C(X)$ respectively.

Remark 2.1. For any topological space (X, τ) . We have $\tau \subseteq \beta O(X)$.

III. TOPOLOGICAL NEAR OPEN ROUGH CLASSIFICATION

In this section, we introduce and investigate the concept of near open approximation space. Also, we introduce the

concepts of near open lower approximation and near open upper approximation and study their properties.

Definition 3.1. Let X be a finite non-empty universe. The pair (X, R_β) is called a near open approximation space where R_β is a general relation used to get a subbase for a topology τ on X and a class of β -open sets $\beta O(X)$.

Remark 3.1. In Definition 3.1, we use the symbol R_β to avoid confusion with R which refers to an equivalence relation.

Example 3.1. Let $U = \{a, b, c, d, e, f, g\}$ be a universe of multi valued information system of seven patients (A, B and C are conditional attributes and D is the decision attribute) as in Table 1, the relation R defined on patients by $R = \{(a, a), (a, c), (a, d), (b, b), (b, d), (c, a), (c, b), (c, d), (d, a)\}$, thus $aR = \{a, c, d\}$, $bR = \{b, d\}$, $cR = \{a, b, d\}$ and $dR = \{a\}$. Then the topology associated with this relation is $\tau = \{U, \phi, \{a\}, \{d\}, \{a, d\}, \{b, d\}, \{a, b, d\}, \{a, c, d\}\}$ and $\beta O(X) = \{X, \phi, \{a\}, \{d\}, \{a, c\}, \{a, d\}, \{b, d\}, \{c, d\}, \{a, b, d\}, \{a, c, d\}, \{b, c, d\}\}$. So (U, R_β) is a near open approximation space.

TABLE I. MULTIVALUED INFORMATION SYSTEM

U	A	B	C	D
a	{A2}	{B1,B2,B4}	{c1}	{d3}
b	{A1,A2}	{B1,B2}	{c1,c3}	{d3}
c	{A2}	{B1,B2}	{c1}	{d3}
d	{A1}	{B1,B2,B4}	{c4}	{d1}
e	{A1}	{B5}	{c1,c2}	{d2}
f	{A1}	{B1,B2}	{c1}	{d3}
g	{A1}	{B1,B3,B4}	{c1,c3}	{d3}

Example 3.2. Let $X = \{a, b, c\}$ be a subset of the universe U , and a relation R defined on X by $aR = \{a, b\}$, $bR = \{b\}$, $cR = \{a, b\}$. Then the subtopology associated with this relation is $\tau = \{X, \phi, \{b\}, \{a, b\}\}$ and $\beta O(X) = \{X, \phi, \{b\}, \{a, b\}, \{b, c\}\}$. So (X, R_β) is a subnear open approximation space.

Definition 3.2. Let (X, R_β) be a near open approximation space near open lower (resp. near open upper) approximation of any non-empty subset A of X is defined as:

$$\begin{aligned} \underline{R}_\beta(A) &= \bigcup \{G \in \beta O(X) : G \subseteq A\} \\ \overline{R}_\beta(A) &= \bigcap \{F \in \beta C(X) : F \supseteq A\} \end{aligned}$$

Definition 3.3. Let (X, R_β) be a near open approximation space. From the relation

$int(A) \subseteq \beta int(A) \subseteq A \subseteq \beta cl(A) \subseteq cl(A)$, for any $A \subseteq X$. The Universe X can be divided into 12 regions with respect to any $A \subseteq X$ as shown in Figure 1.

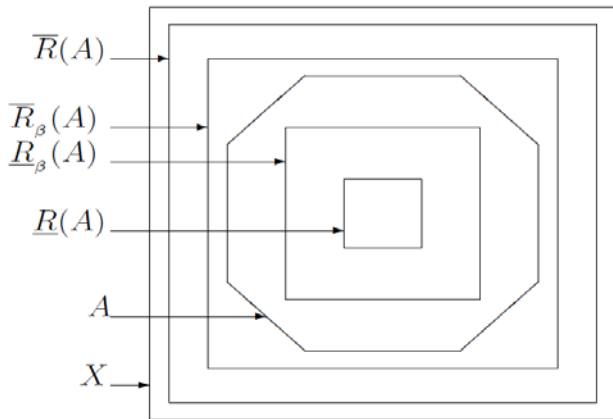


Fig. 1. Regions of the universe

Remark 3.2. The study of near open approximation spaces is a generalization for the study of approximation spaces. Because of the elements of the regions $[R_\beta(A) - R(A)]$ will be defined well in A , while this point was undefinable in Pawlak's approximation spaces. Also, the elements of the region $[\bar{R}(A) - \bar{R}_\beta(A)]$ do not belong to A , while these elements was not well defined in Pawlak's approximation spaces.

In our study, reduce the boundary of A , in Pawlak's approximation space by near open boundary of A . Also, we extend exterior of A which contains the elements did not belong to A by near open exterior of A .

Proposition 3.1. For any near open approximation space (X, R_β) , the following are hold of any $A \subseteq X$:

$$(1) b(A) = \underline{Edg}(A) \cup \overline{Edg}(A), \underline{Edg}(A) = A - \underline{R}(A), \overline{Edg}(A) = \bar{R}(A) - A$$

$$(2) \beta b(A) = \beta \underline{Edg}(A) \cup \beta \overline{Edg}(A), \beta \underline{Edg}(A) = A - \underline{R}_\beta(A), \beta \overline{Edg}(A) = \bar{R}_\beta(A) - A$$

Proof (2). It follows from

$$\begin{aligned} \beta b(A) &= \bar{R}_\beta(A) - \underline{R}_\beta(A) \\ &= (\bar{R}_\beta(A) - A) \cup (A - \underline{R}_\beta(A)) \\ &= \beta \overline{Edg}(A) \cup \beta \underline{Edg}(A) \end{aligned}$$

Proposition 3.2. For any near open approximation space (X, R_β) , the following are hold of any $A \subseteq X$:

$$(1) \bar{R}(A) - \underline{R}_\beta(A) = \overline{Edg}(A) \cup \beta \underline{Edg}(A)$$

$$(2) \bar{R}_\beta(A) - \underline{R}(A) = \beta \overline{Edg}(A) \cup \underline{Edg}(A)$$

Proof. Obvious.

Proposition 3.3. For any near open approximation space (X, R_β) . The following are hold to any $A \subseteq X$:

$$(1) \underline{Edg}(A) = \beta \overline{Edg}(A) \cup (\underline{R}_\beta(A) - \underline{R}(A))$$

$$(2) \overline{Edg}(A) = \beta \underline{Edg}(A) \cup (\bar{R}(A) - \bar{R}_\beta(A))$$

Proof. Obvious.

Definition 3.4. Let (X, R_β) be a near open approximation space and $A \subseteq X$. Then there are memberships $\underline{\in}$, $\bar{\in}$, $\underline{\in}_\beta$ and $\bar{\in}_\beta$, say, near open strong and near open weak memberships respectively which defined by:

$$(1) x \underline{\in} A \text{ iff } x \in \underline{R}(A)$$

$$(2) x \bar{\in} A \text{ iff } x \in \bar{R}(A)$$

$$(3) x \underline{\in}_\beta A \text{ iff } x \in \underline{R}_\beta(A)$$

$$(4) x \bar{\in}_\beta A \text{ iff } x \in \bar{R}_\beta(A)$$

Remark 3.3. According to Definition 3.4, near open lower and near open upper approximations of a set $A \subseteq X$ can be written as

$$(1) \underline{R}_\beta(A) = \{x \in A : x \underline{\in}_\beta A\}$$

$$(2) \bar{R}_\beta(A) = \{x \in A : x \bar{\in}_\beta A\}$$

Remark 3.4. Let (X, R_β) be a near open approximation space and $A \subseteq X$. Then

$$(1) x \underline{\in} A \Rightarrow x \underline{\in}_\beta A$$

$$(2) x \bar{\in}_\beta A \Rightarrow x \bar{\in} A$$

The converse of Remark 3.4 may not be true in general as seen in the following example.

Example 3.3. Let $X = \{a, b, c, d\}$ be a universe and a relation R defined by $R = \{(a, a), (d, c), (d, d), (c, a), (c, d), (c, c)\}$, thus $aR = \{a\}$, $bR = \phi$, $cR = \{a, c, d\}$ and $dR = \{c, d\}$. Then the topology associated with this relation is $\tau = \{X, \phi, \{a\}, \{c, d\}, \{a, c, d\}\}$. So (X, R_β) is a β -approximation space. Let, $A = \{b, c, d\}$, we have $b \underline{\in}_\beta A$ but $b \underline{\in} A$. Also, let $B = \{c\}$. We have $d \bar{\in} B$ but, $d \bar{\in}_\beta B$.

We can characterize the degree of completeness by a new tool named β -accuracy measure defined as follows:

$$\alpha_{R_\beta}(A) = \frac{|\underline{R}_\beta(A)|}{|\overline{R}_\beta(A)|} \text{ where } A \neq \phi$$

Example 3.4. According to Example 3.1, we can deduce Table 2 below that show the degree of accuracy measure $\alpha_\beta(A)$ and β -accuracy measure $\alpha_{R_\beta}(A)$ for some proper subsets of the universe of discourse.

TABLE II. ACCURACY MEASURES

$A \subseteq X$	$\alpha_\beta(A)$	$\alpha_{R_\beta}(A)$
{a}	1/2	1
{a,c}	1/2	1
{b,d}	1/3	1
{b,c,d}	2/3	1

We see that the degree of exactness of the set $A = \{a\}$ by using accuracy measure equal to 50% and by using near open accuracy measure equal to 100%. Consequently near open accuracy measure is the accurate measure in this case.

We investigate near open, rough equality and near open rough inclusion based on rough equality and inclusion which introduced by Pawlak and Novotny in [10, 11].

Definition 3.5. Let (X, R_β) be a near open approximation space, $A, B \subseteq X$. Then we say that A and B are:

- (i) Near open roughly bottom equal if $\underline{R}_\beta(A) = \underline{R}_\beta(B)$
- (ii) Near open roughly top equal if $\overline{R}_\beta(A) = \overline{R}_\beta(B)$
- (iii) Near open roughly equal ($A \approx_\beta B$) if

$$\underline{R}_\beta(A) = \underline{R}_\beta(B) \text{ and } \overline{R}_\beta(A) = \overline{R}_\beta(B)$$

Example 3.5. According to Example 3.1, we have the sets $\{a, c\}$, $\{a, b, c\}$ are near open roughly bottom equal and $\{c, d\}$, $\{b, c, d\}$ are near open roughly top equal.

One can easily show that \approx_β is an equivalence relation on $P(X)$, hence the pair $(P(X), \approx_\beta)$ is an approximation space. Also, this relation \approx_β is called near open, rough equality of the near open approximation space (X, R_β) .

Definition 3.6. Let (X, R_β) be a near open approximation space. We define the equivalence relation E_β on the set $P(X)$ given by the condition:

$$(A, B) \in E_\beta \text{ if } \beta - \text{int}(A) = \beta - \text{int}(B) \text{ and } \beta - \text{cl}(A) = \beta - \text{cl}(B).$$

The equivalence relation E_β is precisely the same of \approx_β ,

since $\underline{R}_\beta(A) = \beta \text{int}(A)$ and $\overline{R}_\beta(A) = \beta \text{cl}(A)$.

Remark 3.5. For any subset A of X , the equivalence class of the relation $(\approx_\beta \text{ or } E_\beta)$ containing A is denoted by $[A]_{\approx_\beta}$ or $[A]_{E_\beta}$. Thus,

$$[A]_{\approx_\beta} = \{D \subseteq X : \underline{R}_\beta(D) = \underline{R}_\beta(A) \text{ and } \overline{R}_\beta(D) = \overline{R}_\beta(A)\}.$$

We denote by $R_\beta(X)$ of the family of near open rough classes of a near open approximation space (X, R_β) .

Definition 3.7. Let (X, R_β) be a near open approximation space, $A, B \subseteq X$. Then we say that

- (i) A is near open roughly bottom included in B if $\underline{R}_\beta(A) \subseteq \underline{R}_\beta(B)$
- (ii) A is near open roughly top included in B if $\overline{R}_\beta(A) \subseteq \overline{R}_\beta(B)$
- (iii) A is near open roughly included in B if $\underline{R}_\beta(A) \subseteq \underline{R}_\beta(B)$ and $\overline{R}_\beta(A) \subseteq \overline{R}_\beta(B)$

Example 3.6. According to Example 3.1, we have $\{a, c\}$ is near open roughly bottom included in $\{a, b, c\}$. Also, $\{c, d\}$ is near open roughly top included in $\{b, c, d\}$.

IV. PROPERTIES OF NEAR OPEN, ROUGH SETS

In this section, we introduced a new concept of near open rough set.

Definition 4.1. For any near open approximation space (X, R_β) , a subset A of X is called:

- (1) R_β -definable (β -exact) if $\overline{R}_\beta(A) = \underline{R}_\beta(A)$ or $\beta b(A) = \phi$
- (2) near open rough if $\overline{R}_\beta(A) \neq \underline{R}_\beta(A)$ or $\beta b(A) \neq \phi$

Example 4.1. Let (X, R_β) be a near open approximation space as in Example 3.3. We have the set $\{a, c\}$ is near open exact while $\{b\}$ is near open rough set.

Proposition 4.1. Let (X, R_β) be a near open approximation space. Then:

- (1) Every exact set in X is near open exact.
 - (2) Every near open rough set in X is rough.
- Proof. Obvious.

The converse of all parts of Proposition 4.2 may not be true in general as seen in the following example.

Example 4.2. Let (X, R_β) be a near open approximation space as in Example 3.3. Then the set $\{a, c\}$ is near open exact, but not exact and the set $\{a, b, d\}$ is rough but not near open rough.

Remark 4.1. The intersection of two near open exact sets need not be near open exact set.

Example 4.3. Let (X, R_β) be a near open approximation space as in Example 3.3. We have $\{b, c\}$ and $\{a, b\}$ are two near open exact sets but $\{b, c\} \cap \{a, b\} = \{b\}$ does not near open exact.

Definition 4.2. Let (X, R_β) be a near open approximation space, the set $A \subseteq X$ is called:

- (1) Roughly R_β -definable, if $\underline{R}_\beta(A) \neq \phi$ and $\overline{R}_\beta(A) \neq X$.
- (2) Internally R_β -undefinable, if $\underline{R}_\beta(A) = \phi$ and $\overline{R}_\beta(A) \neq X$.
- (3) Externally R_β -undefinable, if $\underline{R}_\beta(A) \neq \phi$ and $\overline{R}_\beta(A) = X$.
- (4) Totally R_β -undefinable, if $\underline{R}_\beta(A) = \phi$ and $\overline{R}_\beta(A) = X$.

We denote the set of all, roughly R_β -definable (resp. Internally R_β -undefinable, externally R_β -undefinable and totally R_β -undefinable) sets by $\beta RD(X)$ (resp. $\beta IUD(X)$, $\beta EUD(X)$ and $\beta TUD(X)$).

Remark 4.2. For any near open approximation space (X, R_β) . The following are held:

- (1) $\beta RD(X) \supseteq RD(X)$
- (2) $\beta IUD(X) \subseteq IUD(X)$
- (3) $\beta EUD(X) \subseteq EUD(X)$
- (4) $\beta TUD(X) \subseteq TUD(X)$

Example 4.4. According to Example 3.3, we have the set $\{a, b\} \in \beta RD(X)$ but $\{a, b\} \notin RD(X)$. The set $\{b, d\} \in IUD(X)$ but $\{b, d\} \notin \beta IUD(X)$. Also, the set $\{a, d\} \in EUD(X)$ but $\{a, d\} \notin \beta EUD(X)$.

Lemma 4.1. For any near open approximation space (X, R_β) and for all $x, y \in X$, the condition $x \in \overline{R}_\beta(\{y\})$ and $y \in \overline{R}_\beta(\{x\})$ implies $\overline{R}_\beta(\{x\}) = \overline{R}_\beta(\{y\})$.

Proof. By definition of near open upper approximation of a

set is a β -closure of this set, and since $\beta cl(\{y\})$ is a β -closed set containing x (by the condition) while $\beta cl(\{x\})$ is the smallest β -closed set containing x , thus $\beta cl(\{x\}) \subseteq \beta cl(\{y\})$. Hence $\overline{R}_\beta(\{x\}) \subseteq \overline{R}_\beta(\{y\})$. The opposite inclusion follows by symmetry $\beta cl(\{y\}) \subseteq \beta cl(\{x\})$. Hence $\overline{R}_\beta(\{y\}) \subseteq \overline{R}_\beta(\{x\})$, which complete the proof.

Lemma 4.2. Let (X, R_β) be a near open approximation space, which every β -open subset A of X is β -closed. Then $y \in \overline{R}_\beta(\{x\})$ implies $x \in \overline{R}_\beta(\{y\})$ for all $x, y \in X$.

Proof. If $x \notin \overline{R}_\beta(\{y\})$, then there exists a β -open set G containing x such that $G \cap \{y\} = \phi$ which implies that $\{y\} \subseteq (X \setminus G)$, but $(X \setminus G)$ is a β -closed set and also is a β -open set does not contain x , thus $(X \setminus G) \cap \{x\} = \phi$. Hence $y \notin \overline{R}_\beta(\{x\})$.

Proposition 4.2. Let (X, R_β) be a near open approximation space, and every β -open subset A of X is β -closed. Then the family of sets $\{\overline{R}_\beta(\{x\}) : x \in A\}$ is a partition of the set X .

Proof. If $x, y, z \in A$ and $z \in \overline{R}_\beta(\{x\}) \cap \overline{R}_\beta(\{y\})$, then $z \in \overline{R}_\beta(\{x\})$ and $z \in \overline{R}_\beta(\{y\})$. Thus by Lemma 4.2, $x \in \overline{R}_\beta(\{z\})$ and $y \in \overline{R}_\beta(\{z\})$ and by Lemma 4.1 we have $\overline{R}_\beta(\{x\}) = \overline{R}_\beta(\{z\})$ and $\overline{R}_\beta(\{y\}) = \overline{R}_\beta(\{z\})$. Therefore $\overline{R}_\beta(\{x\}) = \overline{R}_\beta(\{y\}) = \overline{R}_\beta(\{z\})$. Hence, either $\overline{R}_\beta(\{x\}) = \overline{R}_\beta(\{y\})$ or $\overline{R}_\beta(\{x\}) \cap \overline{R}_\beta(\{y\}) = \phi$.

V. PROPERTIES OF NEAR OPEN APPROXIMATION SPACES

The purpose of this section is to investigate some properties of near open approximation spaces.

Proposition 5.1. Let (X, R_β) be a near open approximation space and $A, B \subseteq X$. Then

- (i) $\underline{R}_\beta(A) \subseteq A \subseteq \overline{R}_\beta(A)$
- (ii) $\underline{R}_\beta(\phi) = \overline{R}_\beta(\phi) = \phi$, $\underline{R}_\beta(X) = \overline{R}_\beta(X) = X$.
- (iii) If $A \subseteq B$ then $\underline{R}_\beta(A) \subseteq \underline{R}_\beta(B)$ and $\overline{R}_\beta(A) \subseteq \overline{R}_\beta(B)$

Proof. (i) Let $x \in \underline{R}_\beta(A)$ which mean that

$x \in \bigcup\{G \in \beta O(X), G \subseteq A\}$. Then there exists $G_0 \in \beta O(X)$ such that $x \in G_0 \subseteq A$. Thus $x \in A$. Hence $\underline{R}_\beta(A) \subseteq A$. Also, let $x \in X$ and by definition of $\overline{R}_\beta(A) = \bigcap\{F \in \beta C(X), A \subseteq F\}$, then $x \in F$ for all $F \in \beta C(X)$. Hence $A \subseteq \overline{R}_\beta(A)$.

(ii) Follows directly.

(iii) Let $x \in \underline{R}_\beta(A)$, by definition of near open lower approximation of A , we have $x \in \bigcup\{G \in \beta O(X), G \subseteq A\}$ but $A \subseteq B$, thus $G \subseteq B$ and $x \in G$, then $x \in \underline{R}_\beta(B)$. Also, let $x \notin \overline{R}_\beta(B)$ this means that $x \notin \bigcap\{F \in \beta C(X), B \subseteq F\}$ then, there exists $F \in \beta C(X)$, $B \subseteq F$ and $x \notin F$ which means that, there exists $F \in \beta C(X)$, $A \subseteq B \subseteq F$ and $x \notin F$ which implies $x \notin \bigcap\{F \in \beta C(X), A \subseteq F\}$, thus $x \notin \overline{R}_\beta(A)$. Therefore $\overline{R}_\beta(A) \subseteq \overline{R}_\beta(B)$.

Proposition 5.2. Let (X, R_β) be a near open approximation space and $A, B \subseteq X$. Then

- (i) $\underline{R}_\beta(X \setminus A) = X \setminus \overline{R}_\beta(A)$
- (ii) $\overline{R}_\beta(X \setminus A) = X \setminus \underline{R}_\beta(A)$
- (iii) $\underline{R}_\beta(\underline{R}_\beta(A)) = \underline{R}_\beta(A)$
- (iv) $\overline{R}_\beta(\overline{R}_\beta(A)) = \overline{R}_\beta(A)$
- (v) $\underline{R}_\beta(\underline{R}_\beta(A)) \subseteq \overline{R}_\beta(\underline{R}_\beta(A))$
- (vi) $\underline{R}_\beta(\overline{R}_\beta(A)) \subseteq \overline{R}_\beta(\overline{R}_\beta(A))$

Proof. (i) Let $x \in \underline{R}_\beta(X \setminus A)$ which is equivalent to $x \in \bigcup\{G \in \beta O(X), G \subseteq X \setminus A\}$. So there exists $G_0 \in \beta O(X)$ such that $x \in G_0 \subseteq X \setminus A$. Then there exists G_0^c such that $A \subseteq G_0^c$ and $x \notin G_0^c, G_0^c \in \beta C(X)$. Thus, $x \notin \overline{R}_\beta(A)$. So $x \in X \setminus \overline{R}_\beta(A)$. Therefore $\underline{R}_\beta(X \setminus A) = X \setminus \overline{R}_\beta(A)$.

(ii) Similar to (i).

(iii) Since $\underline{R}_\beta(A) = \bigcup\{G \in \beta O(X), G \subseteq A\}$. This implies that

$$\begin{aligned} \underline{R}_\beta(\underline{R}_\beta(A)) &= \bigcup\{\bigcup\{G \in \beta O(X), G \subseteq A\}\} \\ &= \bigcup\{G \in \beta O(X), G \subseteq A\} = \underline{R}_\beta(A) \end{aligned}$$

$$\begin{aligned} \overline{R}_\beta(\overline{R}_\beta(A)) &= \overline{R}_\beta(X \setminus \underline{R}_\beta(X \setminus A)) = \\ \text{(iv)} \quad X \setminus \underline{R}_\beta(X \setminus \underline{R}_\beta(X \setminus A)) & \text{ . From (i), (ii)} \end{aligned}$$

and (iii), we get

$$\overline{R}_\beta(\overline{R}_\beta(A)) = X \setminus \underline{R}_\beta(X \setminus A) = X \setminus (X \setminus \overline{R}_\beta(A)) = \overline{R}_\beta(A)$$

(v) Since $\underline{R}_\beta(A) \subseteq \overline{R}_\beta(\underline{R}_\beta(A))$ and by (iii) we have

$$\underline{R}_\beta(\underline{R}_\beta(A)) = \underline{R}_\beta(A) \text{ , then}$$

$$\underline{R}_\beta(\underline{R}_\beta(A)) \subseteq \overline{R}_\beta(\underline{R}_\beta(A)) \text{ .}$$

(vi) Since $\underline{R}_\beta(\overline{R}_\beta(A)) \subseteq \overline{R}_\beta(A)$ and by (iv), we have

$$\overline{R}_\beta(\overline{R}_\beta(A)) = \overline{R}_\beta(A) \text{ , then}$$

$$\underline{R}_\beta(\overline{R}_\beta(A)) \subseteq \overline{R}_\beta(\overline{R}_\beta(A)) \text{ .}$$

Proposition 5.3. Let (X, R_β) be a near open approximation space and $A, B \subseteq X$. Then

- (i) $\underline{R}_\beta(A \cup B) \supseteq \underline{R}_\beta(A) \cup \underline{R}_\beta(B)$
- (ii) $\overline{R}_\beta(A \cup B) \supseteq \overline{R}_\beta(A) \cup \overline{R}_\beta(B)$
- (iii) $\underline{R}_\beta(A \cap B) \subseteq \underline{R}_\beta(A) \cap \underline{R}_\beta(B)$
- (iv) $\overline{R}_\beta(A \cap B) \subseteq \overline{R}_\beta(A) \cap \overline{R}_\beta(B)$

Proof. (i) Since we have $A \subseteq A \cup B$ and $B \subseteq A \cup B$. Then $\underline{R}_\beta(A) \subseteq \underline{R}_\beta(A \cup B)$ and $\underline{R}_\beta(B) \subseteq \underline{R}_\beta(A \cup B)$ by (iii) in Proposition 5.1, then $\underline{R}_\beta(A \cup B) \supseteq \underline{R}_\beta(A) \cup \underline{R}_\beta(B)$.

(ii), (iii) and (iv) Similar to (i).

The equality of all parts in Proposition 5.3 are not held as shown in the following example.

Example 5.1. According to Example 3.1:

(i) If $A = \{d\}$, $B = \{a, b\}$, then we have $\underline{R}_\beta(A \cup B) = \{a, b, d\}$, $\underline{R}_\beta(A) = \{d\}$, $\underline{R}_\beta(B) = \{a\}$. Therefore $\underline{R}_\beta(A \cup B) \neq \underline{R}_\beta(A) \cup \underline{R}_\beta(B)$.

(ii) If $A = \{d\}$, $B = \{a, b\}$, then we have $\overline{R}_\beta(A \cup B) = X$, $\overline{R}_\beta(A) = \{b, d\}$, $\overline{R}_\beta(B) = \{a, b\}$. Therefore $\overline{R}_\beta(A) \cup \overline{R}_\beta(B) \neq \overline{R}_\beta(A \cup B)$.

(iii) If $A = \{a, b, c\}$, $B = \{b, c, d\}$, then we have $\underline{R}_\beta(A \cap B) = \emptyset$, $\underline{R}_\beta(A) = \{a, c\}$ and $\underline{R}_\beta(B) = \{b, c, d\}$. Therefore

$$\underline{R}_\beta(A \cap B) \neq \underline{R}_\beta(A) \cap \underline{R}_\beta(B) \text{ .}$$

(iv) If $A = \{b\}$, $B = \{c, d\}$, then we have

$\bar{R}_\beta(A \cap B) = \phi$, $\bar{R}_\beta(A) = \{b\}$, $\bar{R}_\beta(B) = \{b, c, d\}$.
Therefore $\bar{R}_\beta(A) \cap \bar{R}_\beta(B) \neq \bar{R}_\beta(A \cap B)$.

The following theorems are a generalization of Proposition 5.3.

Theorem 5.1. Let (X, R_β) be a near open approximation space and $A, B \subseteq X$. If A is R_β -definable. Then the following are held.

(i) $\underline{R}_\beta(A \cup B) = \underline{R}_\beta(A) \cup \underline{R}_\beta(B)$

(ii) $\bar{R}_\beta(A \cap B) = \bar{R}_\beta(A) \cap \bar{R}_\beta(B)$

Proof. (i) It is clear that $\underline{R}_\beta(A) \cup \underline{R}_\beta(B) \subseteq \underline{R}_\beta(A \cup B)$.

For the converse inclusion, let $x \in \underline{R}_\beta(A \cup B)$, that means, $x \in \bigcup \{G \in \beta O(X), G \subseteq A \cup B\}$. Then there exists $G_0 \in \beta O(X)$ such that $x \in G_0 \subseteq A \cup B$. We distinguish three cases:

Case (1) If $G_0 \subseteq A$, $x \in G_0$ and G_0 is a β -open set, then $x \in \underline{R}_\beta(A)$.

Case (2) If $G_0 \cap A = \phi$, then $G_0 \subseteq B$ and $x \in G_0$, thus $x \in \underline{R}_\beta(B)$.

Case (3) If $G_0 \cap A \neq \phi$. Since $x \in G_0$ and G_0 is a β -open set, then $x \in \beta cl(A)$, for every G_0 which has the above condition, thus, thus $x \in \bar{R}_\beta(A)$, then $x \in \underline{R}_\beta(A)$, because A is R_β -definable. Hence, in three cases $x \in \underline{R}_\beta(A) \cup \underline{R}_\beta(B)$.

(ii) It is clear that $\bar{R}_\beta(A \cap B) \subseteq \bar{R}_\beta(A) \cap \bar{R}_\beta(B)$. We prove the converse inclusion. Let $x \in \bar{R}_\beta(A) \cap \bar{R}_\beta(B)$, then $x \in \bar{R}_\beta(A)$ implies $x \in \underline{R}_\beta(A)$ and $x \in G \subseteq X$, where G is a β -open set and $x \in \bar{R}_\beta(B)$ implies for all $G \in \beta O(X)$, $G \cap B \neq \phi$. Therefore $G \cap (A \cap B) = (G \cap A) \cap B = G \cap Y \neq \phi$. Hence $x \in \bar{R}_\beta(A \cap B)$.

Theorem 5.2. Let (X, R_β) be a near open approximation space and $A, B \subseteq X$. Then the following are held.

(i) $\bar{R}_\beta(cl(A) \cup B) = cl(A) \cup \bar{R}_\beta(B)$

(ii) $\underline{R}_\beta(int(A) \cap B) = int(A) \cap \underline{R}_\beta(B)$

Proof. (i) By Proposition 5.1 (i) and Proposition 5.3 (ii), we have $cl(A) \subseteq \bar{R}_\beta(cl(A))$. Then

$cl(A) \cup \bar{R}_\beta(B) \subseteq \bar{R}_\beta(cl(A)) \cup \bar{R}_\beta(B)$. On the other $\subseteq \bar{R}_\beta(cl(A) \cup B)$

hand, since $cl(A) \cup B \subseteq cl(A) \cup \bar{R}_\beta(B)$ and the union of a β -open set and a closed set is β -closed, then $\bar{R}_\beta(cl(A) \cup B) \subseteq \bar{R}_\beta(cl(A) \cup \bar{R}_\beta(B)) = cl(A) \cup \bar{R}_\beta(B)$. Therefore, $\bar{R}_\beta(cl(A) \cup B) = cl(A) \cup \bar{R}_\beta(B)$.

(ii) Since the intersection of an open set $int(A)$ and a β -open set $\underline{R}_\beta(B)$ is β -open, $int(A) \cap \underline{R}_\beta(B) = \underline{R}_\beta(int(A) \cap \underline{R}_\beta(B)) \subseteq \underline{R}_\beta(int(A) \cap B)$. On the other hand, by using Proposition 5.3 (iii), $\underline{R}_\beta(int(A) \cap B) \subseteq \underline{R}_\beta(int(A)) \cap \underline{R}_\beta(B) \subseteq int(A) \cap \underline{R}_\beta(B)$. Therefore $\underline{R}_\beta(int(A) \cap B) = int(A) \cap \underline{R}_\beta(B)$.

VI. CONCLUSIONS

In this paper, we used general binary relations to generate rough approximations. Properties of rough belonging, rough equality and rough power set are now clear with respect to any relation type raised from any field of applications.

We have proved that there exist similar properties and there exist different properties among the approximations raised from this paper and that studied from many authors [5,6,7,11,13,14,15,16,25,27,30]. We will investigate new applications of these tools using topological generalizations in a future paper.

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Analysis of Cloud Network Management Using Resource Allocation and Task Scheduling Services

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Abstract—Network failure in cloud datacenter could result from inefficient resource allocation; scheduling and logical segmentation of physical machines (network constraints). This is highly undesirable in Distributed Cloud Computing Networks (DCCNs) running mission critical services. Such failure has been identified in the University of Nigeria datacenter network situated in the south eastern part of Nigeria. In this paper, the architectural decomposition of a proposed DCCN was carried out while exploring its functionalities for grid performance. Virtualization services such as resource allocation and task scheduling were employed in heterogeneous server clusters. The validation of the DCCN performance was carried out using trace files from Riverbed Modeller 17.5 in order to ascertain the influence of virtualization on server resource pool. The QoS metrics considered in the analysis are: the service delay time, resource availability, throughput and utilization. From the validation analysis of the DCCN, the following results were obtained: average throughput (bytes/Sec) for DCCN = 40.00%, DCell = 33.33% and BCube = 26.67%. Average resource availability response for DCCN = 38.46%, DCell = 33.33%, and BCube = 28.21%. DCCN density on resource utilization = 40% (when logically isolated) and 60% (when not logically isolated). From the results, it was concluded that using virtualization in a cloud DataCenter servers will result in enhanced server performance offering lower average wait time even with a higher request rate and longer duration of resource use (service availability). By evaluating these recursive architectural designs for network operations, enterprises ready for Spine and leaf model could further develop their network resource management schemes for optimal performance.

Keywords—Resource Provisioning; Virtualization; Cloud Computing; Service Availability; Smart Green Energy; QoS

I. INTRODUCTION

A. Background Study

Middleware solutions for heterogeneous distributed cloud datacenters aim to respond to high requirements of large scale distributed applications relating to performance, flexibility, portability, availability, reliability, trust and scalability in the context of high number of users. These are usually considered in large geographic distribution of heterogeneous hardware and software resources. The concepts used in the design, implementation, and deployment of systems with such

capabilities could be based on demand side management and monitoring, optimization via scheduling, sharing, load balancing, consolidation and other high performance grid based techniques. In most cases, new services and functionalities could be added to the middleware to enhance data-intensive and highly demanding applications with low cost and high performance. New cloud computing architectures must be designed to incorporate solutions for the management of data, resources, tasks, and applications. It must ensure fault tolerance, accounting, service on demand, and other functions required by user communities to operate effectively in a shared services environment.

These observations formed the philosophy of an on-going research known as Smart Green Energy Management System (SGEMS). The system is a renewable energy system based on Solar PV microgrid, cloud energy meter and Distributed Cloud Computing Network (DCCN). In this system, the cloud datacenter server acts as the supporting platform for Enterprise Energy Tracking Analytic Cloud Portal (EETACP) deployment. But as energy users send their job tasks, fairness must be maintained optimally. Fairness in context refers to the method of having each job receive equal (or weighted) share of computing resources at any given moment. The DCCN must satisfy the fairness criteria for EETACP workload in the SGEMS research.

Studies have shown that datacenters are now the enterprise foundations that support many Internet applications, enterprise operations, and novel scientific computations like cloud computing services for distributed energy management platforms. In fact, they are large-scale data-intensive computing infrastructure. The major challenge facing smart green IT researchers is how to build a scalable cloud based DCN platform that delivers significant aggregate bandwidth and excellent Quality of Service (QoS) for smart grid web platforms. On this issue, research efforts such as Fat-tree [1], [2], VL2 [3], Monsoon [4], DCell [5], MDCube[6], BCube [7], FiConn [8], DPillar[9], DRweb[10], SVLAN [11], and Scafida [12], etc, have been proposed in recent years based on their switch and server-centric network architectures, with no attention on excellent resource management schemes. However, these works have made significant contributions on server interconnectivity primarily. For fairness in this type of

network, resource allocation and scheduling remains indispensable.

According to [13], by running large-scale computation and data-intensive services on inexpensive server clusters and other large-scale data parallel systems, cloud provisioning, (i.e., allocating resources for cluster requests) remains key to consolidating such clusters. Basically, resource management problems in multi-cluster environments are broadly classified into three large categories viz:

1) *Cloud providers provisioning/delivering raw clusters based on resource requirements of their customers.*

2) *Customers running cluster operating systems to manage critical server resources and schedule jobs from multiple frameworks.*

3) *Tasks Scheduling frameworks with or without assistance from the cluster operating system to get the job done.*

The fundamental goal of a well-developed resource management scheme on a server cluster is to create a cost effective model taking cognizance of the aforementioned categories while formulating a validation mechanism that will justify performance of the proposed system. The SGEMS EETACP platform that runs on the DCCN places a computation demand on resource allocation in cloud based environment. Since stability criterion must be satisfied, the use of virtual machine as the minimum resource allocation unit in the DCCN could suffice. When a user starts an application, a virtual machine that satisfies the minimum resource requirement for the application is allocated via scheduling map. When workload of the application increases as a result of user traffic, a new virtual machine is allocated for this application. This must allocate more physical resources (CPU, Memory, etc.) to the existing virtual machine, without shutting down the existing virtual machine for resource reallocation. This is most ideal for EETACP as a mission critical application since downtime is not an option or desired.

To provide guarantees of server operations, the datacenter with a cluster of servers must provision sufficient resources to meet application needs. Such provisioning can be based either on a dedicated or a shared model. In the dedicated model, a number of cluster nodes are dedicated to each application and the provisioning technique must determine how many nodes to allocate to the application [14]. In the shared model, running cloud virtualization for resource scheduling can allow running applications to share server resources with other applications. Once the cloud driver allocates a set of resource units such as virtual machines, the MapReduce system uses the resources that are heterogeneous shared among multiple jobs in EETACP context.

Therefore, this paper focuses on studying resource allocation and scheduling at both the application level and the backend core to see how to map the physical resources to Virtual machines for better resource utilization in DCCN environment.

B. Contributions

In this work, an experimental investigation on metrics associated with job scheduling and resource allocation on a

shared heterogeneous server cluster was carried out using DCCN, DCell and BCube as a case studies. Virtual Machine (Vm) algorithms were developed to provide good performance while guaranteeing fairness in an operational setup. Consequently, this will represent the ideal mode for EETACP service provisioning. The perspective offered in context is that an efficient task resource scheduling algorithm based on virtualization should be implemented at the broker domain for the DCCN. This was carried out to facilitate the deployment of a previous work on EETACP service proposed for SGEMS in an earlier work. The aim is to dynamically allocate the virtual resources in the EETACP as well as other services based on their workload intensities. This is to improve resource utilization, throughput, and availability and reduce the usage cost.

The rest of this paper is organized as follows. Section II presented the literature review as well as foundational concepts. Section III discussed the methodology and relevant system Models. Section IV presents the system Validation from the simulation environment. Section V concludes the paper.

II. LITERATURE REVIEW

In this section, two interrelated concepts will be clarified so as to provide a working foundation for EETACP deployment in DCCN at large. These are virtualization and resource allocation. Secondly, this work will then present the related works.

A. Cloud Virtualization and Resource Allocation

In [15], Virtualization is defined as the mirror imaging of one or more workstations/servers, etc within a single physical computer utilizing the same system resources. Virtualization makes cloud computing possible, since scalability is the major consideration in cloud computing. Cloud computing servers use the same operating systems, enterprise and web applications as localized virtual machines and physical servers [16]. Other views on the concept are detailed in [17] and [18]. Other the other hand, resource allocation is the process of assigning available resources to the needed cloud applications over the internet via the cloud DataCenter. A dynamic resource allocation framework (resource controller) in cloud environment helps to monitor traffic load changes, analyse workload and facilitate the implementation of an automated elastic resource controller that ensures high availability.

Resource controller in context, controls all the components on the cloud side and it has access to the load balancer, monitoring data, and front end (open nebula) for requesting additional resources on demand [19]. Some vendor services on a distributed cloud may include: computational resource configuration of the Virtual Machines (VMs), the programmer's degree of control, network service configuration, the nature of hardware/software security services, portability guarantees, storage scalability, etc, as such there is a need for a comprehensive resource allocation and scheduling system for cloud datacenter networks (CDCN) [19].

It is worthy to note that the allocation of resources to dedicated servers without virtualization schemes could be

problematic, while over-provisioning resources based on worst case workload estimates can result in potential network crash/failure and then violate of guarantees of QoS. An alternate approach is to allocate resources to servers dynamically based on the variations in user workloads profiles. In this approach, each server is given a certain minimum share based on coarse-grain estimates of its user resource needs.

Such dynamic resource sharing can yield potential multiplexing gains, while allowing the system to react to unanticipated increases in application load thereby meeting the QoS guarantees if both are cloud based.

In addition, for an excellent resource provisioning technique, there is need to determine the influence of service availability and processing delays on the server backend. By sharing server resources, this can provide guarantees to applications in the cloud DataCenter model. However, such guarantees are provided by reserving a certain fraction of node resources (CPU, network, and disk) for each application. In this regard, the size of the resources allocated to each cloud server will depend on the expected workload and the QoS requirements of the application.

For these workloads, there is a need to actually ascertain the influence of resource allocation using virtualization in enterprise computing cloud applications. Knowing the kind of servers that will scale in the event of high traffic density is very vital.

Consequently, this paper will use the concept of virtualization to explain the resource allocation and scheduling features in a cloud based management platform. The emphasis is on user jobs (workload) on server pools for a proposed DCCN. In developing this paper, strong emphasis is placed on dynamic resource allocation and scheduling technique via virtualization so as to handle changing application workloads in shared distributed Cloud Computing backend environment.

B. Related Research Efforts

Several works have been carried out on resource allocation and scheduling focusing on cloud infrastructure enhancement. This work will discuss these efforts below leveraging the Systematic Literature Review (SLR) approach.

The authors in [20] classified resource allocation models into three categories viz: processing resources, network resources and energy efficient resources. The work opined that network performance and resource availability could pose the tightest bottleneck for any cloud platform. Traditional Resource Management Systems (RMSs) such as Condor [21], LoadLeveler [21], Load Sharing Facility (LSF) [22], and Portable Batch System (PBS) [168], all adopt system-centric resource allocation approaches that focus on optimizing overall cluster performance. However, these have not been exploring in Spine leaf DCCN. In [24], a SLA-oriented resource management system built using Aneka [170] for cloud computing was proposed. A representative sample of works on resource allocation in the cloud environment for job task processing, and other resource categories is detailed in [25], [26], [27], [28], [29] and [30]. These works failed to justify their relevance in Spine leaf DCCN. Also, a process

scheduling and its algorithm was presented in [31],[32]. The works considered a resource allocation scheme on multiple clouds in cases of under load and the over load conditions. The paper in [33] a proposed model for cloud computing scheduling based on multiple queuing models for improve the quality of service by minimize execution time per jobs, waiting time and the cost of resources to satisfy user's requirements. The experimental results indicate that the model increases utilization of global scheduler and reduce waiting time. But, achieving resource allocation in a DCCN is only feasible through scheduling as a DCCN service. Similarly, the authors in [33], established some scheduling schemes and their strategies which were explained in [34]. The opined that both cannot be used in cloud computing for Application Processing Requests (APR) as found in [35],[36] owing to some identified QoS limitations. In [37], the authors proposed a cost-optimal scheduling in hybrid IaaS due to divergent users' requirements and heterogeneous workload characteristics.

The authors observed that the problem of scheduling a user's workload in the cloud remains a complex thereby proposing an optimal cost scheduling scheme. In [38], a genetic algorithm scheduling approach was proposed for addressing the problems of scheduling with traditional algorithms resulting in load imbalance and high migration costs. Other efforts made in literature in these areas of resource scheduling include: Greedy Particle Swarm Optimization (GPSO) [39], Task Length and User Priority (ie. Credit Based Scheduling Algorithm) [40], Cost based scheduling [41], Energy efficient optimization methods [42], Activity based costing [43],[44], Reliability Factor Based [45], Context aware scheduling [46],Dynamic slot based scheduling [47],[48], Multi-Objective Tasks Scheduling Algorithm [49], Public Cloud Scheduling Algorithm with Load Balancing [50], Agent-based elastic Cloud bag-of-tasks concurrent scheduling [51], Analytic hierarchy process (task scheduling and resource allocation) [52], Swarm scheduling [53], Profit-driven scheduling [54], Dynamic trusted scheduling [55], Community-aware scheduling algorithm [56], Adaptive energy-efficient scheduling [57], grid, cloud and workflow scheduling [58]. In these algorithms, job/task length and priority are mostly the parameters analyzed. However, the SGEMS DCCN inherits from the characteristics of the above works, but focused on improving the network QoS with respect to state of art Spine Leaf network model. The research gaps below were used to conclude the findings from literature review above.

C. Research Gaps

The following were clearly identified from literature study.

- QoS Resource Management

From the literature reviews, it has been shown that resource allocation, scheduling and service provisioning are the critical concepts in DataCenter network operations which must be considered when designing an efficient cloud based network. But existing works have not resolved the issues of excellent quality of service in cloud servers via virtualization scheme particularly for DCCN running on a spine leaf operational mode.

Consequently, this work proposes a dynamic architecture that handles all the resources in the DCCN by managing client requests, directing resource allocation, eliminating performance constraints, minimizing cost while ensuring the overall QoS. In this paper, resource management for client requests is carried out in the server cluster pool

- Validation Comparison with DCell and BCube

Based on the heuristic branch-and-bound concept with Riverbed modeller, a scenario based study with similar network architectures (DCell and BCube) will be carried out considering throughput, resource availability and network density as metrics. To the best of our knowledge, this work is the only work that have carried out a scenario based comparison with scalable DCell and BCube on the basis of heuristic tasks and priority scheduling in spine leaf DCCN.

III. RESEARCH METHODOLOGY

The method used in this work is referred to procedural benchmarking with Riverbed Modeller 17.5. In this case, a step by step approach was employed in studying BCube and DCell legacy x86 server consolidations as typified in University of Nigeria Nsukka (UNN) DCN as a case study. With the identified QoS issues in the network, this work leveraged server virtualization based on VMware vSphere which is a more mature and trusted technology in the enterprise spine leaf DCCN. In this regard, this work considered three key conditions when applying parallel processing in executing tasks in DCCN server viz: 1) how to allocate resources to tasks; 2) in what order is the task executed in the cloud; and 3) how to schedule overheads when VMs prepare, terminate or switch tasks. Task scheduling and resource allocation basically address these issues. The procedural benchmarking approach used in this work took care of the initial design specification and composite process model of the DCCN. This model architecture is presented next

A. DCCN Model Architecture/Specifications

Considering the DCCN architecture shown in Fig 1. The design comprises of two layers functional areas, viz: remote user access and the hybrid speed redundancy layer. The gateway load balancer (GLB)/speed redundancy layer was used interchangeably with the Integrated Service OpenFlow load balancer (ISOLB) in this work. The ISOLB connects the cloud layer to the broker which coordinates the *Vms*.

The cloud computing architecture in Fig.1 uses the cloud broker to mediate negotiations between EETACP Software as a Service (ESaaS) and cloud provider. This negotiation is driven by QoS requirements. The broker acts on behalf of ESaaS for allocation of resources that can meet application's QoS requirements. In the DCCN, the ISOLB is the major component in the hybrid speed redundancy layer. However, the hybrid speed redundancy layer comprises the Virtual machines interconnected server subnet clusters and the ISOLB.

The architectural decomposition of DCCN will be discussed next while exploring its functionalities for grid performance using virtualization metrics. A generalized

specification of the proposed DCCN datacenter is presented below.

- Let $DCCN_{lb}$ be an acronym chosen for the DCCN server cluster managed by the ISOLB controller. $DCCN_{lb}$ is designed to have various subnets for its clusters (eg. subnet l to n) referred to as $DCCN_{sa}$, $DCCN_{sb}$, $DCCN_{sc}$, $DCCN_{sd}$ interconnected together. $DCCN_{sa}$ represents a subnet as shown in Fig1. where n is a subnet factor such that $n > 0$. Each cluster ($DCCNs$) uses High Performance Computing (HPC) servers running Vm with the ISOLB controller layered in linearly defined architecture. Since the design of datacenter network is for efficient server load balancing and EETACP application integration, the requirement of 4-ports from ISOLB controller and few servers necessitated the choice of four subnets. Virtual server instances running on the HPC servers expanded the server cluster capabilities.
- Servers in DCCN cluster are connected to ISOLB port of the load balancer corresponding to it, and owing to the running virtual instances V_i , a commodity 4-port switching/routing device with 40GB/s per port serve the design purpose. Also, each of the $DCCNs$ is interconnected to each other through the ISOLB switch ports.
- The virtualized servers used in this work have two ports for redundancy (in Gigabytes). Each server is assigned a 2-tuple $[a_1, a_0]$ in consonance with its ports (a_1, a_0 are the redundant factors) together with an OpenFlow VLAN id.
- Emulated NEC IP8800 OpenFlow controller was the ISOLB used in this work; hence, the number K is the maximum number of OpenFlow-VLAN that can be created in it. The load balancer switch is a multilayer commodity switch that has a load balancing capability. This capability together with its OpenFlow VLAN capability was leveraged to improve the overall DCCN performance.
- Each server has its interface links in $DCCN_s$. One connects to an ISOLB, and other servers connect as well but all segmented within their subnets via OpenFlow-VLAN segmentation, as shown in Fig 2. OpenFlow DCN_s servers have virtual instances running on it and are fully connected with every other virtual node in the architecture.

Virtualization facilitates efficient use of hardware and software resources in DCCN. Hence, Virtual Machines *Vms*, are allocated to the user based on their job in order to reduce the number of physical servers in the cloud environment particularly in a high gird environment. But most VM resources are not efficiently allocated based on the characteristics of the job to meet out Service Level Agreements (SLA). Hence, this work introduced smart *Vm* allocation algorithm based on the characteristics of the EETACP job which can smartly reconfigure virtual resources thereby improving resource utilization in the server clusters.

Again, the DCCN port interface model for the ISOLB is shown in Fig.2. This creates advanced redundancy and multiplexing of job requests to the server clusters in the DCCN. For the DCell and BCube, these are not visible in

their architecture. The Algorithm for Vm allocation and scheduling in a multiplexed server setup is shown in Algorithm I.

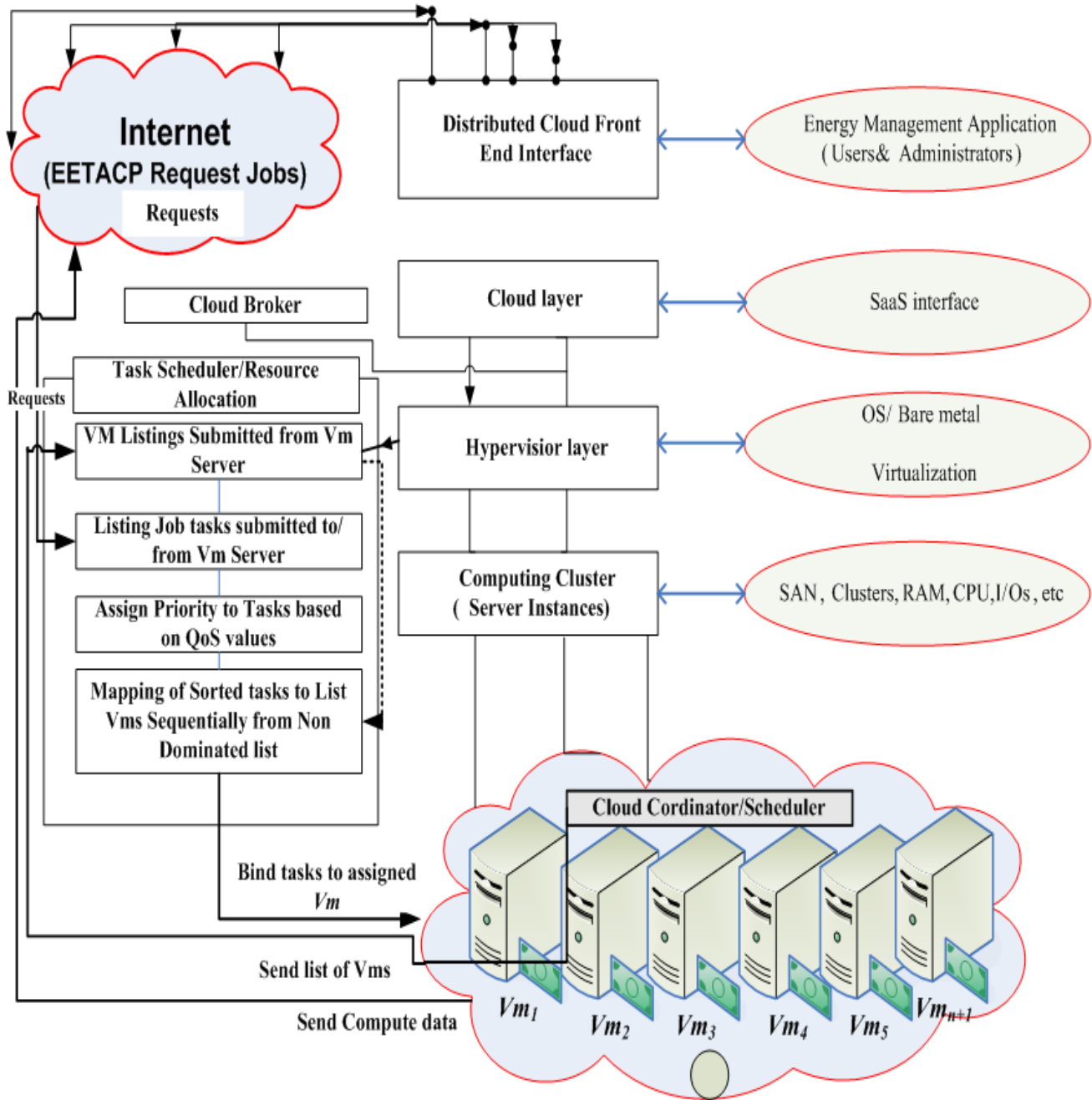


Fig. 1. DCCN Resource Allocation and Task Scheduling with Virtual Machines

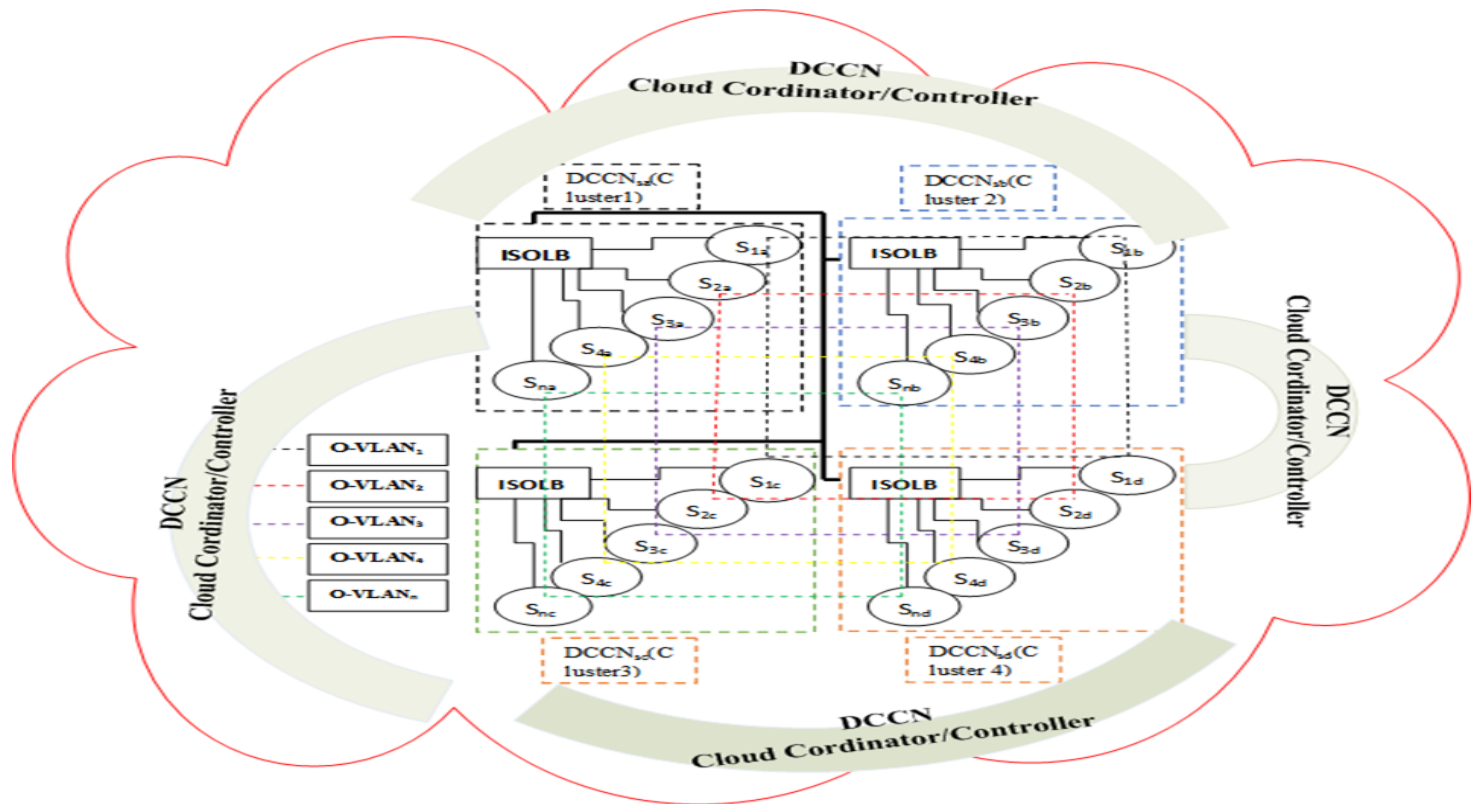


Fig. 2. DCCN ISOLB logical Server Mapping with VLAN Service

Algorithm I shows this concept which was not found in DCell and BCube models.

Algorithm I: Execution of High priority job when all existing resources are allocated in DCCN Server Cluster

```

Input: New job, all jobs running in EETACP host
Output: Execution of all jobs submitted to the EETACP host
Process
1 Begin ()
2 Arrival of New job for user I into Fig 1
3 if (New job.deadline < all jobs running in host)
4 High priority job from user i → New job
5 if (VM is available)
6 allocate High priority job to that VM
7 else
8 Suspend job ← Selection of Job for execution of high priority job();
9 Suspend (Suspend job)
10 allocate High priority job to VM from which a job was suspended
11 end if;
12 Execution of all jobs running in the VM Instance
13 if (completion of a job which is running in VM)
14 resume (Suspend job)
15 allocate the resumed job request to that VM Instance
16 End if
17 Execution of resumed job in active state
18 End
    
```

B. DCCN Service Coordinator

In DCCN, a virtualization algorithm known as DCCN logical construction algorithm was introduced to consolidate the efforts of the cloud coordinator controller in Fig.1 and Fig.2.

Recall that Fig 2 shows the design architecture for DCCN ISOLB logical server mapping with VLAN service. In this work, the ISOLB supports embedded VLAN service which offers excellent computing characteristics like flexibility, security, broadcast regulation, congestion control, etc. This is referred to as the Cloud Coordinator Service. It primarily isolates each server cluster from another, facilitates virtualization scheduling explained in Fig 1 and carries out resource allocation for EETACP workload in the cloud servers. The DCCN OpenFlow coordinator service algorithm is shown in Algorithm II.

The first section checks whether DCCN_s server subnet cluster is constructed. If so, it connects all the server nodes n to a corresponding ISOLB port and ends the recursion. The second section interconnects the servers to the corresponding switch port and any number of $PS_K.Vm$ or P_i servers are connected with one link. Each server in the subnet cluster DCCN_{lb} is connected with 40GB links for all OpenFlow VLAN_{id}. The role of the service coordinator in the cloud is enormous. The DCCN logical architecture with the OpenFlow VLAN segmentation shown in Fig 2 uses the linear construction algorithm II for VLAN resource scalability.

Algorithm II: DCCN Service Coordinator OpenFlow VLAN Construction Algorithm.

```

/* 1 stands for the level of DCCNs subnet links, n is the
number of nodes in a cluster
DCCNlb
pref is the network prefix of DCCNlbs is the number of
servers in a DCCNlbcluster*/
Build DCCNs(l, n, s)
Section I: /* build DCCNs */
    If (l = 0) do
        For (inti = 0; i < n; i++) /* where n is=4*/
            Connect node [pref, i] to its switch;
    Return;
Section II: /*build DCCNs servers*/
    For (inti = 0; i < s; i++)
        Build DCCNs ([pref, i], s)
        Connect DCCNs(s) to its switch;
    Return;
End

```

In the DCCN logical structure shown in Fig 2, the servers in one subnet are connected to one another through one of the ISOLB ports that is dedicated to that subnet. Each server in one subnet is also linked to another server of the same order in all another subnets. Incast collapse in cloud datacenter must be avoided at all cost. As such, each of the servers has two links, with one, it connects to other servers in the same subnet (intra server connection) and with the other it connects to the other servers of the same order in all other subnets (inter server connection). Apart from the communication that goes on simultaneously in the various subnets, the inter server connection is actually an OpenFlow VLAN connection. This OpenFlow VLAN segmentation of the servers logical isolates them for security and improved network performance. Together with other server virtualization schemes ultimately improves the performance in terms of throughput and other QoS metrics. The OpenFlow VLAN segmentation gave each DCCN_s (subnet) the capacity to efficiently support enterprise web applications (EETACP, Web Portals, Cloud applications such as ESaaS) running on server virtualization in each port thereby lowering traffic density.

C. Coordinator Logical Isolation of Server Clusters

As shown in Fig.2, the application of OpenFlow VLAN in each subnet creates full logical isolation of the DCCN server cluster architecture of Fig 1. In order to achieve this, each server and nodes in DCCN_s is assigned virtualization identity, [$V_{id} = av_1, av_2, \dots, av_{n-1}$] and OpenFlow VLAN identity (V_{id}) greater than 0, where $av_1, av_2, \dots, av_{n-1}$ is the virtualization instances on DCCN_s servers. As such each server can be equivalently identified by a unique V_{id} in the range $V_{id} \leq 1005^*$.

Hence the total of V_{id} for servers in the DCCN_s is given by (1)

$$T(V_{id}) = \sum_{id=1}^N V_{id} * V_m(i, \phi) \tag{1}$$

Where N is the maximum number of OpenFlow VLAN ids, and $V_m(i, \phi)$ is the virtual instances in the DCCN_{lb} physical servers. The mapping between a unique V_{id} and the DCCN_{lb} physical servers considering that there are four subnet clusters DCCN_{lb} in Fig 2 is given by (2)

$$DCCN_{lb_Mapping} = 4 * V_{id} * V_m(i, \phi) \tag{2}$$

Following the DCCN architecture in Fig.1, in order to minimize broadcast storms and reduce network traffic/demand density, an OpenFlow VLAN mapping scheme for the servers in the subnet clusters DCCN_{lb} was applied resulting to the system validation model discussed in section IV.

Now, consider DCCN_{sa}, DCCN_{sb}, DCCN_{sc} and DCCN_{sd} with servers S_1 to S_n . The servers in each of the subnet cluster are mapped into different OpenFlow_VLAN services with their corresponding *ids* as follows:

- OpenFlow_VLAN₁ → $S_{1a}, S_{1b}, S_{1c}, S_{1d}, \dots, S_{1n}$
- OpenFlow_VLAN₂ → $S_{2a}, S_{2b}, S_{2c}, S_{2d}, \dots, S_{2n}$
- OpenFlow_VLAN₃ → $S_{3a}, S_{3b}, S_{3c}, S_{3d}, \dots, S_{3n}$
- OpenFlow_VLAN₄ → $S_{4a}, S_{4b}, S_{4c}, S_{4d}, \dots, S_{4n}$
- OpenFlow_VLAN_n → $S_{na}, S_{nb}, S_{nc}, S_{nd}, \dots, S_{nn}$

Where $S_{1a}, S_{2a}, S_{3a}, S_{4a}$ are the servers in DCCN_{sa}
 $S_{1b}, S_{2b}, S_{3b}, S_{4b}$ are the servers in DCCN_{sb}
 $S_{1c}, S_{2c}, S_{3c}, S_{4c}$ are the servers in DCCN_{sc}
 $S_{1d}, S_{2d}, S_{3d}, S_{4d}$ are the servers in DCCN_{sd}.

With this OpenFlow VLAN mapping scheme, a logical isolation of the DCCN cluster subnets in the DCCN architecture was achieved. This make for smart flexibility, improved network security, agility and control of traffic flow in the DCCN core layer.

D. DCCN Resource Allocation

For user workloads in the virtualized server subnet cluster of Fig 2, by quantifying the workload requirements for the physical servers, the virtual machines can significantly benefit from the resource allocation strategies offered by the full virtualization scheme. It will be established that virtualization can offer excellent resource allocation of distributed server resources. In the DCCN, user tasks are assigned resources for effective performance in the server domain. The tasks represents user job request for server execution. I/Os, CPU, Memories are resources that are assigned in context. Resource allocation via virtualization makes for bandwidth availability, delay reduction and service availability in general. These are vital QoS metrics analyzed in this work.

From the framework of Fig.2,

- Let n_n = User U_i task number or job numbers
- m_m = DCCN resource number,
- K_k = DCCN QoS parameter number

By introducing QoS parameters for each resource in a virtualized server, let's define a vector $V\beta_k^{res}$ which gives the functional capabilities of a resource given in Equation (3).

$$\beta^{res} = \langle \beta_1^{res}, \beta_2^{res}, \beta_3^{res}, \dots, \beta_k^{res} \rangle \tag{3}$$

User U_i sends job request accompanied by a vector of QoS parameters, but the weights for the parameters are given in Equ 4 and 5 such that

$$\beta^{job} = \langle \beta_1^{job}, \beta_2^{job}, \beta_3^{job}, \dots, \beta_K^{job} \rangle \quad (4)$$

Vector weight $W_V = \langle W_1, W_2, W_3, \dots, W_k \rangle$

for $0 \leq W_i \leq 1$

$$\text{Hence, } W_V = \sum_{i=1}^k W_i = 1 \quad (5)$$

Here, each weight shows the importance of each parameter. Since CPU for instance is important for job execution, then it is activated or enabled by default. The same for other resources like I/Os, RAM, Storage disks, etc. If the resource provides solution for a requested service or task, the validity operator was introduced such that $R_{mm} \rightarrow T_{nn}$ in Equation (6)

$$\text{Hence, } R_{mm} \rightarrow T_{nn} = \left(\langle \sum_{i=0}^k \frac{\beta_1^{resmm}}{\beta_1^{jobnn}} * W_{V_i} \rangle \right) \geq 1 \quad (6)$$

Where K = the number of the QoS Parameters.

From Equ 6, the resource specification for DCCN for Users U_i was optimized.

When the resource capability exceeds the task demand, then the virtualization function in (7) holds for the DCCN server.

$$F(R_{mm} \rightarrow T_{nn}) = \frac{\beta_1^{resmm}}{\beta_1^{jobnn}} > 1 \quad (7)$$

To expand the optimization problem of assigning m_m resources to n_n tasks with K_k QoS parameters in the DCCN server cluster subnets, the compact matrixes was introduced in Equ 8, 9, and 10 for virtualization scheduling. These compact matrixes are described for resource allocation model in context. For users U_i , let task matrix requirements, be given by Equation (8).

$$T_{n_n * K_k} = \begin{pmatrix} \beta_1^{job_1} & \dots & \beta_{k+1}^{job_1} \\ \vdots & \ddots & \vdots \\ \beta_1^{job_n} & \dots & \beta_{k+1}^{job_n} \end{pmatrix} \quad (8) \quad PS_K.Vm = \sum_{i=1}^m (1) *$$

For user U_i , let task weight matrix requirements be given by Equ 9

$$W_{n_n * K_k} = \begin{pmatrix} W_1^{job_1} & \dots & W_{k+1}^{job_1} \\ \vdots & \ddots & \vdots \\ W_1^{job_n} & \dots & W_{k+1}^{job_n} \end{pmatrix} \quad (9)$$

For users U_i , let server resource capability matrix requirements be given by Equ 10

$$R_{m_m * K_k} = \begin{pmatrix} \beta_1^{res_1} & \dots & \beta_{k+1}^{res_{m_m}} \\ \vdots & \ddots & \vdots \\ \beta_k^{res_1} & \dots & \beta_{k+1}^{res_{m_m}} \end{pmatrix} \quad (10)$$

By dividing (9) by (8) to obtain a V_m vector λ_{K_k} , to get Equation (11)

$$\lambda_{K_k} = W_{n_n * K_k} / T_{n_n * K_k} = [W_{n_n * K_k}] [T_{n_n * K_k}]^{-1} \text{ such that}$$

$$\lambda_{K_k} = \begin{pmatrix} W_1^{job_1} / \beta_1^{job_1} & \dots & W_{k+1}^{job_1} / \beta_{k+1}^{job_1} \\ \vdots & \ddots & \vdots \\ W_1^{job_n} / \beta_1^{job_n} & \dots & W_{k+1}^{job_n} / \beta_{k+1}^{job_n} \end{pmatrix} \quad (11)$$

But resource allocation by V_m instance is given by (12),

$$V_{n_n * M_m} = \sum_{i=1}^{\phi} (\lambda_{K_k} * R_{m_m * K_k}) \quad (12)$$

Where ϕ = max virtual instance that can be accommodated by the physical server.

By substituting (10) and (11) into (12), a model for virtualization was developed by putting the task/job numbers (n_n), resource numbers (m_m) and QoS parameters (K_k) in (13)

$$V_{n_n * M_m} = \begin{pmatrix} \sum_{i=1}^{\phi} \left(\frac{W_1^{job_1}}{\beta_1^{job_1}} * \beta_1^{res_1} \right) & \dots & \sum_{i=1}^{\phi} \left(\frac{W_1^{job_1}}{\beta_{k+1}^{job_1}} * \beta_1^{res_{m_m}} \right) \\ \vdots & \ddots & \vdots \\ \sum_{i=1}^{\phi} \left(\frac{W_1^{job_n}}{\beta_1^{job_n}} * \beta_1^{res_1} \right) & \dots & \sum_{i=1}^{\phi} \left(\frac{W_1^{job_n}}{\beta_1^{job_n}} * \beta_1^{res_{m_m}} \right) \end{pmatrix} \quad (13)$$

Originally, it was established that the total server virtual instances in the DCCN server $PS_K.Vm$ is given by

$$PS_K.Vm = \sum_{i=1}^m V_m(i, \phi) = [V_{mi1} + V_{mi2} + \dots + V_{mi\phi}] \quad (14)$$

From the DCCN architecture shown in Fig 1, if the resource allocation model is given by Equ 13, then the consolidation model via virtualization for the physical servers can be obtained by substituting Equ 13 in Equ 14. This gives (15)

$$PS_K.Vm = \sum_{i=1}^m (1) * \begin{pmatrix} \sum_{i=1}^{\phi} \left(\frac{W_1^{job_1}}{\beta_1^{job_1}} * \beta_1^{res_1} \right) & \dots & \sum_{i=1}^{\phi} \left(\frac{W_1^{job_1}}{\beta_{k+1}^{job_1}} * \beta_1^{res_{m_m}} \right) \\ \vdots & \ddots & \vdots \\ \sum_{i=1}^{\phi} \left(\frac{W_1^{job_n}}{\beta_1^{job_n}} * \beta_1^{res_1} \right) & \dots & \sum_{i=1}^{\phi} \left(\frac{W_1^{job_n}}{\beta_1^{job_n}} * \beta_1^{res_{m_m}} \right) \end{pmatrix} \quad (15)$$

Equation (15) satisfies the requirements used for assigning resources to user request via EETACP (traffic load on DCCN server). From (13), if $V_{n_n * M_m} = 1$, then resources R_{m_m} will be precisely provide the tasks or job requirements T_{n_n} while maintaining the QoS requirements, but if $V_{n_n * M_m} < 1$, then the resource will be degraded by the tasks or job requirements T_{n_n} and affect server performance.

Again, in the DCCN, since user requests or demand arrives randomly into the gateway before the ISOLB, it was assumed that the job arrival follows the stochastic process such that the packet size is exponentially distributed, and the system is considered as an M/M/1 queuing system. Besides, traffic capacity management and optimum utilization of server resources takes care of network instability issues. But, to address the issue of traffic flow from the access layer, Little's

law suffices. It takes care of the system response time and scheduling distribution thereby maintaining traffic flow stability criterion.

Now, if the average arrival rate per unit time is given by $R_{L_m * K}$ and $P_{V m L_m}$ is the average service rate per unit time, stability condition for the system resource management can be deduced.

Considering the user task/job numbers (n_n), resource numbers (m_m) and QoS parameters (K_k), the proposed DCCN for enterprise cloud datacenter is considered stable only if $R_{L_m * K} < P_{V m L_m}$. If on the other hand, the average arrivals happen faster than the job service completions such that $R_{L_m * K} > P_{V m L_m}$, the job queues on the server clusters will grow indefinitely long and making the system to be unstable.

In this work, by generating an equivalent DCCN that will support the deployment of the proposed EETACP while exploring the mathematical models derived for the system, the QoS metrics evaluation will primarily be service availability, throughput, and fault tolerance. The algorithms for the distributed cloud management architecture were discussed previously. Fig 3 shows the testbed used for carrying out basic experimentations and validations in the context of application hosting and provisioning. Fig 4 shows the UNN datacenter which has network failures resulting from traffic imbalance and disturbances. This was addressed in validation section.



Fig. 3. Completed Testbed Setup for DCCN Vm Scheduling for Application Provisioning (SGEMS EETACP).

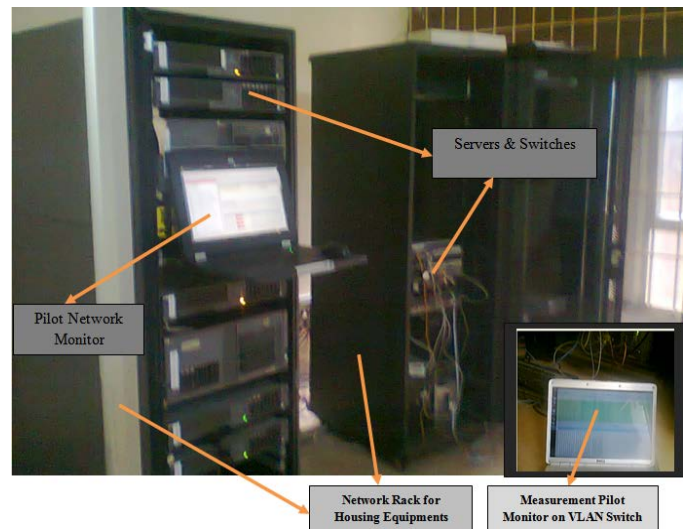


Fig. 4. UNN Testbed Setup without Vm Scheduling for Application Provisioning

IV. SYSTEM VALIDATION

As explained previously, procedural benchmarking with server virtualization for the spine leaf DCCN was used in this work. This is driven by virtual machines and task schedulers. In carrying out the validation analysis on DCCN, procedural benchmarking was applied in creating the relevant heuristic algorithms. Virtual machines and task schedulers were configured as extended attributes after importing the objects from the object palette in Riverbed Modeller/C++ Version 17.5 simulator [60]. Other components configured include the OpenFlow load balancer and the server cluster links. The mathematical characterization above was considered in the design. After comparing the service delay and availability of the two DCCN using the heuristic algorithms for task length and user priority, this work then introduced two related datacenter architectures having task based scheduling without virtualization i.e. DCell [5] and BCube [7]. Both have been extensively studied in [61]. The metrics computed includes average throughput (bytes/Sec), Average resource availability response and DCCN density on resource utilization. By extending the work carried out in [62] that focused on the impact of virtualization on server computing clusters, the contribution of the work now focused on server consolidation via virtualization for fault tolerance improvement in order to reduce down time scenario to the barest minimum. In this work, we focused on resource management using virtual machine for improved QoS. The simulation is done under the following conditions as enlisted in Table I, II and III.

TABLE I. BASIC CONFIGURATION

Number of Datacenter hosts	5
Number of Datacenters	5
Number of Brokers	1
Task Schedulers	10
Number of Cloudlets	10

Each datacenter consists of several hosts. Each host has its own configuration. Here, the same configuration is applied for each hosts. Host configuration is depicted in Table II.

TABLE II. HOST BASIC CONFIGURATION

Buffer Capacity	256000
Storage	1GB
Bandwidth (MB/Sec)	10k
Number of Virtual Machines	6
MIPS (Lines of Codes)	1000

The host in the datacenter consists of several virtual machines. Each virtual machine has its own configuration. Here, same configuration is applied for each VMs. Virtual machine configuration is mentioned in Table III.

TABLE III. VIRTUAL MACHINE BASIC CONFIGURATION

Number of Cores	2
Size (MB)	10000
Bandwidth (MB/Sec)	1000
RAM (MB)	512
MIPS (Lines of Codes)	1000

From Fig 5, it was observed that the proposed DCCN with optimal Virtual instance allocation coordinator had relatively a better throughput. The average throughput responses in percentages were obtained as follows: For DCCN = 40.00%, DCell = 33.33% and BCube = 26.67%. This shows that virtualization enhances performance.

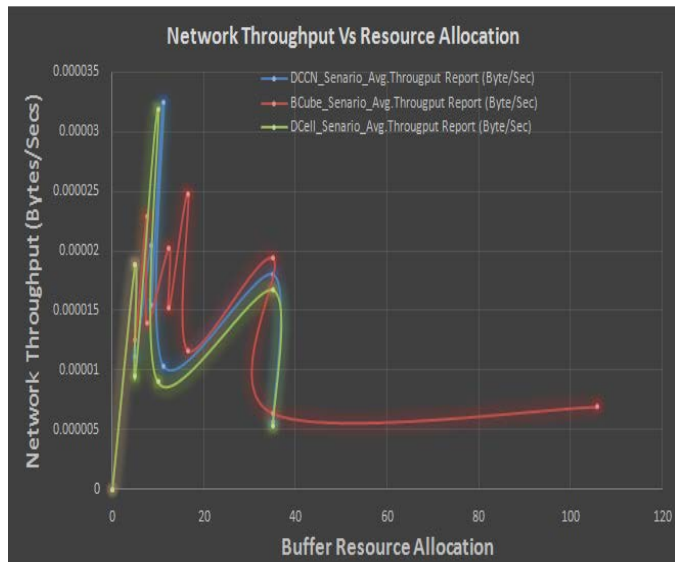


Fig. 5. Plot of Network throughput against Resource Allocation

As shown in Fig.6, resource availability refers to ability to access the DCCN server clusters on demand while completing the job requests. It was shown that the proposed model had fairly good resource availability leading to enhanced performance. Hence, a good resource allocation strategy will enhance performance at large. This still re-echoes the advocacy for Vm based cloud networks particularly in the cell based and spine leaf models like the DCCN in this research.

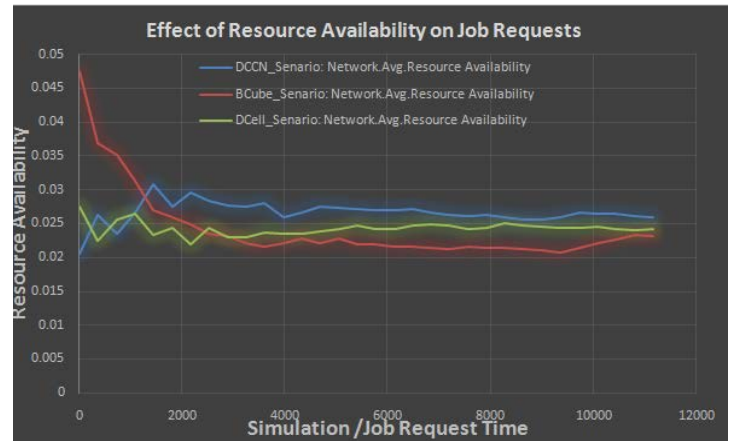


Fig. 6. Plot of resource Availability Vs Job requests time

The legacy UNN DCN in Fig 4 can be improved with this scheme.

From Fig. 7, it was shown that lower resource utilization for the proposed DCCN was achieved compared with BCube and DCell scenarios. When all existing resources (VMs) are allocated to low priority jobs and a high priority job comes in, the low priority job (deadline is high) has to be pre-empted so that its resources can allow a high priority job (deadline is low) to run in its resource tasks. When a job arrives, availability of the VM is checked based on the network density. If the Vm is available, then job is allowed to run on the VM. If the VM is not available, then the algorithm find a low priority job taking into account the job’s lease type. The low priority job pauses its execution by pre-empting its resource. In all cases, the high priority job is allowed to run on the resources pre-empted from the low priority. When any other job running on server VMs is completed, the job which was halted previously can be resumed. This Vm process in Algorithm1 facilitates lower resource utilization at large. DCCN density on resource utilization gave 40% (ie. when logically isolated) while the others offered 60% (ie. when not logically isolated).

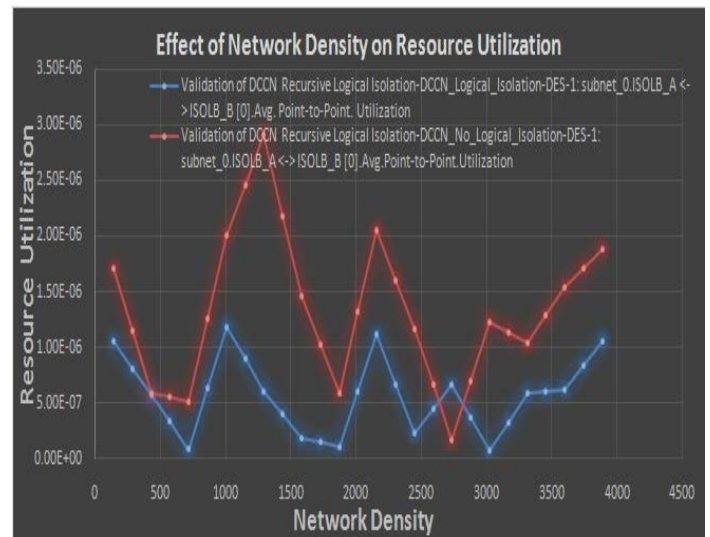


Fig. 7. Plot of Network density Vs Resource utilization

V. CONCLUSION

This paper focused on the critical design factors involved in the deployment of SGEMS DCCN as well as analyzing the impact of virtualization on resource allocation and scheduling strategies for managing Cloud DataCenter server clusters. In this respect, optimizing computational resources (ie. network resources) with low cost are the vital considerations for a successful cloud computing service deployment and its operations. The work used CloudSim equivalent tool Riverbed 17.5 with scenario based setups for DCCN, DCell and BCube. The implemented network was simulated while ensuring that *Vms* were allocated as hosts based on the capacity of the cloud service coordinator available.

Jobs are given to the *Vms* for execution based on First Come First Serve (FIFO) basis. The deadline was checked for high and low priorities. From the work, QoS metrics such as throughput, resource availability, and resource utilization were investigated. With the latest state of art in enterprise application network, the proposed Spine leaf DCCN offered throughput improvement of 6.67% over DCell while offering 13.33% over BCube. Also, the network offered 5.13% availability improvement over DCell while offering 10.25% over BCube. Consequently, the more resource pool and allocation on a virtualized cloud server, the better the overall system performance owing to virtualization effects. This can also stabilize the DCN operations marginally. Conversely, an inefficient resource allocation/scheduling scheme can adversely degrade the network performance. In conclusion, virtualization can smoothly facilitate resource allocation/scheduling schemes in a distributed server domain considering user job workloads. Future work will focus on *Vm* allocation in high density Spine and leaf architecture for the deployment of EETACP service in SGEMS (microgrid framework) while building datasets for big data analytics using an FPGA based big data environment.

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The Impact of the Implementation of the ERP on End-User Satisfaction Case of Moroccan Companies

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Abstract—In recent years, the implementation of ERP is as a lever for development and inter-organizational collaboration. The ERP is a powerful tool for integration, sharing of information, and fluidizing of the process within the organizations (El Amrani et al. 2006; Kocoglu and Moatti, 2010).

The company must not only equip and computerization but it must opt for the establishment of an IT infrastructure "optimal" who will respond to its present and future needs. OF or the interest of the application integration, and especially of the ERP who come remedy the situations mentioned. This article proposes a model and tests to evaluate the success of a system "Enterprise Resource Planning "(ERP) based on a measure of user satisfaction. Referring to the model DeLone & McLean (1992) and the work of Seddon & Kiew (1994) . The criteria that can influence user satisfaction, to ensure the successful implementation of the ERP system are identified.

The results of the exploratory study, carried out on 60 users in 40 Moroccan companies, shows that user satisfaction of ERP is explained by the quality of the ERP system, perceived usefulness and quality of information provided by this type of system. The study also found that the quality of change is a predictor of satisfaction measured by user involvement in the implementation of ERP, the quality of communication within such a project and the quality of training given to users.

Keywords—Enterprise Resource Planning (ERP); User Satisfaction; Quality Change; Information Technology (IT); Information Systems(IS); success; evaluation approaches; Evaluation Success Factors

I. INTRODUCTION

The current context of global economic activity is characterized by a large and permanent competition as well as a large customer requirement for immediate and complex solutions. In this context, process control and continuous improvement become prerequisites for success. As a result, numerous companies around the world are trying to take advantage of an overhaul, using software packages, their information systems, and hundreds of them have opted for systems integrated management ERP (Enterprise Resource Planning) as a basis for the integration of their industrial management (Marbert, Soni & Venkataraman, 2000). [3]

The use of information and communication technologies (ICTs) such as ERP seems to be a great contribution to the profound changes in the functioning of companies. These packages make reference to information systems (IS) implemented to integrate the flow of information at the level of

the entire organization. To achieve this integration between functional areas, ERP operate a centralized database that stores the collection and organization of data in "real time". Technological innovation of these software management can realize the old dream of the companies to have a single repository, and integrated their management information system (GIS) (Davenport, 1998) [4]; Rowe, 1999). Organizationally, the implementation of ERP is considered a change in the information system and in the process of guidance and control (Gomez et al. 2002). In fact, if many companies are attracted by the ERP, it is because this is supposed to make gains in productivity and efficiency, including the ability to make more integrated organization. This integration concerns both automated business processes that information processed by the software (Perotin, 2002). [5] Indeed, integration is placed among the main reasons companies to equip an ERP (Marciniak 2001).

Lequeux (1999) [6] defines the ERP system as "a subset of the IS able to take over the complete management of the company, including accounting and financial management, production management and logistics, managing human resources, administrative management and the management of sales and purchases. "

In other words, Perotin (2002) argues that the ERP system is defined as all "configurable and modular software applications, designed to integrate and optimize business management processes by providing a single repository and consistent and based on standard business rules". However, these systems are, in many cases, adopted without their suitability to the organizational context of the company is evaluated. Hence their implementation could result in low levels of user satisfaction, and therefore low levels of success. Moreover, these companies do not have often adequate tools that allow them to evaluate these systems and whether they meet the needs of individuals who use them.

The change must be seen as a solution enabling the organization to respond to quality management problem and adapting to look as well as its environment itself (Florescu Dumitru & 2007). A driving change plan is able to reduce resistance now. This study is interesting on two levels:

- The objective of this article is therefore to identify the drivers of satisfaction of users of ERP systems. On a finer way, we try to determine the satisfaction and enhance the need for good conduct ERP projects to

increase the degree of the satisfaction. To do this, it was reduced to build a model for the explanation of this satisfaction.

- In what follows, we will try to review the state of the art in measurement of user satisfaction of IT before submitting the research model and the results of an exploratory study conducted with a sample of Moroccan companies.

II. ERP AND ITS CONTRIBUTIONS FOR USERS

The evolution of computing, which is progressing towards greater information sharing and flexibility is a key factor explaining the growing success of ERP to companies. Despite the unquestionable progress they make today, ERP do not fully meet satisfactorily the needs of companies.

A. The emergence of integrity management software

Historically, functional systems businesses were developed on different materials following different methodologies: the achievements are generally heterogeneous both in terms of data representation at the level of processing modes. It follows multiple disadvantages:

- Communication problems between areas expected to share common data;
- Process control challenges due to the multiple treatments required to obtain synthetic statements;
- Students maintenance costs in the absence of modularity resulting low scalability;
- Complexity of the training was the use of very varied software;
- Difficulties for many controllers, in the collection and re-keying data from different systems and serving to consolidate budgets, develop reporting tables, etc.

Faced with these recurrent difficulties, software companies and large consulting firms offer a single answer in the form of integrity management software to improve the overall consistency while allowing modularity.

B. What is an ERP?

Acronym of American origin, ERP (Enterprise Resource Planning) is commonly used to designate the integrity management software. The term "ERP" is not totally adequate because it puts only evidence planning appearance. However the French translation "ERP" does not include the planning dimension and its use is problematic.

As defined by Robert Reix (1999), an ERP is a computer application that incorporated the following general characteristics:

- An ERP is a software package: according CXP4, a software package is "a coherent and independent set is service programs, supports, or handling of information and documentation, designed to perform standard IT processes, including the distribution is of a commercial nature and that a user can independently use after installation and limited training "(Sourdeau 1997, p20).

- An ERP is customizable: standardized product, the ERP is designed originally to meet the needs of various businesses. It usually are different versions by sector (automotive, banking, etc.) and prolonged use. In addition, the adaptation of the product to the needs of a particular business is by setting (choice of management rules, choice of treatment options, choice of data format, etc.). The setting may be accompanied by an appeal has additions of specific programs articulated around standard programs.

- An ERP is modular: it is not a monolithic structure but a set of programs or separable modules each corresponding to a management process: installation and operation can realize autonomously. The division into modules allows you to dial a specific solution for assembly and extend the implementation has different areas of management.

- A ERP is integrated: the various modules are not designed independently they can exchange information according to patterns provided. The PGI guarantees at all times a perfect integrity and data consistency for all users, allowing DC to end interfacing problems, synchronization and double entries.

- An ERP is a management application: it captures the company's transactions (accounting, stock management, order tracking and production program ...) and propagates the information collected to the appropriate levels. However, it contains no optimization program or automatic decision.

C. Literature review on approaches, models and frameworks ERP success

This part focuses on literature research is the success to summarize the two theoretical backgrounds and empirical studies. The presentation will be followed chronologically in terms of frames, models and approaches developed in the IS field. Then we will focus our attention on ERP as the main subject of this study. A review of the measurement of different approaches regarding the assessment of the success of ERP will be discussed to highlight the importance of the measure in the information system and in particular the ERP software.

1) successful frameworks for the evaluation of ERP

Development of a framework is the first step in the success of the assessment which must be adapted to the characteristics of the information system (Chand et al., 2005 [7]; Irani et al, 2014 [8]; Stefanou, CJ, 2001 [9]; Uwizeyemungu and Raymond, 2010 [10]) many frameworks have been proposed taking into account several phases and dimensions of evaluation of the system's success: strategic, tactical and operational. Generally, the framework describes eight categories: theoretical basis, the research approach, the object of analysis, the unit of analysis, the prospect evaluation, data collection, data analysis and type methodology (Urbach and Smolnik, 2008) [11]).

2) The PAC Setting

A proposal CPC could be considered an important framework for assessing the success of the ERP system because it incorporates three major dimensions part of the evaluation: content, context and process (Irani and love, 2008, Irani, 2002; Song and Letch, 2012). This framework was developed by (Irani and Love, 2008; Irani, 2002) [12] to help

managers and decision makers in the process of assessing the benefits of IT / IS. They argue that there is not a good framework to evaluate the impact of IT in the performance of the organization in the literature and they added that there is no good framework to select appropriate tools for IS investment. For these reasons, they try to offer a CCP framework for assessing the cost and benefits based on three concepts: Content, context and process. But we conclude after analyzing this framework it is too big and general to be applicable to conduct an evaluation of the success of the ERP system.

3) framework Stefanou

Part of Stefanou consists of four phases: the first considers the vision of the company as a starting point for ERP integration. The second phase focuses on the needs of business and the company's ability to support and adapt the ERP system. The third phase requires estimation of the costs and benefits of integration of the ERP system. The last phase refers to the analysis of issues related to the use ERP, maintenance and evolution.

The product quality is one of the most important of the ESF in the project; in this case, the quality of the product means that the quality of the ERP system. Many measures are proposed to assess the quality such as response time, convenience of access, implementation of user requirements, error correction, data security and models, system integration, system flexibility the effectiveness of the system, database content, data currency, correctness and accuracy of the data system (DeLone and McLean, 1992).

4) Soh and Markus framework

The ultimate goal (Markus and Tanis, 2000; Soh and Markus, 1995) that works is to create a new framework that allows a better understanding of the concept of ESS (Enterprise System Success). Answering these questions: how companies can successfully integrate this technology? And what can we do to improve the chances of success? Authors define the result of success as a multidimensional concept, a dynamic concept, and a report (the concept of "optimal success," that represents the best of an organization can hope to achieve with enterprise system). P. 184.

Success can be defined by (Markus and Tanis, 2000) in terms of implementing the project, or in terms of business results. The first definition answers the question: the company managed to get the system running within some reasonable budget and schedule? The second answers the question: the company has managed to achieve its business objectives for the project?

Based on the theories of fusion process because (Markus and Tanis, 2000) consider that these theories combine both objectives and actions with the external forces and luck. They build their frame on a process designed by including emerging theory (Soh and Markus, 1995) to explain how the system of the company as a technology creates business value in organizations.

D. The measurement models of successful ERP system

Many models have been developed to evaluate the systems and the success of technology (Davis, 1989b; DeLone and McLean, 2003 1992 Gable et al., 2003; Ifinedo and Nahar,

2006; Sedera and Gable, 2010; Shang Seddon, 2002). These models have been validated empirically by numerous studies in the information system. The results show that many case studies are studied by applying the DeLone & McLean model success using a modeling approach structural equation (Dörr et al., 2013).

However, these models assess the success of three levels of impact. The first is an individual impact (Davis, 1989a) that lights on user behavior. The second level is the group of impact (Gable et al., 2003; Sedera and Gable, 2010) interesting on the working group and its influence on performance, and the third is an organizational impact (DeLone and McLean, 1992)[1]. While one model could evaluate more than one level of impact, for example, and DeLone McLean model takes into account two levels of impact, individual and organizational performance. (Gable, et al., 2003; Ifinedo and Nahar, 2006; Sedera and Gable, 2010) in their models on the success of the extent of ERP, they take into account three levels of impact, the individual impact, 'workgroup impact and the impact of the organization to evaluate the success of the ERP system. Finally Davis in his model of technology acceptance model TAM takes into account a level; individual impact to assess the user perception and behavior.

1) Acceptance Model Technology TAM (Davis, 1989)

This model has been widely used in the information system and considered one of the main theoretical foundations (King and He, 2006). TAM has proven to be one of the most powerful models to explain user acceptance of the technology and user behavior (Wu et al., 2011). Davis says that the use of technology is determined by two factors, perceived usefulness and perceived ease of use, this individual impact is the main purpose of the technology acceptance model. Many studies apply this model to understand the behavior and attitude of the users of the ERP system and assess the satisfaction following use of the system, measuring the latter to use the ERP system is CST Computer Self-Efficacy (Bradford Florin, 2003; Kwahk and Lee, 2008; Scott and Walczak, 2009).

2) DeLone and McLean model of success

D & M is the model city in the success of the information system (Kronbichler et al, 2010; Sedera and Gable, 2010); it is one of the most famous models adopted by researchers to measure success the information in the last two decades system. (Seddon, 1997) in his article respecification extension and D & M Model of the SI, criticized this model about the inclusion of both causation and process interpretations, which lead to the significance of the confusion reduce the value of the model (Seddon, 1997). DeLone and McLean have updated their model based on these criticisms (Delone and McLean, 2003). Despite this update their model, the first version adopted and remains the most cited in the literature review is a success.

The strength of D & M model is its theoretical foundation based both on Shannon & Weaver communication theory and communication systems approach Mason (Mason, 1978; Weaver and Shannon, 1949). They claim that information is considered as an output of an information system that can be measured at three main levels: technical, semantic level and efficiency, referring to the mathematical theory of

communication (Weaver and Shannon, 1949) and its levels to analyze the message following the communication system. Defining and measuring the output of any system is always difficult, especially if the output is rather intangible nature. Information output is represented in symbolic form, the concept of signs is essential to both the information and communication; it is considered a single system the key link in the way affect the other, and involves the system context and the sign its self (Mason, 1978)[13].

Based on these theoretical foundations, D & M developed six categories or aspects of information system that will become the construction of their separate model, these constructions are: system quality (SQ), information quality (IQ) USE, user satisfaction, individual and organizational impact. These variables are most adopted in measuring the success of information in the last two decades system. However, the problem is the model building that tries to combine both causal processes and explanations of IS success (Seddon, 1997) [14]. The result of the combination of variance and process model is that a lot of boxes and arrows may have both a variance and an event in a process of interpretation, giving a feeling of different parts of the model will cause slippage of a direction arrow in a box or another (Seddon, 1994)[2], the subsequent claim that the major difficulties with D & M model can be demonstrated by drawing attention to the use as a construction. This box can take three possible meanings: as a variable that proxies for the benefits of using as the dependent variable in a model variant of the future is to use and thirdly as an event in a process leading to the individual or organizational impact.

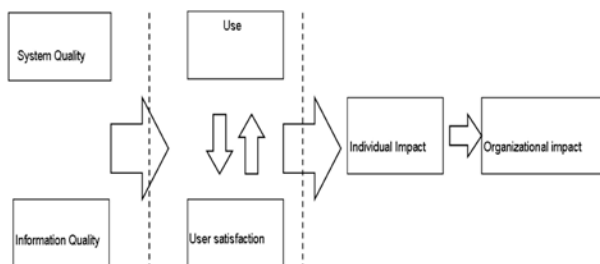


Fig. 1. D&M IS Success Model (Delone& McLean, 1992)

Seddon shows the meaning of the categories in DeLone & McLean model is successful, and explains the combination of three types:

- A model of the variance is success, where quality system and quality of information are considered as independent variables and the dependent variables are the East use and user satisfaction.
- The second model is a variance model to use as a behavior, which can take a second meaning, is to use
- The third model is a process model, where the use is considered an event necessarily precedes the following

constructions: the user satisfaction, individual impact and organizational impact. (Seddon, 1997).

Beyond the combination of the two dimensions of causality and processes to explain the construction and confusion in the direction of D & M model (Seddon, 1997) other considerations would occur as the level considered to explain the success of an information system and the impact on performance. Success is the evaluation is not limited to internal factors according to D & M in their model based on the theory of Shannon and Weaver. For example the quality of the ERP system is not only a causal variable leading to success, but can also be seen as a result of other external factors, such as organization, innovation, and environmental factors (Bradford Florin, 2003; Ifinedo, 2011[15]; Sedera and Gable, 2010)[16].

E. evaluation approaches

Many researchers have tried to understand the relationship between IT investments and performance, focusing on five main approaches for evaluating IT projects (Bellaaj, 2010) [17]. These approaches are:

- Evaluation Approach economic theory (Brynjolfsson, nd): the main objective of this approach is to understand the gap between IT investment and productivity of the organization according to certain economic criteria.

- Evaluation Approach Social Psychology (Davis, 1989a, 1989b; Venkatesh et al., 2003)[18]: beyond the economic approach, it incorporates the human factor as a key factor in the process of IT investment and impact assessment.

- Evaluation Approach Based on the analysis of competition: this approach is developed by (Porter and Millar, 1985) explains how technology affects all business. The authors outline the information technology needs to be understood more than just computers, it must be broadly conceived to encompass information that companies create and use as well as a wide range of technologies more increasingly convergent and linked this process the information in their perception of IT they adopt the concept of the value chain to explain the competitive advantages of IT investments.

- Evaluation Method based on strategic alignment: This approach is developed by (Henderson and Venkatraman, 1993), it is widely used by researchers in the information to understand two key concepts system; the first is the adequacy of the information technology goals and strategic objectives of the organization; the second is the functional integration (integration between business and functional areas). This approach suggests that the IT strategy must be consistent with the business strategy to improve organizational performance.

- Evaluation Process Approach: a new conception of assessment Is success was brought by this approach based on the theory developed by emerging process (Markus and Tanis, 2000; Soh and Markus, 1995) [20]. This approach highlights the failure of the economic model to assess the success, and proposes a new vision of evaluation not only on the input evaluation (assessment of IT investments) on the base, but also based on the use and impacts of IT, by virtue of a valuable creative process.

Three main approaches could be considered to assess the success of the ERP system; the first is based on the financial performance criteria (Nicolau and Bhattacharya, 2006)[19] to assess the benefits of ERP (tangible benefits), the second approach is based on the non-financial approach to evaluating the intangible benefits system ERP, and the last is a mixed approach, for example to assess the ERP system, many perspectives measurement must be taken into account, such as behavioral perspective (user acceptance), the strategic perspective (alignment between strategic organizational goals and ERP), the economic perspective (cost, expenses ..) and (Fit and organizational integration of ERP system) technology perspective. These four dimensions of evaluation of ERP were treated separately in the literature on measuring the success of the ERP system.

In this section, we present two examples of evaluation approaches that synthesize the different perspectives of assessment mentioned above. First, we will propose an AHP approach to assessing performance measures ERP (Tsai et al., 2006) [21]. Second, we will introduce the Balanced Scorecard approach widely adopted by many researchers to assess the benefits of the ERP system (Chand et al., 2005; Rosemann and Wiese, 1999; Velcu, 2010).

1) AHP approach to performance evaluation of ERP

The approach AHP (Analytic Hierarchy Process approach) is to assess the relative importance weight measuring the performance of the ERP; it can be used to select key performance indicators of the ERP system, and explains the contribution of the ERP system in organizational performance (Tsai et al., 2006). This approach is applied to problems of decision making to choose the best and appropriate according to the importance of each alternative. In the case of the ERP system two stages were presented by (Tsai et al., 2006) to assess the relative importance of performance measurement of ERP. The first is to list all performance measurement ERP and assessing their importance. The second stage focuses on building a framework for analysis and AHP achieve weight relative importance of 80 ERP performance measures using a questionnaire with Likert-type scale 7 points (1 = very little importance, 7 = extremely important).

This approach focuses the post-ERP implementation phase. Based on D & M model 1992, this approach provides a new taxonomy of performance measurement: the category of quality, and class of measuring impact. Quality concerns the ERP system information, system usage and user satisfaction, impact category covers both individual and organizational levels. The result of this study shows that a company can select specific performance metrics based on three main factors: the objectives of its ERP system, and the specific needs of the business context. This means that each company must build its primary measure of performance, taking into account three main factors mentioned above.

2) Balanced Scorecard approach to measuring the performance of ERP

This approach is developed by (Kaplan and Norton, 1992)[22] to better understand and classify measures of organizational performance. They claim that the balanced scorecard enables managers to analyze business performance

from four perspectives, the financial perspective, the internal business perspective, the perspective of innovation / learning and finally customer perspective. This part of BSC is widely used in management science in various disciplines to assess organizational performance. However, our focus is on using this approach to evaluate the performance put up by the ERP system. Some researchers have been interested in this question about the evaluation of the performance of the ERP system in a BSC approach (Rosemann and Wiese, 1999; Velcu, 2010). They explain how the BSC approach can be used to evaluate the performance of the company set up by the ERP implementation at both operational and strategic levels.

This application of the BSC sheds some light on understanding the three levels of ERP impact on performance at the operational level, tactical level and strategic level. These levels provide a framework for the analysis of benefits based on the strategy and organizational goals ERP system throughout the ERP lifecycle.

F. The theoretical foundations

First, we present our conceptual model which is based on both theoretical and empirical background. This framework will be considered a success evaluation model of ERP system that combine causal processes and considerations for evaluating the success of the ERP project in three performance levels: The individual performance, the performance of the task force and performance Organizational (and Ifinedo Nahar, 2006; Ifinedo 2011; Ifinedo et al, 2010; Myers et al .., 1997). The levels of analysis included in this model were based on three theories: the first theory is the mathematical theory of communication used by DeLone and McLean is a success model to analyze the quality of the system and its impact on quality information on the one hand, and the impact of the quality of information in the efficiency users, on the other hand; The second theory is the theory of diffusion of innovation used to analyze and classify the different factors in three main boxes: innovation factors, organizational factors and environmental factors; and finally the theory of the structure to analyze the contribution of the ERP technology in organizational performance.

1) Mathematical Theory of Communication

The mathematical theory of communication (Mason, 1978; Weaver and Shannon, 1949)[23] explains the interaction of three factors: the information system, information such as a product and the impact of information on individual performance and organizational. This approach is used by (DeLone and McLean, 1992) in their model of success for developing sexual constructions considered the main variable to evaluate the success of the information system.

2) Innovation diffusion theory

Based on the theory of diffusion of innovation, mainly paradigm variables determining the adoption of innovation (Rogers, 1983)[24], three main factors emerged: Innovation / Technological factors, environmental factors and factors Organization. In this taxonomy, each of these factors can be explained in the context of the ERP system. These factors are extremely important in the adoption of ERP phase and they must be integrated in the process of successful ERP system (no success without adopting one hand technologies).

3) Structuring theory (AST approach)

Structuration theory associated with institutional theory Giddens social assessment has been widely applied to understand and explain organizational technology adoption (DeSanctis and Poole, 1994) [25]. We focus solely on the AST proposed by DeSanctis and Poole, 1994 to explain how technology brings productivity, efficiency and satisfaction to both individuals and organizations. This approach is based on the school of technology was applied and explained by DeSanctis and Poole, 1994 in their approach to the theory Adaptive Structuring. The ASP is considered a framework to study the variation in the change of the organization and illustrating the impact of advanced technology on organizations. It has been tested on a GDSS (Group Support System to the decision) to answer questions about how technology affects people and organizations that use it, and how it improves the performance of the working group.

We consider this approach AST as an extension of the determinant variables paradigm adopting the technology because the technology adoption is an important step in the process of ownership leading to improve performance in the top three levels (individual, group and organizational performance). (Desanti and Poole, 1994) stress the importance played by the members of the organization in the process of choosing the most appropriate technology.

III. DEVELOPMENT OF A MODEL SEARCH

While ERP systems generate several technical problems: integration of ERP solutions with existing applications (legacy systems) or new business software (eg SCM Supply Chain Management, Electronic Data Inter exchange EDI ...) but mainly managerial issues regarding aspects related to the cost, the project period and company organization.

Therefore, if a company wants to incorporate an ERP system, even though its operations are not integrated, it should not, alone, buy a software package and associated computer equipment but it is called, also, to acquire know-how and establish a suitable organization of work.

Therefore, methods of effective use of ERP systems require something other than a good computer. Moreover, several companies say they face serious difficulties in the implementation of an ERP system without the technical aspects are actually involved: this is due, in fact, to disregard and neglect human and organizational factors (Anonyme1, 1997).

Thus, and in support of some researchers (Bancroft, 1996; Kaemmergarrd & Moller, 2000)[26], the factors considered can be classified keys to the success of an engineering change under the under the following dimensions: the involvement of management Generally, user involvement, communication management, training and the implementation strategy that includes both reengineering business processes (BPR) that the same approach of implementation of these systems.

ERP, adding a new dimension, which includes the sub dimensions listed above, turns out advisable to measure user satisfaction of the TSI, including ERP.

At the basis of this reasoning, it is assumed that an ERP system is effective at the individual level where its users are

satisfied. This level of satisfaction is determined by the quality system implemented in the company, a good quality of the information it provides, high value perceived by users and good engineering changes necessary for its implementation.

Thus, the various built the model proposed for measuring user satisfaction of an ERP system, detailed below, may be diagrammed as in Figure 2.

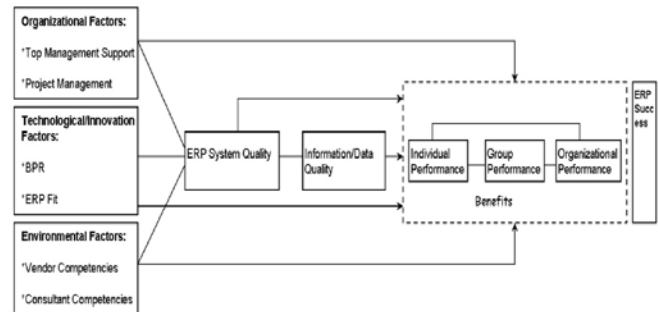


Fig. 2. The conceptual model of measuring user satisfaction of an ERP system

IV. DEVELOPMENT HYPOTHESIS

This dimension has been used extensively in the literature as the dependent variable success SI. DeLone & McLean (1992) fall within a number of 33 empirical studies published between 1981 and 1987 who enjoy success in terms of user satisfaction (Bailey et Pearson, 1983; Ives, Olson & Baroudi, 1983; Doll & Torkzadeh 1988 ...).

Melone (1990)[27] stated that there is no consensus on a conceptual definition of the construct of user satisfaction. Indeed, user satisfaction has been associated with many terms such as "need felt", "acceptance of the system," "perceived usefulness", "appreciation of MIS", "feelings" (Ives, Olson & Baroudi, 1983).

In general, this satisfaction was defined by "the attitudes and perceptions" (Lucas, 1975). In specifically, this satisfaction was defined as the result of the evaluation that individuals are on continuum "content - dissatisfied" (Naylor, Prichard & Ilgen, 1980); or the sum of feelings and attitudes towards each of a variety of factors affecting the situation (Bailey & Pearson, 1983).

However, it should be noted that the definition proposed by Seddon & Kiew (1994) will be the one used for this article because the items related to the "user satisfaction" dimension of their work will be adopted. Thus, satisfaction is defined as feeling "net" of pleasure or displeasure that results from an aggregation of all the benefits that a person hopes to receive from the interaction with the information system (Seddon & Kiew 1994).

A. Quality system *erp*

This dimension is widely used in the literature (Doll & Torkzadeh, 1988; Davis, 1989; DeLone & McLean, 1992; McGill et al., 1999; etc.). It is a powerful determinant of the

effectiveness of IT as well as user satisfaction. The quality of the system relates to the quality of application itself (the different system functionality, ease of use and learning). In addition, it summarizes some issues such as the lack of "bugs" in the system, the user-friendly interface, etc. Therefore, the hypothesis H1 states: "The better the quality of the system (ERP) is good, more satisfaction is high."

B. Quality of information provided by the system erp

The concept of quality of information has been widely used as a key success factor in research in SI. In fact, this construct has been measured primarily by Bailey & Pearson (1983) and Doll & Torkzadeh (1988) as a measure among other satisfaction. This dimension usually includes attributes related to the quality of the information provided by the ERP system, such as the format of the information, clarity of information, accuracy of information, availability of necessary information in real time, the information content, etc. Therefore, the second hypothesis H2 states: "The better the quality of information provided by the system (ERP) is good, more user satisfaction is high."

C. Perceived utility

This construct is defined by Davis (1989)[28] as the degree to which a person believes that the use of a particular system would increase the work performance. This dimension has been considered as a factor affecting the satisfaction of users that it comprises, on the one hand, items related to the perceived ease of use and, on the other hand, those related to the perceived usefulness. Moreover, Davis (1989) shows that the acceptance of a technology depends on perceptions of users of this technology. Indeed, the Technology Acceptance Model (MAT) assumes two types of beliefs, perceived ease of use and perceived usefulness, determine the intent of the individual who influences the use of technology. This allows, therefore, bringing forward the third hypothesis H3 namely: "The greater the perceived usefulness by users, the greater their satisfaction is high."

D. Quality of change

As shown above, this new dimension can be understood by the five under following detailed dimensions.

1) Involvement

To drive change caused by the implementation of an ERP system, it is essential that this project will become the project of the entire company: from top management to operational (Mckerise & Walton, 1995; Bingi, Sharma and Godla 1999; Rivard, 2000; Tomas, 1999; etc.)

a) The involvement of senior management

Indeed, the leaders are not called, only to finance the project but also to take an active role in managing change (Bingi, Godla and Sharma, 1999). This role is mainly to guide the overall operation, encourage local initiative, indicate very clearly the kind of organization that wishes to establish, define the corresponding steps of achievements, etc. (Mckerise & Walton, 1995).

b) The involvement of users

Added to the commitment of senior management and middle management, the implementation of an ERP system can

be conducted only by the involvement of the community of operational users and a user project manager full time representing the whole of this community (Tomas, 1999).

However, it is important to note that the involvement of users could not be, in itself, a prerequisite for the proper conduct of change. The latter requires, in addition, good communication management.

2) Communication

Certainly, the quality of communication within work groups plays an important role in employee attitudes towards change. Where communication and atmosphere were good, new technologies were generally welcomed with enthusiasm, while in groups where members felt compelled to comply with the new rules, reactions were much less favorable. In fact, communication is essential not only to create an understanding and approval of the establishment, but also to win the agreement of users. This communication should begin early, be consistent and continuous (Kaemmergarrd & Moller, 2000).

In addition to good communication during an implementation project of an ERP system, it is inevitable to provide training to users.

3) Training

Training is seen as an important factor to facilitate change in the organization and introduction of new technologies (Mckerise & Walton, 1995). This training aims mainly to prepare staff and help them adapt to their new tasks in order to be successful organizational change. It is not intended; only use new systems but also the understanding of new processes and their integration into the system. Hence, training is an ongoing process and updating a challenge (Bingi, Godla and Sharma, 1999).

4) The implementation strategy of an ERP system

The implementation of an ERP system means a continuous learning cycle in which the organizational process supported by ERP systems is aligned gradually with the company's goals. Lequeux (1999) says: "Far from leading a purely IT project, the adoption of ERP should be an opportunity to reconsider the mechanisms and improve the flow participating in the operation of the business, even to consider a business process reengineering or BPR, Business Process Reengineering".

a) The Business Process Reengineering (BPR)

Moreover, the re-engineering of business processes and implementation of ERP systems are inseparable. They should be carried out simultaneously in order to obtain the best fit between the technologies and processes. This adjustment requires considering the role of ERP systems such as infrastructure, which now support the process and no longer functions and, therefore, improve their organizational effectiveness.

b) The ERP system implementation approach

Akkermans and Helden (2001) have focused on ERP systems implementation approach while trying to show that the incremental approach, scalable, based on continuous improvement is a key success factor in the implementation of a project ERP. They add that users of an ERP system are less satisfied if there was a radical approach (Revolutionary) that

this approach results in a rigid management style based on a high degree of control and command, Intensive use of external experts, even non staff involvement and therefore a loss of skills and know-how internally. Thus, and from the previous development on engineering changes, it was agreed to present the hypothesis H4 on this new dimension, "the higher the quality of engineering change is good, more user satisfaction is better".

This hypothesis derived secondary hypotheses for sub dimensions of engineering change. They are formulated as follows:

- H4a "More DG is involved in the project implementation of an ERP system, more user satisfaction has increased."
- H4b: "More user involvement, the greater their satisfaction is high."
- H4c: "More communication is good, most users are well satisfied."
- H4d: "More training is good, more user satisfaction is very high."
- H4e "The incremental implementation approach can increase user satisfaction more than the radical approach."

V. RESEARCH METHODOLOGY

Once part of the research is defined and the variables of the research are identified, it is important to conduct data collection. For this, a questionnaire, multi-scale, was built and tested with users belonging to both different hierarchical levels as various services, and finally administered face to face in Moroccan companies.

Given that companies have adopted ERP systems are not numerous, it was not possible to focus on a specific industry. The selection of the study population was guided by a single criterion, namely: the existence of an ERP system that is already operating at all levels (all modules are already functional) or at least a good part of the system east. The different ERP vendors (Oracle Applications (Oradist), MFG/Pro (DISCOVERY) ADONIX X3 (STAR ENGINEERING), JDEdwards (LPI)) are chosen as the starting point for the definition of the population.

Data collection has collected a sample of 40 companies surveyed; representing an effective response rate (60.45%). However, it should be noted that the unit of this study is defined as the user of an ERP system. Therefore, the respondent is either the project leader or the leader or one of the senior or middle managers, or one of the last entry clerks. What mattered was the use of the ERP system.

VI. RESULTS AND INTERPRETATION

It is important to note that the measurement scales were either adopted from previous work or created for the need of this research.

A. Descriptive analyzes of research variables: evaluation of measures

After proposing measures to the various concepts identified in the model and collected the data from the selected population, it is appropriate now to ensure the quality of these measures before making adequate statistical treatment. To do this, we made two types of tests for evaluating the measures namely: tests

On the dimensionality and reliability test (Cronbach's alpha) (Evrard, Pras & Roux, 1997). Through these purification tests, which are based on principal component analysis ACP was determined for each building its KMO MSA and each of its items.

So we tried to conclude whether built or not is one dimensional and to specify the contribution of each item to the formation of the factor. Finally, we calculated, for each cleared factor, Cronbach's alpha.

B. Explanatory analyzes of research

Once the measures have been evaluated and the new structures are identified, we proceeded to test hypotheses. This part, devoted to the operationalization of the model and test hypotheses, has identified the following results.

Results thus obtained confirmed the work of DeLone & McLean and those Seddon & Kiew. These results have shown that this satisfaction is explained:

- Primarily by the quality of the system, the quality of information provided by this system and the utility perceived by the users;

- Partially by the quality of engineering changes needed to implement the ERP system. It is true that the data analysis performed could provide only partial verification of this dimension engineering change because, firstly, user involvement, communication and training partially affect that satisfaction on the other hand, the other two sub-dimensions i.e., the involvement of the DG and the implementation strategy does not seem to affect the satisfaction.

It is important to note that future research should be directed towards the new way of "Management Information Systems": evaluating the success of ERP systems. In addition, we can consider further use of research variables in this model.

It seems necessary to consider a more precisely the concept of "quality of engineering change," or override variables to study to eliminate those that are redundant and introduce other variables, such as those relating to culture, user profiles, etc.

So it will be wise to take this model while increasing the sample size to allow better analysis to improve results. This should be possible since the number of Moroccan companies that are in the process of implementing ERP systems is increasing.

VII. CONCLUSION

The implementation of an ERP decomposes different things, which is three time horizons: the front project before implementation of the ERP project: during the implementation of ERP and after project after the failover, ownership of ERP.

Given the rapid spread of integrated software packages in the industrialized world, and minority of scientific publications that provide potential answers to questions on the ERP, a research that helps to explain, prior to the implementation of an ERP and its impact on company performance is probably beneficial for academics and professionals to be more successful the implementation of this new innovation.

In conclusion, it should be noted that in our time, "The information system has become the cornerstone of consolidating the company's strategy" (Baumard & Benvenuti, 1998). Thus, the IS manager is asked "to provide future solutions enabling the company to be more competitive. It is no longer to increase productivity but to provide the general direction the technological know-how through which the company will be able to adapt its service to the needs of its customers while controlling costs "(Baumard & Benvenuti 1998).

Through this article, it is important to note the prominence that ERP systems are currently in Moroccan companies. In fact, these integrated management systems, which are increasingly "backbone" of the SI of the company, need special attention, including in their implementation and evaluation.

Closer to the work of the "Management Information Systems" relating to the determinants of success of IF including the determinants of user satisfaction, the results of this research show that the dimensions outlined in previous studies (Quality System, quality of information and usefulness) remain well determinants of user satisfaction of an ERP system.

However, the current trend concerning the implementation of ERP systems shows that user satisfaction of a SI especially those ERP systems increasingly depends on more than one dimension as well as organizational managerial: he s 'comes to engineering change. Indeed, it seems that there is an increased importance of the effort required in order to conduct an implementation project of an ERP system, mainly in terms of user involvement, communication and training.

This attempt to develop a new model of success of the evaluation of the ERP system is motivated by the need for businesses to justify and understand their investments in this kind of information technology project. Draft ERP system should not be regarded only as a project of upper management, but a project of organization that integrates all actors and stakeholders, for this reason, in our model of assessment of the success of the ERP system, we take into account the role of all partners and actors of different level of analysis and different phases of the integration of the ERP project.

Three categories of assessment factors have been proposed: the organizational factors, environmental factors, and technological factors. These factors are crucial to assess the success of the project of ERP system; they contribute

significantly to understanding the process of the success of the ERP system. The organizations should give more attention to these factors for their successful project of information system and to obtain a high quality system, accepted and used by the employees. As emphasized in our model the success should be evaluated from three main levels of analysis: at the individual level, at the level of the group and of the organization

This research work provides a new tool for the practitioners by allowing them a better understanding of the project of success of the ERP system. The managers of the information system, the direction, and the ERP users need to understand the implication of their actions in the process of success and how they contribute to the improvement of performance. Thus, this work seeks to highlight the suppliers and consultants contributions to perform the ERP project. To cope with more than three-quarters of ERP project fails, organizations must be able to assess their information systems projects. This need led us to investigate this issue by developing a new model that explains the relationship between the partners of the ERP on the one hand, and to propose the main evaluation factors to assess the success of the ERP project.

Notwithstanding, the results presented are limited to enterprises in the sample and should be interpreted with caution in view of the nature and structure of the sample, but also the data collection methods used.

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Vision Based Geo Navigation Information Retrieval

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Abstract—In order to derive the three-dimensional camera position from the monocular camera vision, a geo-reference database is needed. Floor plan is a ubiquitous geo-reference database that every building refers to it during construction and facility maintenance. Comparing with other popular geo-reference database such as geo-tagged photos, the generation, update and maintenance of floor plan database does not require costly and time consuming survey tasks. In vision based methods, the camera needs special attention. In contrast to other sensors, vision sensors typically yield vast information that needs complex strategies to permit use in real-time and on computationally con-strained platforms. This research work show that map-based visual odometer strategy derived from a state-of-the-art structure-from-motion framework is particularly suitable for locally stable, pose controlled flight. Issues concerning drifts and robustness are analyzed and discussed with respect to the original framework. Additionally, various usage of localization algorithm in view of vision has been proposed here. Though, a noteworthy downside with vision-based algorithms is the absence of robustness. The greater parts of the methodologies are delicate to scene varieties (like season or environment changes) because of the way that they utilize the Sum of Squared Differences (SSD). To stop that, we utilize the Mutual Information which is exceptionally vigorous toward global and local scene varieties. On the other hand, dense methodologies are frequently identified with drift drawbacks. Here, attempt to take care of this issue by utilizing geo-referenced pictures. The algorithm of localization has been executed and experimental results are available. Vision sensors possess the potential to extract information about the surrounding environment and determine the locations of features or points of interest. Having mapped out landmarks in an unknown environment, subsequent observations by the vision sensor can in turn be used to resolve position and orientation while continuing to map out new features. In addition, the experimental results of the proposed model also suggest a plausibility proof for feed forward models of delineate recognition in GEO-location.

Keywords—*Vision; Geo-Navigation; Information Retrieval*

I. INTRODUCTION

These days, cell phones, e.g., cellular telephones and individual computerized associates are generally furnished with inserted GPS (Global Positioning Systems) and camera chips. Therefore, visual information connected with land or area labels can be effectively created in our day by day lives. Furthermore, numerous social sight and sound sharing stages, for example, Picasa and Flickr datasets empower clients to tag the areas of their transferred interactive media content on the land map. Furthermore, it is currently possible to derive the certain area labels from the sight and sound metadata like area names, phone numbers, and postal divisions. Such express and

understood geo labeling have brought a wealthy association among the sound substance, virtual sight and our physical world, bringing about an enormous yet steadily expanding measure of "geo-social interactive media" accessible on the web.

This geosocial interactive media has started another door for the scanning, hunt, examination and social mixed media mining, which includes a new "measurement" above the first visual features and data about data, means metadata connection. In the late decade, broad endeavors are given to the figuring, acknowledgment, mining, and investigation of geo-social interactive media, with developing applications going from area acknowledgment to scene rundown, tourism proposal, three dimensional building demonstrating, and city route and so forth. As we trust, the time has desired a convenient overview paper outlining late propel, central issues, and also open inquiries in geo-social sight and sound. In General, social mixed media alludes to the method for collaboration among individuals in which they make, offer, and/or trade mixed media data, e.g. pictures, recordings, sound, writings in virtual groups and organizes. We quickly audit the information source of social mixed media as beneath:

Explicit GPS tagging: A predominant extent of geo-social interactive media is delivered amid the visual substance creation strategy, e.g., amid the photograph catching by utilizing the inserted GPS chips alongside the cameras. Fig. 1 demonstrates some run of the mill gadgets furnished with implanted camera and GPS chip. An extensive asset originates from adjusting the time stamps between GPS gadgets and video streams [1][2].

The implicit geographical tagging: A new significant extent of geo-social media is delivered from the verifiable metadata depiction of sight and sound, from which the geological areas can be parsed utilizing the "area extraction" strategies [3][4][5][6]. Such verifiable portrayals might incorporate, yet not limit to, the URL address prefixes, dialect sorts, and land related things (such as, nation and city names, phone and post numbers). Case in point, picture metadata with "mass of china" demonstrates this picture is taken in Beijing, China.

Community Consensus: Web client group is the significant patron of geo-social sight and sound, which varies from the customary information acquisitions as far as its constantly expanding versatility and uncontrolled quality. From this point of view, the geo-social interactive media additionally mirrors the client inclination or "knowledge". One illustration is that the geo-labeled photographs in Panoramio or Flickr generally display an appropriation predisposition towards famous historic points. Inside of a given topographical area, the visual

insights additionally show a decent agreement to the agent points of interest, and in addition its delegate sees, which is normally the center in shooting for social interactive media clients. In said situations, by breaking down the metadata or visual measurements, the ubiquity of given points of interest or perspectives can be evaluated, that can be further abused with the end goal of touristic suggestion.



Fig. 1. GPS integrated mobile devices

II. LITERATURE REVIEW

Amartya *et al.* [7] emphasize on the design of an unmanned aerial system for handling of post disaster identification, effectively. Image segmentation is the key process in the domain of image processing. Rough set based image segmentation approach has been discussed by Roy *et al.* [8], it provides satisfactory method for the segmentation of image. Rough set theory is an approach to deal with vagueness and uncertainty. Chowdhuri *et al.* [9] presented a review for Rough set based Ad Hoc network. Images are used as data in the area of data mining. Current research work in the field of image mining and various techniques of this field has also been summarized by Dey *et al.* [10]. Watermarking is a security technique which provides data authentication. Acharjee *et al.* [11] proposed an algorithm to integrate the watermarking inside the motion vector. A unique algorithm has been discussed by Sakib *et al.* [12] for solving and exploring any kind of maze. Proposed system of maze mapping was based on coordinate system. For the extraction of shortest path and time Dijkstra's algorithm has been used, in adjacency list representation method whole maze as a graph can be saved. A prototype was developed in Thirumurugan *et al.* [13], of line following robot. Sensor based motors used to line following robot to keep track the line path.

Andersen *et al.* [14] explored the maturity of robot which is used for vacuum cleaning. Its described that vacuum cleaning robot was performed better cleaning as compared to conventional vacuuming. An integrated lane following system based on vision for a robot was presented by Huang and Houshang [15]. LBPE and Hough Transform used for lane position. Experimental results show that lane detection information is very much useful to navigate the mobile robot successfully. Coordinating multi robot system is still a problem in the field of robots. Montano and Suarez [16] proposed a solution to said problem. Proposed decoupled method can coordinate the multi participant robots in on-line mode. Two

robots were used to perform different tests in real environment and graphical simulation to assess the proposed approach.

In Kanayama [17] system for robot task design and management was developed. This system executes all processes from teaching to execution of task. An approach based on vision was proposed by Ismail *et al.* [18] for a mobile robot which follows the line. As a sensor pc camera was used, and via image segregation method image buffers was processed to output compulsory knowledge for the robot's controller. Experimental results shows that robot successfully follow and detect the provided path. Inchoative integration has been done for content based image retrieval by Khan *et al.* [19]. The proposed method just required very cheap hardware like webcam and PC. Based on the related algorithms and proposed approach software system was developed that shows efficiency and effectiveness of the system. Shukla and Tiwari [20] incorporated a approach based on fuzzy logic in the design of an autonomous mobile robot controller for steering and speed control. Portability and transportability are the advantages of proposed system.

A vision based scheme which comprises on two levels was presented for moving a mobile robot to catch a running target by Freda and Oriolo [21]. Nada *et al.* [22] presented a project Teleoperated Autonomous Vehicle. The main purpose of autonomous vehicle was to navigate a predefined route without touching any of the obstacles the vehicle may encounter. A machine vision based line follower robot was provided by Roy *et al.* [23]. A camera was used to image acquisition; algorithms and suitable image processing software were used to process the acquired image, robot was tracked on the basis of generated results. Genetic Algorithm (GA) based system was investigated by Wagner and Hagra [24] to change the type-2 membership functions parameter of interval type-2 Fuzzy Logic Controller purpose was robot control in real environment. By Shaikh *et al.* HMM can use for the content factors segmentation with content based image retrieval in real world [25].

III. APPLICATIONS SCENARIOS

There are expanding measure of various applications identified with geo-social mixed media, which can be quickly arranged into the accompanying four gatherings, i.e., (1) recognition of geographical location, (2) land mark mining and summarization, (3) Three dimensional city navigation and scene modeling.

A. The Geographical Location Recognition

To perceiving the topographical area of the info visual information, for example, a picture or a video arrangement. The acknowledgment is regularly accomplished by close copy visual inquiry from the question to the reference picture or video dataset. Vision-based area acknowledgment can be embraced as an integral or option answer for the GPS-based area acknowledgment. The acknowledgment is accomplished by means of coordinating the inquiry to an arrangement of reference photographs that are connected with land areas. This question might originate from a photograph or a video succession; either caught utilizing the cell phone or transferred by the web client. What's more, the comparable photographs of the question are recognized by utilizing close copy visual hunt

procedures, taking into account that location of the most comparative photo(s) can be distinguished as the acknowledgment yield. Databases that are used in this section are Zubud (Zurich city building dataset), Oxford buildings dataset, SCity contains more than 20,000 photos of street view, PKUBench (Dataset of Peking university).

B. Landmark Mining and Recommendation

Intends to finding delegate historic points from geo-labeled photographs. The mined points of interest can be compressed and prescribed for visual city tourism or online. For this situation, the topographical labels can be either unequivocal or certain. What's more, the mining is principally taking into account investigating the agreement among the connection and substance of such photographs. Currently, there is an expanding enthusiasm for acknowledgment and mining of points of interest from geo-tagged photographs from photograph sharing sites, for example, Flickr and Picasa [26,27,28,29,30]. As a kind of client produced content, these photographs commonly uncover the conduct agreement of online networking clients, bringing about an overwhelming part of milestones in these photographs. From the point of view of the visual appearance, such client agreement additionally compares to the close copy measurements of photographs labeled close-by. Mining milestones from social mixed media represents a wide assortment of utilizations, going from city scene rundown and three dimensional demonstrating to tourism suggestion. Geo-tagged Flickr photos in 20 worldwide capital cities including Boston, Beijing, Chicago, London, New York, Rome etc., Blog photos is a dataset covers social media blogs, Online landmark corpus enables users to upload photos to build three dimensional landmark models online.

C. Three Dimensional Scene Modeling and City Navigation

It plans to remake the three dimensional models, for individual building or for the whole metropolitan city. The prime thought is to do picture correspondence to assess the profundity or three dimensional structures from pictures inside of a sure area, which are recognized utilizing their GPS or area labeling. The vast majority of the world's well known areas have been captured under different conditions, both starting from the earliest stage from the air. It has delivered an extremely rich symbolism of historic points and urban areas empowering three dimensional three dimensional demonstrating and route. Given the three dimensional models of a sure scene or historic point, it turns out to be much simpler for vision-based expanded reality. The increase originates from taking out the past equivocalness in coordinating genuine scenes to two dimensional images. Beneath we survey related work on three dimensional demonstrating and expanded (virtual) reality route by utilizing geo-social sight and sound. City-scale modeling and Landmark-scale modeling are used as geographical modeling.

IV. PROBLEM DOMAIN

In vision based methods, the camera needs special attention. In contrast to other sensors, vision sensors typically yield vast information that needs complex strategies to permit use in real-time and on computationally constrained platforms. Vision is certainly not the only sense nature is using for successful navigation and localization. Some animals, bats for

example, developed sonar-like sensors with remarkable properties. Most accessible for our understanding are daily situations when we merge visual cues with vestibular cues such as accelerations and angular velocities. When looking downwards from a high altitude, we are not able to determine the depth of the scene because of our narrow stereo baseline. Only when we start to move sufficiently we provoke sufficient visual stimulus for depth perception. This phenomenon is also known as Structure from Motion (SfM) in Computer Vision.

V. METHODOLOGY

In geo image mining it is very likely that shadow will be mistaken for objects, since all objects are donated objects.

So geo mining also required to consider for to create the possibility clearly show object for that in this reason work consider two parameter which resolve shadow condition i.e. Brightness and density. Fig. 2 demonstrated geo navigation.

Where

$$\text{Brightness} = \frac{R + G + B + NIR}{4} \tag{1}$$

And

$$\text{Density} = \frac{\sqrt{\#P_r}}{H\sqrt{\sigma_x^2 + \sigma_y^2}} \tag{2}$$

So diameter of square object $\sqrt{\#P_r}$ with diameter of ellp

$$\sqrt{\sigma_x^2 + \sigma_y^2} \tag{3}$$

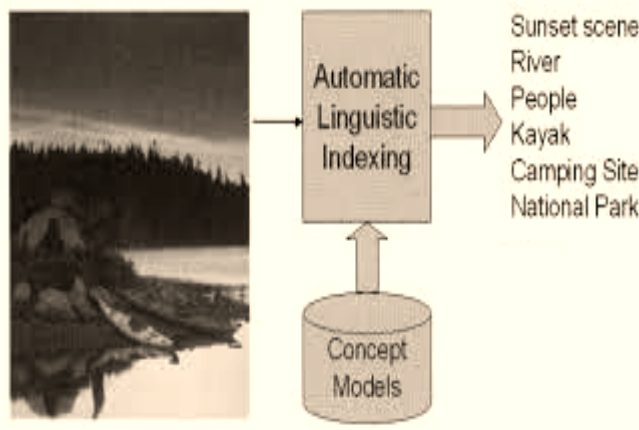


Fig. 2. Geo navigation model

So fuzzy rule used to detect clarity of object in geo map are as

$$\mu_B(x) = \begin{cases} 1 & \text{for } X < M \\ 1 - \frac{x-m}{3\sigma} & 0 \leq x - m \leq 3\sigma \\ 0 & \text{otherwise} \end{cases} \tag{4}$$

$$\mu_D(x) = \begin{cases} 1 & \text{for } x < 1 \\ 1 - \frac{x-1}{.2} & \text{for } 0 \leq x-1 \leq 0.2 \\ 0 & \text{otherwise} \end{cases} \quad (5)$$

So as per above fuzzy function using to detect area, where medium fuzzy function use to detect building detection, low level fuzzy function used to detect vehicle detection. Geo process show in Fig. 3.

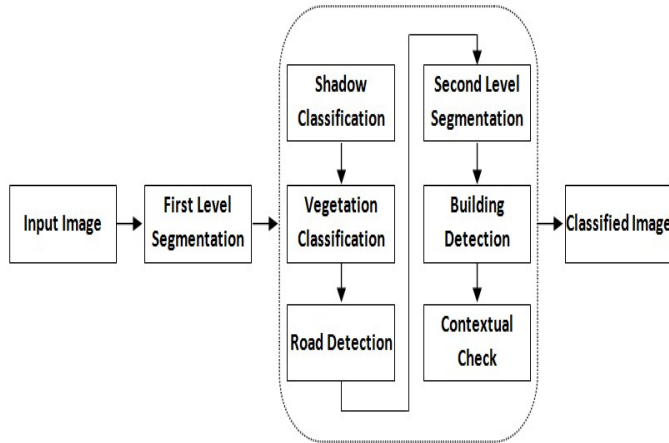


Fig. 3. Geo process

Results and discussions

The whole dataset has been physically annotated to create a complete ground truth for 11 unique landmarks. Each contains 5 queries, 55 queries in total to assess the accuracy of location recognition. One of the accompanying marks are relegated to every picture, i.e.: (1). Excellent: A decent, clear photo of the object; (2). Good: 25% and more of the article is unmistakably visible; (3). Poor: The item is not present; (4). Trash: Less than 25% of the article is visible, or there are elevated amounts of distortion and occlusion. Fig.4 shows the examples of landmark photos from Geo-tagged Flickr photos.



Fig. 4. Flickr photo collection

VI. CONCLUSION

This research work demonstrates that map-based visual odometer strategy derived from a state-of-the-art structure-from-motion framework is particularly suitable for locally stable, pose controlled flight. Issues concerning drifts and

robustness are analyzed and discussed with respect to the original framework. Also this paper proposed a multiple usage localization algorithm based on vision only. One of the principle main impetuses for geological mindful social interactive media is the success of client group, which to a great extent diminishes the broad expense of human work in delivering or gathering the media content and metadata. Such information scale brings another inquiry. In common information scale, because of the information and predisposition, existing methodologies typically embraced parametric forecast models. In any case, whether parametric models are so suitable for displaying the enormous and continually developing information scale is flawed. In such situations, there is more prospect on researching the un-parametric models.

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Contemporary Layout's Integration for Geospatial Image Mining

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Abstract—Image taxonomy and repositioning plays a major role in dealing with large multimedia data on the Internet. Social networks, image sharing websites and mobile application require categorizing multimedia items for more efficient search and storage. Therefore, image classification and retrieval methods gained a great importance for researchers and companies. Image classification can be performed in a supervised and semi-supervised manner and in order to categorize an unknown image, a statistical model created using relabeled samples is fed with the numerical representation of the visual features of images. Analysis of the keywords surrounding the images or the content of the images alone has not yet achieved results that would allow deriving precise location information to select representative images. Photos that are reliably tagged with labels of place names or areas only cover a small fraction of available images and also remain at a keyword level. State of the art of content based retrieval has been analyzed in earth perception image archives concentrating on complete frameworks indicating guarantee for the operational implementation. The methods are taken into consideration, concentrating specifically on the stages after extraction of primitive features. The solutions conceived for the issues such as synthesis and simplification of features, semantic labeling and indexing are reviewed. The approaches regarding query execution and specification are assessed where conclusions are drawn in the research of earth observation mining.

Keywords—Geo-Location; Spatial Layout; Feature Extraction; Image Mining

I. INTRODUCTION

Nowadays, the role of geographic location for images and the connection of web information to a physical location have getting significance attention. Web search is increasingly becoming “local”, search engines not only index content by keywords but also by geographical locations. Users can have information mapped to a certain region-of-interest or place and search accordingly. Web image search, on the other hand, is mostly keyword-based or explores basic image features such as color or texture.

A geospatial Web image search that works on unstructured Web pages with arbitrary images is not yet available. Image search associated with a certain area rapidly demonstrates that Web pictures commonly don't effortlessly uncover their explicit connection to a real geographic position. However, geographic definitions of web pictures are not accessible at all or concealed some place inside of the content of Web page. This absence of explicit area data additionally applies to

pictures contained in Web pages. Despite the fact that GPS has touched base at the consumer, the photographs that really have an area stamp or are geo-labeled cover just a minor portion of all Web pictures. Furthermore, such images are mostly only available in social communities, but seldom find their way into the interesting asset of local Web pages. This represents an intriguing challenge to the field of Web picture retrieval, to be specific, the determination of dependable information of location for Web pictures from disorganized Web pages.

Right now, a Web search photographs of famous place name remains a keyword based search of well-known geographic place names. This can work for known places, landmarks, or regions such as “London” or “Beijing” but cannot bear a precise geo-reference for Web image content. In this context there is need a spatial search of web image to facilitate clients to look for images from a certain geographical coordinate or its spatial neighborhood to complement geospatial Web search engines. In this work, a system for geospatial Web image search has been developed [1]. The aim of this study, at a general approach for common Web images that work beyond keyword search and do not need prepared tags or annotations.

Conceivable location information has been determined for those pictures which exist on the Web pages by a set up spatial connection. The common connection of a content of web page, area, its holds pictures, and related data about data means metadata is a fundamental main thrust behind this methodology. In this study spatial search engine has been utilized for a location-based geo-referenced search down to the address level [2]. It utilizes the information regarding location of a web-page, and it is not unequivocally hold as metadata or other organized annotation yet is fairly a verifiable piece of the textual content.

GEO SPATIAL WEB SEARCH

Querying for Web pages associated to a convinced given location has been allowed by geospatial search. The majority existing search engines are already most proficient for queries based on keywords, but lack geospatial capabilities since the search for geospatial information poses some special requirements that they cannot completely fulfill. For example, certain location-based queries cannot be answered with a purely textual index, such as vicinity queries within a certain radius, results in a specific district of a city, or other geospatial constraints. Furthermore, the result presentation should consider and cater for the geospatial dimension.

Nowadays, location-based Web search and also the services working on prepared geo-referenced information getting very much attention. This is reliable with the high significance of area to the client. On the other hand, the Web contains a massive amount of resources that exhibit geospatial references. Web pages with the required information related with location include home pages of businesses, restaurants, agencies, museums etc. These are excellent sources to answer user's queries for relevant spatial information.

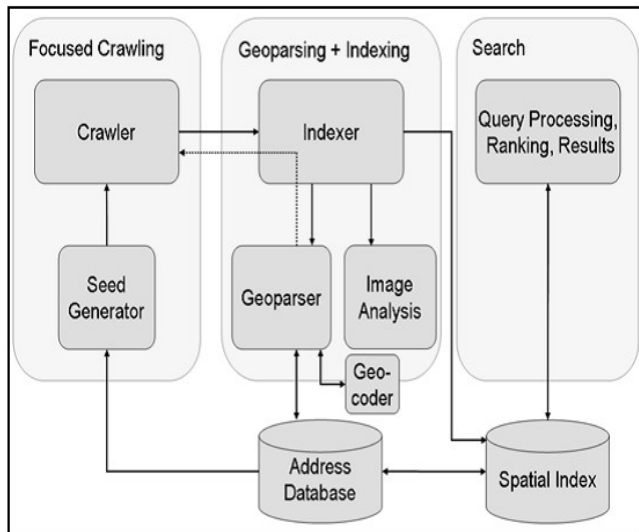


Fig. 1. Spatial search engine

The structural design as shown in Fig. 1 of spatial web search engine prototype mainly comprises on crawling, indexing and search.

II. LITERATURE REVIEW

Images are used as data in the area of data mining. Current research work in the field of image mining and various techniques of this field has also been summarized by Dey *et al.* [3]. Image segmentation is the key process in the domain of image processing. Rough set based image segmentation approach has been discussed by Roy *et al.* [4], it provides satisfactory method for the segmentation of image. Rough set theory is an approach to deal with vagueness and uncertainty. Chowdhuri *et al.* [5] presented a review for Rough set based Ad Hoc network. Karaa *et al.* [6] proposed a novel approach for document clustering MEDLINE based on genetic algorithm. Kausar *et al.* [7] discussed the data mining method applied in the extraction of clinical attribute and classification for diagnosis of cardiovascular patients. In [8] Panchal and Tiwari discussed, analyzed and compared various techniques of image retrieval system based on contents. Some features extraction techniques also described for extraction primitive features like colour, shape and texture. Combinations of these three low level features are also described. Three approaches namely Genetic Algorithm (GA), an Energy Based Model (EBM), and a Binary Integer Programming (BIP), to utilizing spatial contextual information at object level for semantic image analysis are discussed by Papadopoulos *et al.* [9].

A framework based on a number of various combinations of primitive features and algorithms of classification was

developed for the gain of good insight on the use of spatial context. Choudhary *et al.* [10] described the integration of three methods that are Gabor Filter, Spatiogram, and Edge Histogram as a new method. This methodology fundamentally automates the process of retrieving by utilizing image analysis techniques taking into account primitive visual features such as texture, shape and colour with spatial information. Detailed review of various methods has been focused by Priyatharshini *et al.* [11]. Evaluation techniques of such methods used in the current research based on spatial features of image in content based image retrieval systems. In [12] Balasubramani and Kannan discussed about two powerful features of CBIR systems. 1. Edge Histogram Descriptor used for edge distribution 2. Spatial distribution of colours represents by Colour Layout Descriptor (CLD) in an image. More user friendly prototype NWCIBR has been developed and implemented both EHD and CLD features. Various concepts and applications of CBIR have been reviewed by Hirwane [13]. He also presented a technique on the basis of automatically derived primitive features for retrieval of images.

Jiu *et al.* [14] proposed a method to integrate the spatial layout and presented an application to parts classification for the estimation of human pose. Secondly author introduced edge feature as a complement from RGB images used for body parts classification. A method based on a probabilistic spatial context model, as well as individual material detection algorithm was proposed by Singhal *et al.* [15] to determine the scene contents. Spatial context aware material detection method used to reduce misclassification and increase the accuracy of initial classification. Construction of the surface layout of objects, aging a label to the image into geometric classes is discussed by Hoiem *et al.* [16]. Multiple segmentation frameworks were used to provide robust spatial support to contribute to the confidence in each geometrical label. Yadav *et al.* [17] presented a study of the behaviour of several content based image retrieval systems. Various feature extraction texture analysis with representation was also presented.

Shaikh *et al.* [18] proposed a dual layer model SHODHANI, which integrates features and processes and combine neural network approach to sense a specific image and easily retrieved from complex database. Chakraborty *et al.* [19] firstly estimated camera view point and position of the peoples in 3D scene then extraction spatial information from 3D *people* position. Neural network based a novel approach proposed by Khan *et al.* [20] which contributes a novel technical concept for the scene understanding and recognition, which can be important to provide vision intelligence to cloud robot.

The scope of application of intelligent software agents in the domain of web personalization for solving various currently available problems such as user profile learning, management of personalized content adaptive and modeling interactions with users has been described by Singh *et al.* [21]. PCA is a mathematical approach; utilization of principal component analysis in medical image processing has been discussed by Nandi *et al.* [22]. Generally PCA used for image fusion, feature extraction, image segmentation, compression, de-noising, registration etc.

III. PROBLEM DOMAIN

One commonly asked question when confronted with a photograph is “Where is this place?” When talking about a place mentioned on the Web, the question arises “What does this place look like?” Today, these questions cannot reliably be responded in due order regarding Web pictures as they normally don’t unequivocally uncover their relationship to a real geographic position. Analysis of the keywords surrounding the images or the content of the images alone has not yet achieved results that would allow deriving precise location information to select representative images. Photos that are reliably tagged with labels of place names or areas only cover a small fraction of available images and also remain at a keyword level. In this research work representative photographic representations of the mentioned location has been identified. The potential propagation of this location to the images embedded in the page and its reliability is determined by content-based and context-based analysis. In a first step, an analysis of the Web pages identifies photographs, i.e., the image content relevant for a spatial image search for realistic depictions of places. After the identification of photograph, image context and content evaluation can be performed using a location-relevance classification process to determine if the image is originally connected with the location previously recognized for the webpage.

IV. METHODOLOGY

Proposed method for the assessment of image location based on processing chains identifies a probable location-relation of images from website pages. This process is outlined in Fig. 2. The starting step is a web page which is examined all references of image are extracted. After this image is processed and analyzed using a filter chain in the component of photo classification. This component analyzes each and every image and decides whether it meets the requirement to be considered as photo. When condition is satisfied then adjacent page and features of image are given as input for the classification of location.

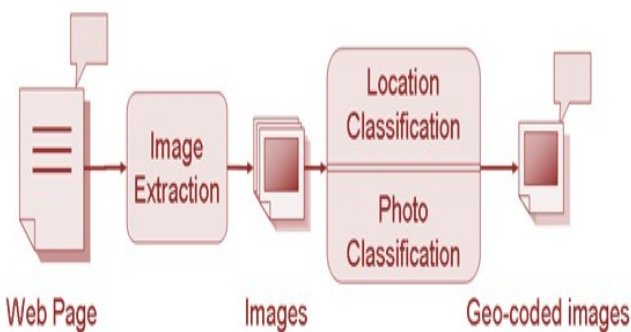


Fig. 2. Geo navigation process

For both location classification and photograph classification different sources of information has been utilized.

A. Photograph classification

Various types of images are used on web pages for different reasons. First huge amount of images are used as illustrative purpose such as, graphics, photographs, sketches,

diagrams, logos, title bars. Second group of images like, bullets menu items, borders, headings, buttons etc. serves as formatting objects. Java applets flash and videos are not considered into this research work because they have different semantics as compared to images. This research study only focuses on photographs being the key possessor of the visual location information and serve illustrative purposes.

Therefore, proposed method identifies images with a high likelihood to be a reasonable photograph of a place. Various features and their relativity for photograph classification are, image size and ratio, size comparison, filename comparison, color histogram analysis, animation, EXIF data, vertical edges, face detection etc. These features are playing very important role for the classification of photographs.

B. Location classification

After photograph classification proposed method has to make connection to the location of the web page images. A picture identified with an area will display certain qualities which exploit to make an arrangement of estimations of how well the area of a Web page can be acquired by the pictures embedded in it. For the execution of the location classification, distance of the picture has been considered to every location on the page and the matching of picture descriptors and key components on the page to parts of the location. An in-page location keyword seek has been utilized.

The examined image tag attributes are name, alt and title, the set of evaluation features are image-attributes, DOM-Distance, Source-Path, Page title and metadata are extremely helpful in location classification.

C. Fused features for location assessment

The procedure appeared before in Fig. 2 portrays two legitimate blocks for location assessment that are location classification and photograph classification. Inside every chain, a few channels serially assess the previously stated features, compute a relevance score, and include an annotation for the assessed picture. Relevance score for *img* per chain is as follows:

$$relevance(img)_{chain} = \sum_{i=1}^n w_i evaluation_i(img) \quad (1)$$

$$veto(img) = \exists i \in o..n : evaluation_i(img) = veto \quad (2)$$

For geo location the overall accuracy of classifier is as per exact situation of their positive or negative situation, where total accurately categorized as follows:

$$= \frac{TP + TN}{TP + TN + FP + FN} \quad (3)$$

At any geo location (R, n) with that X pixel, may include noise and is consider with variance σ^2 as

$$\varepsilon(K, I) \sim N(O, \sigma^2) \quad (4)$$

So Markov process as it condition probability

$$P\{X(S_n) = i | X(S_k)\} \quad (5)$$

Thus

$$X(k, l) = \sum \sum K \sin(Q_r) \quad (6)$$

$$\cos(Q_r) \quad (7)$$

$$\alpha^r \quad (8)$$

This proposed model assumption on the parameter as shown in Fig. 3. Further restriction on the range of the parameter is placed by examining the ability of the model.

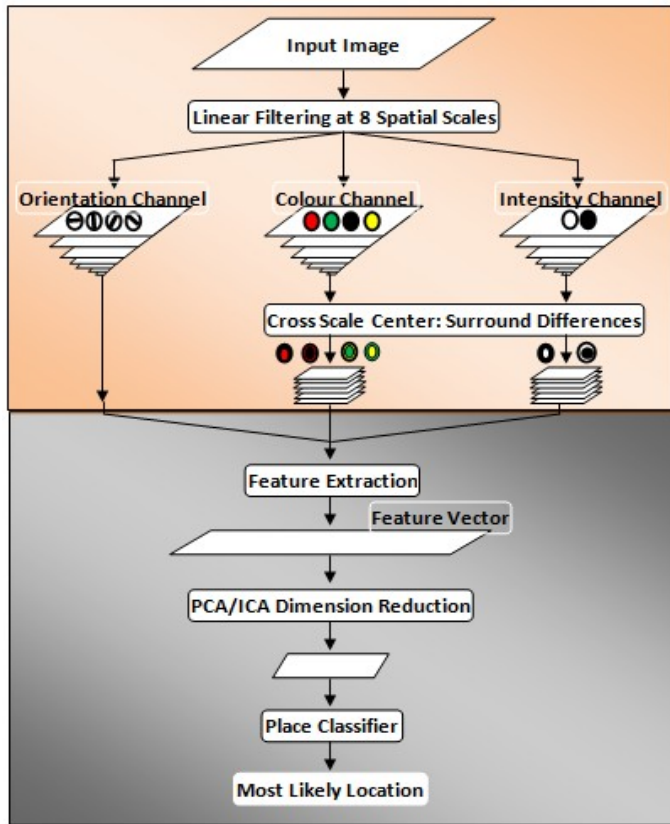


Fig. 3. Proposed process

So similarity condition prefer as per models approach vector as target image T, where

$$Td^2(x, y) = \frac{n_x n_y}{n_x + n_y} (\bar{x} - \bar{y})' A + (\bar{x} - \bar{y}) \quad (9)$$

Where

$$A = \frac{A_x + A_y}{n_x + n_y - 2} \quad (10)$$

So analyze query and tag image are if

$$T^2 \leq \frac{(N_x + N_y - 2)P}{N_x + N_y - P - 1} \quad (11)$$

Where P size.

D. Results and discussions

Series of test crawls on proposed system has been performed to evaluate and assess the validity of proposed approach. Crawl of Rugen (Germany's island) and Oldenburg (Germany's city) has been chosen as a test set. The evaluation of performance has been conducted on crawl of Oldenburg, number of URLs 17250, image references extracted by system 730845. Of these 540825 (74%) has received veto, further 182712 (25%) were vetoed afterwards. Remaining 7308 (1%) received an evaluation score. The evaluation of proposed system demonstrates that the proposed approach has been efficient.

V. CONCLUSIONS AND FUTURE WORK

An automatic method for geo-reference images embedded in webpage on the address level has been proposed in this study. On the basis of location-based search, proposed method and heuristics to propagate the Web page location and assign it to embedded photographs. Proposed approach demonstrates that a fused multi-source analysis of context and content is suitable for this task. On unstructured web pages automatic location identification of pictures now permits pictures to become part of the results of a geospatial Web seek without past manual tagging. The results of proposed methods are promising regarding classification and performance. The present methodology is fairly exploratory, in the future work the outcomes could be made all the more soundly established by comparing diverse classification outcomes by, such as, SVM, neural networks, decision trees and so forth. This could be bolstered by the utilization of more page and features of image for the purpose of classification. The location of a page is as of now known, different sources could be utilized to demonstrate the kind of zone this area is in. Such data could be utilized to authenticate discovered images against a desire for, e.g., urban buildings, landscapes, or rural areas.

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Arabic Stemmer for Search Engines Information Retrieval

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Abstract—Arabic language is very different and difficult structure than other languages, that's because it is a very rich language with complex morphology. Many stemmers have been developed for Arabic language but still there are many weakness and problems. There is still lack of usage of Arabic stemming in search engines. This paper introduces a rooted word Arabic stemmer technique. The results of the introduced technique for six Arabic sentences are used in famous search engines Google Chrome, Internet Explore and Mozilla Firefox to check the effect of using Arabic stemming in these search engines in terms of the total number of searched pages and the search time ratio for actual sentences and their stemming results. The results show that Arabic words stemming increase and accelerate the search engines output.

Keywords—Information Retrieval; Arabic Stemming; Search Engine; Arabic Morphology

I. INTRODUCTION

Arabic language is the one of widely spoken language in the world [21]. It belongs to the Semitic languages branching family of the Asian-African languages Group [8]. The morphology of Arabic poses special challenges to computational natural language processing systems. The exceptional degree of ambiguity in the writing system, the rich morphology, and the highly complex word formation process of roots and patterns all contribute to making computational approaches to Arabic very challenging[6]. Arabic words are formed from abstract forms named roots, the root is the basic form of word from which many derivations can be obtained by attaching certain affixes or suffixes so we produce many nouns and verbs and adjectives from the same root [17,20].

Furthermore Arabic is a highly inflectional language with 85% of words derived from trilateral roots. Nouns and verbs are derived from a closed set of around 10,000 roots [27]. Arabic words are normally derived from bilateral, trilateral, quadri-literal, and pentaliteral verbs. These words represent various modifications of the original verbs. These modifications are represented by special types of measures called templates "اوزان". Arab grammarians use the verb "فعل" 'faal' as the model to represent the templates that can form other verbs and nouns from a specific root. Each letter of the special verb "فعل" has a specific name and meaning applied in the formation of other forms of verbs and nouns, the first letter is called "ف" "fa", the second is "ع" "ain", and the third is "ل" "lam" [26]. Many stemmer techniques for Arabic language

depend on root extraction [20,26]. One of the Arabic language characteristics the formats of the words in Arabic language are the union of templates for the meanings vary in function performed. "فالنظر" "Beholder", "والمنظور" "Perspective", "المنظر" "Theorist" vary in significance with the agreement at the root of the general concept, which is a "نظر" "look".

Stemming has multiple definitions, it is the process of reducing a word to its stem or root form. Stemming is considered by number of authors as word standardization [25]. Shereen Khoja's define stemming as the process of removing all of a word's affixes to produce the stem or root. However, Leah Larkey [19] was more general in her definition stemming refers to any process which conflates related forms or groups forms into equivalence classes, including but not restricted to suffix stripping. Stemming might be useful to Information retrieval systems, text classification systems, text clustering systems, dictionary automation, text compression, etc.

Sharul classify Arabic stemming to four different approaches manually constructed dictionaries used by Al-Kharashi and Evens[4], algorithmic light stemmers which remove prefixes and suffixes which is mentioned by some authors [5,11], morphological analyses which attempt to find roots Several morphological analyzers have been developed for Arabic [3,6,7,10], and statistical stemmers, which group word variants using clustering techniques Statistical techniques have widely been applied to automatic morphological analysis in the field of computational linguistics [9,12,14,15,16].

There are many Stemming techniques for Arabic language , Al-Stem Stemmer developed by Kareem Darwish and modified by Leah Larkey from University of Massachusetts and further modified later by David Graff form LDC, in which (ال, فال, بال, يت, يت, لت, مت, وت, ست, نت, بع, لم, وم, كم, فم, ال, لل, وي, ل) (ي, سي, في, وا, فا, با) are removed from the prefixes and (ات, وا, ون, وه, ان, تي, نه, تم, كم, هن, هم, ها, ي, تك, نا, ين, به, ي, ا) are removed from the suffixes of the word [22]. Aljlayl Stemmer [5] Mohammed Aljlayl developed a light stemmer used for his own information retrieval researches in TREC cross-language track in that stemmer he considered the length of the word for removing suffixes and affixes, also he normalize certain Arabic characters like ا, ا, ا [23]. Light8 Stemmer[19] It is a light stemmer developed by Leah Larkey for the purpose of

researching, the stemmer remove ω if the remainder of the word is 3 or more characters long. Remove any of the definite articles if this leaves 2 or more characters. Remove any of the following suffixes in order from right to left (ي, ها, ان, ات, ون, ين, يه, ه). Light10 stemmer which is a modification of light8 stemmer. Berkeley light stemmer [2] is gives best performance in the track. This stemmer depends on the length of the word if the word is at least five-character long, remove the first three characters if they are one of the following (مال, ال, رسال, لال, وال, بال, فال, كال, ول). If the word is at least four-character long, remove the first two characters if they are one of the following (ل, لل, فا, يا, سي, وم, وت, ال, وي, وا, لا, وب, ول, وس, كا). If the word is at least four-character long and begins with ω , remove it. If the word is at least four-character long and begins with either ب or ل, remove ب or ل, only if, after removing the initial character, the resultant word is present in the Arabic document collection. Recursively strips the following two-character suffixes in the order of presentation if the word is at least four-character long before removing a suffix (ون, ات, ان, ين, تن, تم, كن, كم, هن, يا, ني, وا, ما, نا, هم, يوها). Recursively strips the following one character suffixes in the order of presentation if the character is at least three-character long before removing a suffix (ت, ي, ه). Kadri's linguistic-based stemmer[29] this stemmer depended on the idea that the Arabic word consists of five part their order is; antefixes, prefixes, stem, suffixes and postfixes. The linguistic-based stemmer has two phases, Training Phase a list of stems with its frequency occurrence is built for each word using corpus to avoid ambiguity that may happen when removing affixes. The Stemming Phase where the stemmer truncates possible affixes according to the above list. If there is an ambiguity raised for the stemmer (more than one combination was available), then stemmer selects the most appropriate candidate; according to corpus statistics computed in the training phase. Restrict Stemmer [24] focused on removing conjunctions, و, and ف, and prepositions, ك, ل, و, and ب, that come as prefixes in the beginning of the words. Here didn't mention other affixes such as articles and suffixes in general just tried to find a way to recognize the two types of affixes; the conjunctions and the prepositions. Beltagy Stemmer Samhaa El-Beltagy and Ahmed Rafea [13] have proposed a stemming technique that not only removes prefixes and suffixes from the beginning and the end of the word, but also converts the irregular plural form of the word to its singular form. The stemmer also is a domain specific stemmer which conducts stemming according to the domain of the collection of text to be indexed. The domain specific idea is implemented using a stem list that contains the words and their stems. So, before accepting a stem that produce from a word using stem-based stemmer, the system check whether the produced stem exist in the list or not.

This paper presents Arabic stemming techniques depends on the root of the Arabic word. six Arabic sentences and their stemming results from this technique are used to see the effect of the Arabic stemming usage in the famous search engine Google Chrome, Internet Explore and Mozilla Firefox in terms of the number of the web pages and the time taken by these search engine for the original sentences and the stemming results of this sentences.

The rest of this paper is organized as follows: section II presents the related works ; section III introduce the proposed Arabic stemming techniques, section IV shows the results and discussions; and finally section V concludes the work.

II. RELATED WORKS

Considerable research on stemming and morphological analysis is tedious for the Arabic language, but no standard IR-oriented algorithm has yet emerged. Furthermore there is a good effort done to achieve standard Arabic stemming algorithm, here we show the researchers efforts. Aitao Chen and Fredric Gey [2] developed one MT-based Arabic stemmer and one light Arabic stemmer. The Berkeley light stemmer worked better than the automatically created MT-based stemmer. The experimental results show query expansion substantially improved the retrieval performance. Mohamad Ababneh, Riyad Al-Shalabi, et [20] develop a rule-based light depends on truncation of affixes and introduced a new algorithm uses a set of rules to determine if a certain sequence of characters is part of the original word or not and this helped solving some ambiguity problems, also they introduced a way for handling the majority of broken plural forms and reducing them to their singular form which helped to group words of the same meaning in a common form. Kazem Taghva, Rania Elkhoury, Jeffrey Coombs [18] introduce an Arabic Stemming Without A Root Dictionary their experiment shows that stem lists are not required in an Arabic Stemmer. Riyad Al-Shalabi, et [26] build Arabic stemmer based on Excessive Letter Locations, The stemmer find the trilateral root, quadri-literal root as well as the pentaliteral root for Arabic words based on excessive letter locations. The algorithm locates. The goal of stemmer is to support natural language processing programs such as parsers and information retrieval systems.

III. ARABIC STEMMING TECHNIQUE

The morphology complexity of Arabic makes it particularly difficult to develop natural language processing applications for Arabic information retrieval. Stemming is another one of many tools besides normalization that is used in information retrieval to combat this vocabulary mismatch problem. The proposed Arabic stemming algorithm for our research is constructed from the following process:

- 1) *Deleting of Arabic stop words*
- 2) *Striping of diacritics*
- 3) *Striping of affix (prefixes and suffixes)*
- 4) *Determining the stem*
- 5) *Recoding the stem*
- 6) *Verifying the stem with a dictionary of root words*

Step1: The proposed algorithm eliminates stop words from the document using the Arabic stop words list which contains 162 words like , في ' لم ' قد, من ,

Step2: Normalize the rest of the words: such as removing punctuation, converting words to lowercase, stripping numbers out, as following:

- Remove punctuation.
- Delete numbers, spaces and single letters.

- Convert letters (ا), (ء), (ؤ), (ة) into (ا), and (ة) into (ة).

Step3: Applying techniques to find the basic root of Arabic words by removing affixes (suffixes and prefixes) attached to its root. Prefix like (لال, فال, كال, بال, وال, ال) and suffix like (ان,ها, ي, هو, ون, ين, ات, ية, يه, ان,ها)

Step4: The stemming process begins by processing a word and trying to find its correct stem. In case the word does have a correct stem, then the word without its affixes will be returned. The stemming algorithm will take as input an Arabic word (not a stop word), and the output will be the extracted root (or stem). In cases where the algorithm cannot find a root for the specific word, the word itself will be taken as a root. Such cases are few and it depends on the quality of the algorithm proposed

Step5: The recoding module main concern is to change some of the letters to their correct form. These changes will probably occur during the process of template formation when a word is formed from a root. Some letters may be dropped, changed or replaced by other letters. Table 1 lists some of the most recorded Arabic letters. The following is an example of letter recording for Arabic words.

TABLE I. MOST RECORDED ARABIC LETTERS

Word	Recoding Rule (from→ to)	Word after Recoding
هزى	ؤ → ئ	هزؤ
هزا	أ → ئ	هزأ
نيا	أ → ئ	نؤا
خطى	أ → ئ	خؤا
خسى	أ → ئ	خؤسا
صبي	أ → ئ	صؤبا
سبي	أ → ئ	سؤبا
نبا	أ → ء	نؤبا
دنى	أ → ي	دؤنا
تؤمن	أ → و	تؤمين
ؤمر	أ → و	ؤمؤر
راد	و → ا	رؤد
حيا	ي → و	حؤي

Step6: By the use of stemmers, Words in the collection must be organized into groups, multiple errors are produced and may be used to compare and evaluate stemmers. If the two words belong to the same class of development in meaning and been changed to different origins, this is considered error of under-stemming (i.e. too much of the expressions or terms removed. The stemmer went on correct, if they are changed to the same origin, this will be considered as an over-stemming (i.e. not much of the expressions or words are removed).

IV. RESULTS AND DISCUSSION

Table 2 shows the stemming results of six Arabic sentences varying from sixteen words to two words using the proposed Arabic stemming technique.

TABLE II. ARABIC STEMMING RESULTS

Sentence	Actual Text	Stemmer Text
S1	يختلف اسنان عصر العلم والتقنية الثورة العلمية الخلاقة عن إنسان العصور السابقة في عملية الإبداع والتمكن	خلف سنن عصر علم قنأ ثور علم خلق أنس عصر سبق عمل بدع مكن
S2	طور التعليم في المملكة العربية السعودية مقارنة بدول العالم الاسلامي	طور علم ملك عرب سعد قرن بدل علم سلم
S3	مشاكل العلاقات الدولية وأثرها على مداخل المواطنين العربي	شكل علق دول أثر دخل وطن عرب
S4	طرق التحاق الطالب بالجامعات في اوربا وامريكا	طرق لحق طلب جمع ورب وامريكا
S5	حليل وتصميم انظمة الحاسب الآلي	حلل صمم نظم حسب ابي
S6	جامعة نجران	جمع نجر

From the stemming results it is clear that each word returns to its root. The actual sentences and their stemming results are used to see the effect of the Arabic stemming in the famous search engines Google Chrome, Internet Explorer and Mozilla Firefox in terms of total number of the web pages and the Search Time Ratio (STR) which is calculated by the equation 1.

$$STR = \text{Search Time} * 1000/\text{total searched page} \quad (1)$$

TABLE III. INTERNET EXPLORER SEARCH RESULTS

Internet Explorer				
Actual Sentence			Stemmer Result	
Sentence	Number of searches pages	STR	Number of searches pages	STR
S1	2,290	0.519650655	21,300	0.04835681
S2	171,000	0.004502924	549,000	0.0017122
S3	284,000	0.002640845	616,000	0.00137987
S4	413,000	0.002033898	467,000	0.00152034
S5	467,000	0.00117773	692,000	0.00114162
S6	711,000	0.000675105	86,100,000	5.3426E-06

Table 3 shows the results of the search for actual sentences and their stemming results on the Internet Explorer. From the table the number of searched pages using stemming results is very high compared with the actual sentences results, while the search time ratio for stemming is less than the actual sentences. This means the stemming accelerates and increase the search output in internet Explorer.

Table 4 shows the results of the search for actual sentences and their stemming results on the Google Chrome. From the table the number of searched pages using stemming results is very high compared with the actual sentences results, while the search time ratio for stemming is less than the actual sentences.

TABLE IV. GOOGLE CHROME SEARCH RESULTS

Google Chrome				
Actual Sentence			Stemmer Result	
Sentence	Number of searches pages	STR	Number of searches pages	STR
S1	2,300	0.308696	21,300	0.032394
S2	171,000	0.003626	548,000	0.000949
S3	611,000	0.000753	615,000	0.000813
S4	216,000	0.002546	465,000	0.00086
S5	627,000	0.000829	679,000	0.000766
S6	716,000	0.000656	87,100,000	3.9E-06

Table 5 shows the results of the search for actual sentences and their stemming results on the Mozilla Firefox. From the table the number of searched pages using stemming results is very high compared with the actual sentences results, while the search time ratio for stemming is less than the actual sentences. The search time ratio for Mozilla Firefox is less than Internet Explorer for actual and stemming result even the total search page is comparable, while Mozilla Firefox results is comparable with Google Chrome for the total searched pages and the search time ratio.

TABLE V. MOZILLA FIREFOX SEARCH RESULTS

Mozilla Firefox				
Actual Sentence			Stemmer Result	
Sentence	Number of searches	STR	Number of searches	Search Ratio Time
S1	2,280	0.22807	21,300	0.030986
S2	189,000	0.004074	547,000	0.000951
S3	548,000	0.001113	615,000	0.000732
S4	217,000	0.002442	467,000	0.000835
S5	627,000	0.000797	679,000	0.000736
S6	716,000	0.000712	87,100,000	3.21E-06

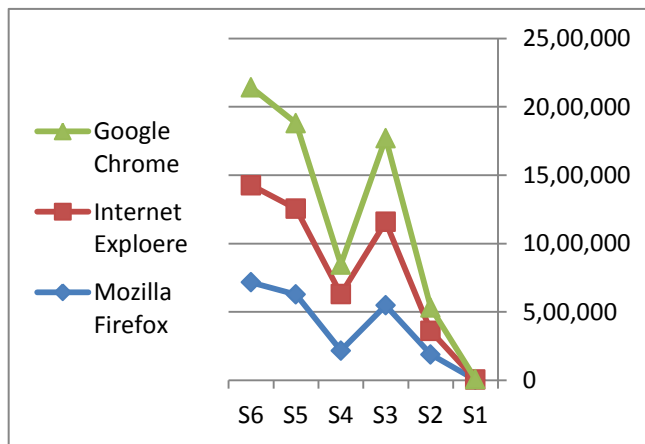


Fig. 1. Actual sentences search results

Figure 1 shows the output of the three search engine for the actual sentences in terms of total searched pages. From the figure it is clear that Google Chrome outperforms Internet Explorer and Mozilla Firefox in the number of searched pages for the actual sentences.

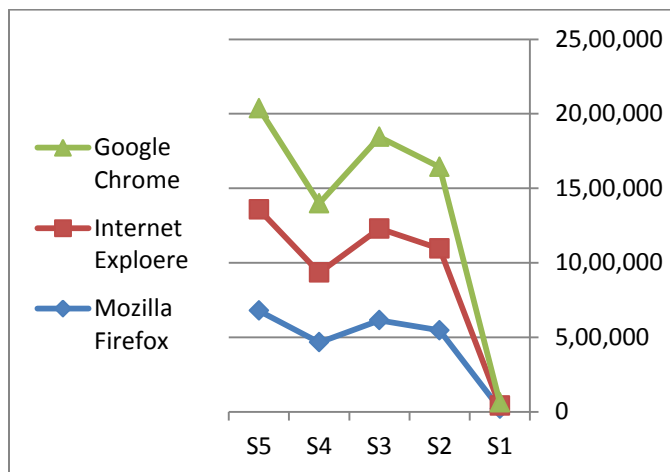


Fig. 2. Stemming results search pages

Figure 2 shows the output of the three search engine for the stemming output. Also Google Chrome outperforms Internet Explorer and Mozilla Firefox in the number of searched pages for the stemming results.

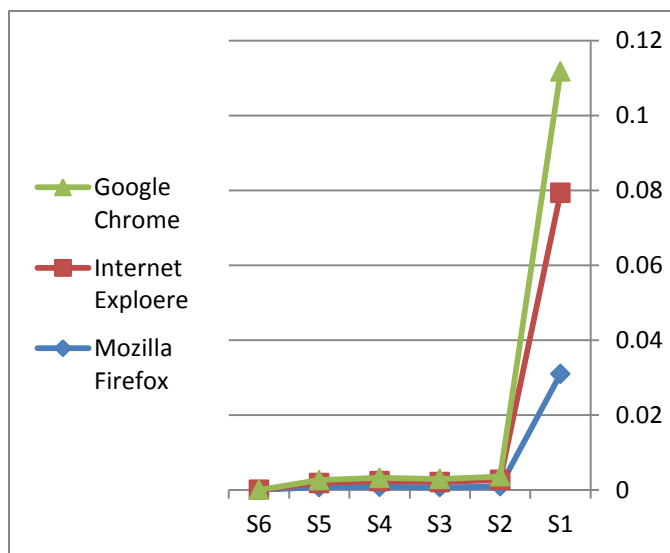


Fig. 3. The STR for the stemming results search pages

Figure 3 shows the STR of the three search engine for the stemming results. It is clear Mozilla Firefox takes short time to perform the search output than the Google Chrome and Internet Explorer.

V. CONCLUSION

The fact that there is a lack of usage of Arabic stemming in the search engines gave the underlying rational for conducting the study presented in this paper. The paper presents the usage of Arabic stemming increases and accelerates the search engines output, where Google Chrome outperforms Internet Explorer and Mozilla Fire fox in terms of total number of searched pages.

ACKNOWLEDGEMENT

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Implementation of a Neural Network Using Simulator and Petri Nets*

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Abstract—This paper describes construction of multilayer perceptron by open source neural networks simulator - NeuroPh and Petri net. The described multilayer perceptron solves logical function "xor" - exclusive or. The aim is to explore the possibilities of description of the neural networks by Petri Nets. The selected neural network (multilayer perceptron) allows to be seen clearly the advantages and disadvantages of the realizing through simulator. The selected logical function does not have a linear separability. After consumption of a neural network on a simulator was investigated implementation by Petri Nets. The results are used to determine and to consider opportunities for different discrete representations of the same model and the same subject area.

Keywords—neural networks; simulators; logical or; petri net

I. INTRODUCTION

In the beginning we will examine Petri nets and their possible applications. In this paper we will emphasize precisely modeling power of Petri nets, as will examine their ability to present neural networks.

The simulator selected for implementation to the neural network in this study is NeuroPh, which is Java - based, object - oriented simulator. NeuroPh is also open-source and it offers many opportunities for different architectures of neural networks [8]. NeuroPh is lightweight frameworks allowed to simulate neural networks and can be use basic for the development of standard types of neural network architectures. It contains well designed open source library and a small number of core classes that correspond to basic concepts in neural networks. There is a good graphics editor to quickly build java - based components of neural networks.

One thing that makes Petri nets interesting is that they provide a balance between modeling power and analyzability: many things one would like to know about concurrent systems can be automatically determined for Petri nets, although some of those things are very expensive to determine in the general case. Several subclasses of Petri nets have been studied that can still model interesting classes of concurrent systems, while these problems become easier [1].

Since firing is nondeterministic, and multiple tokens may be present anywhere in the net (even in the same place), Petri nets are well suited for modeling the concurrent behavior of distributed systems [3, 7]. Petri nets are state-transition systems that extend a class of nets called elementary nets [5, 6]. Unless an execution policy is defined, the execution of Petri nets is

nondeterministic: when multiple transitions are enabled at the same time, any one of them may fire.

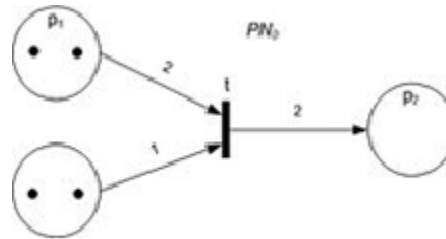


Fig. 1. Petri Net with an enabled transition

In the given diagram of a Petri net [3], the place circles may encompass more than one token to show the number of times a place appears in a configuration. The configuration of tokens distributed over an entire Petri net diagram is called a marking. In the diagram of a Petri net, places are conventionally depicted with circles, transitions with long narrow rectangles and arcs as one-way arrows that show connections of places to transitions or transitions to places. If the diagram were of an elementary net, then those places in a configuration would be conventionally depicted as circles, where each circle encompasses a single dot called a token. Syntactically a Petri net is described by graph of the network. There is many alternative definitions. The following formal definition is loosely based on [2]. A Petri net graph (called Petri net by some, but see below) is a 3-tuple, (S, T, W) where

- S is a finite set of places
- T is a finite set of transitions
- S and T are disjoint, i.e. no object can be both a place and a transition

$$W: (S \times T) \cup (T \times S) \rightarrow \mathbb{N} \quad (1)$$

is a multiset of arcs, i.e. it assigns to each arc a non-negative integer arc multiplicity (or weight); note that no arc may connect two places or two transitions.

Here, in this definition we have conditions mainly for sets S and T.

The flow relation is the set of arcs:

$$F = \{(x, y) | W(x, y) > 0\} \quad (2)$$

In many textbooks, arcs can only have multiplicity 1. These texts often define Petri nets using F instead of W. When using

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this convention, a Petri net graph is a bipartitemultigraph(S U T, F)with node partitions S and T.

The preset of a transition t is the set of its input places:

$$t = \{s \in S \mid W(s, t) > 0\} \tag{3}$$

itspost set is the set of its output places:

$$t = \{s \in S \mid W(t, s) > 0\} \tag{4}$$

Definitions of pre- and post-sets of places are analogous.

A marking of a Petri net (graph) is a multiset of its places, i.e., a mapping M: S → N .We say the marking assigns to each place a number of tokens.

Here we can introduce one more definition of Petri nets. As you can see, this definition is now based on the concept of graph of the network: A Petri net (called marked Petri net by some, see above) is a 4-tuple (S, T, W, M₀), where

- (S, T, W) is a Petri net graph;
- M₀ is the initial marking, a marking of the Petri net graph.

In words:

- firing a transition t in a marking M consumes W(s, t) tokens from each of its input places s, and produces W(t, s) tokens in each of its output places s
- a transition is enabled (it may fire) in M if there are enough tokens in its input places for the consumptions to be possible [4]., i.e. if.

$$\forall s: M(s) \geq W(s, t) \tag{5}$$

In this study first will be implemented neural network through simulator for neural networks and then will be used Petri nets. It has been made realization of a neural network of logical function exclusive or (XOR). This standard is a logical function which can be realized with NeuroPh and it is suitable for realization with Petri nets. This results from values, which suggests the logical function - 0 and 1. They are relatively simple and suitable for different types of presentations.

II. METHODOLOGY

In the beginning of the study we will realize neural network through neural network simulator. The neural network which we have choose for the study implemented a simple logical function “exclusive or”. This logical function is linearly inseparable. The neural network which we are building in this case is a multilayer perceptron. We choose this type of neural network, because linear inseparable function can be realized with a single-layer perceptron. Table 1 shows the essence of the logical operation “exclusive or”.

TABLE I. EXCLUSIVE OR

A	B	A XOR B
0	0	0
0	1	1
1	0	1
1	1	0

Commonly used artificial neural network simulators include theStuttgart Neural Network Simulator(SNNS), Emergent, JavaNNS, Neural LabandNetMaker. For this study is selected the simulator NeurophStudio. This simulator has the potential to realize the different algorithms for training neural networks and arbitrary architectures. This neural network implements logical operation “exclusive or”. There are two input neurons, one neuron in the hidden layer and one output neuron. Figure 3 shows a model of the neural network implemented on NeurophStudio. In this structure of the neural network is used some results from "Research of simulators for neural networks through the implementation of multilayer perceptron". [9].

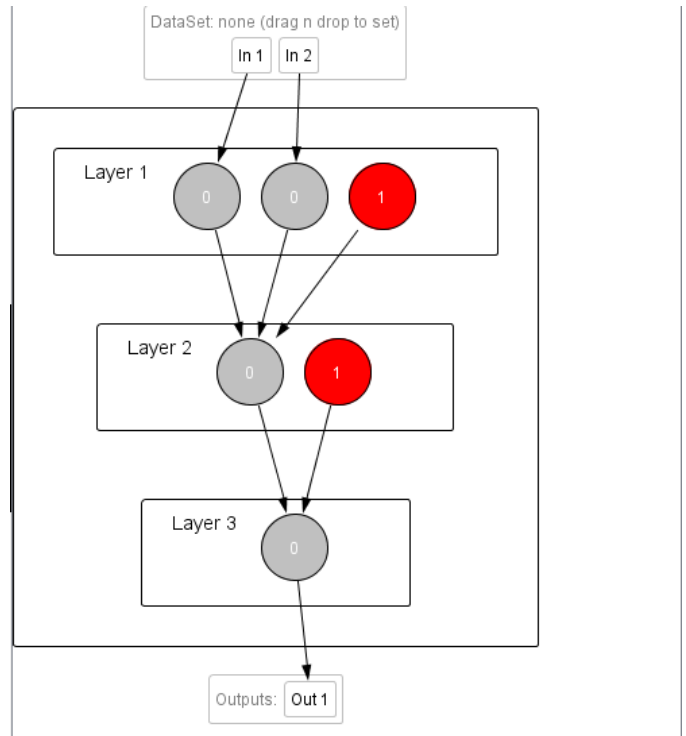


Fig. 2. Neural network which realizes a logic operation XOR (simulatorNeurophStudio)

The training of the network is performed using the truth table of the logical operation "exclusive or", including following parameters:

Max error: 0.01

Learning rate: 0.2

It can be seen that the network is fully trained after 4000 iterations. After transferring this threshold begins a process of re-training and error begins to grow again.

Once the network is trained, we ask sample inputs - 1; 1. At testing network with a sample input 1; 1 result is the following - the value of neuron in the hidden layer is 0,781; value of the neuron in the output layer is 0.263.

This is shown in Figure 4.

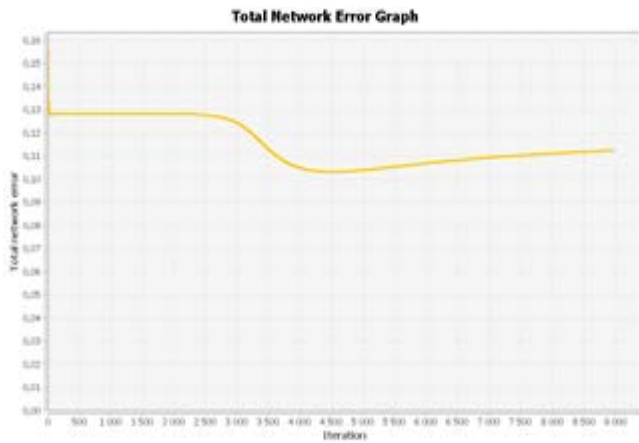


Fig. 3. Total Network Error Graph

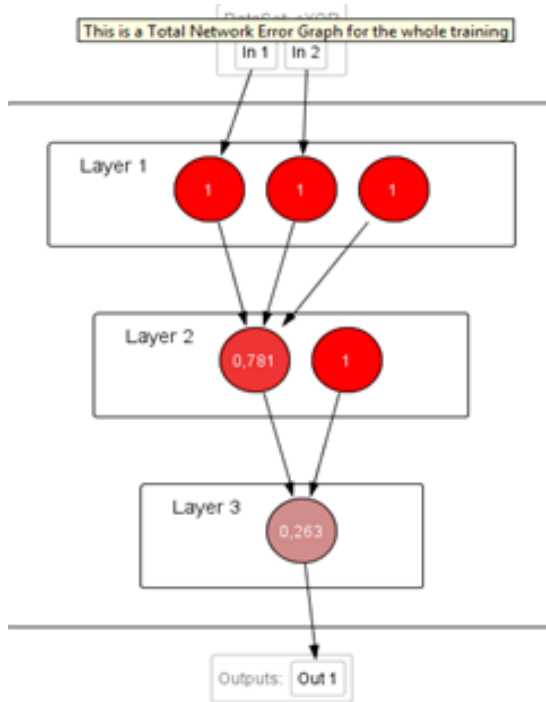


Fig. 4. Results for inputs - 1;1

We testing network with a sample input 1; 0. The result is following - the value of neurons in the hidden layer is 0,839 and 0,998; value of the neuron in the output layer is 0.871.

This is shown in Figure 5.

We continue the study with another neural network. To see the difference in test cases, we will recreate the neural network by another type multilayer perceptron. This multilayer perceptron will decide the same task like the neural network above. The new neural network will have again one hidden layer, but with two neurons in it.

Figure 6 shows a model of a neural network implemented the same logical function. There are two input neurons, two neurons in the hidden layer and one output neuron.

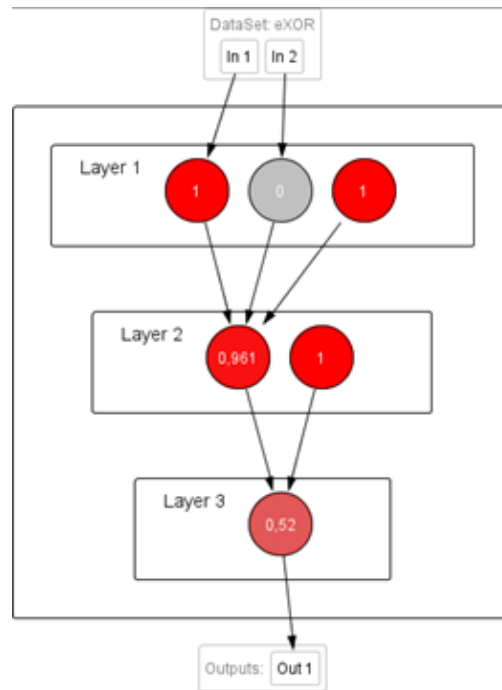


Fig. 5. Results for inputs - 1;0

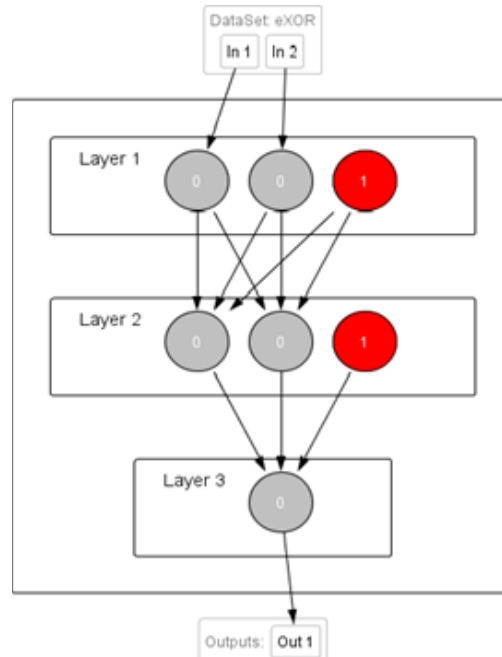


Fig. 6. Neural network, which realizes a logic operation XOR, including two neurons in hidden layer

It can be seen that the network is fully trained after 1900 iterations.

Once the network is trained, we ask sample inputs - 1; 1. In network testing with a sample input 1; 1 result is the following - the value of neurons in the hidden layer is 0,101 and 0,902; value of the neuron in the output layer is 0.125.

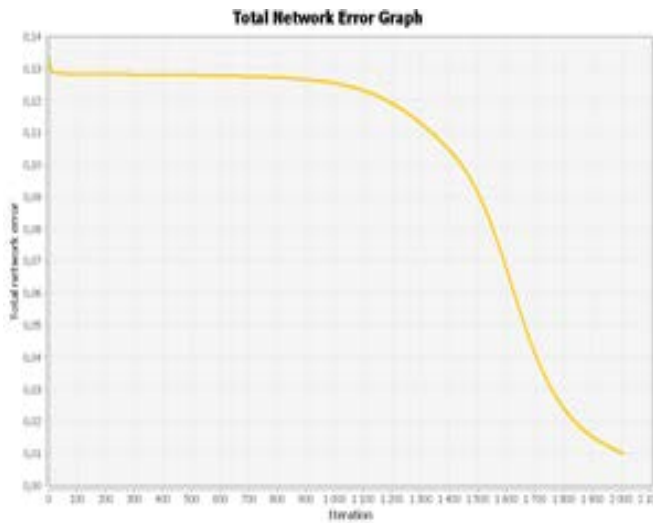


Fig. 7. Total Neural Network Graph

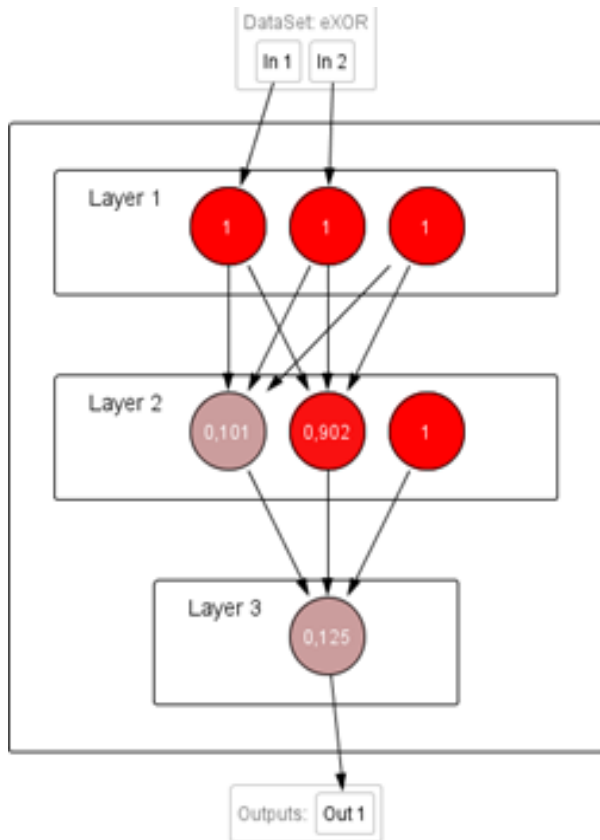


Fig. 8. Results for inputs - 1,1

We testing network with a sample input 1; 0. The result is following - the value of neurons in the hidden layer is 0,839 and 0,998; value of the neuron in the output layer is 0.871.

It can be seen that the results for the second architecture of the neural network are very good. This is particularly in the case when the neural network should give as a result 1. Thus we did tests of two input suit in the simulator for Neural Networks.

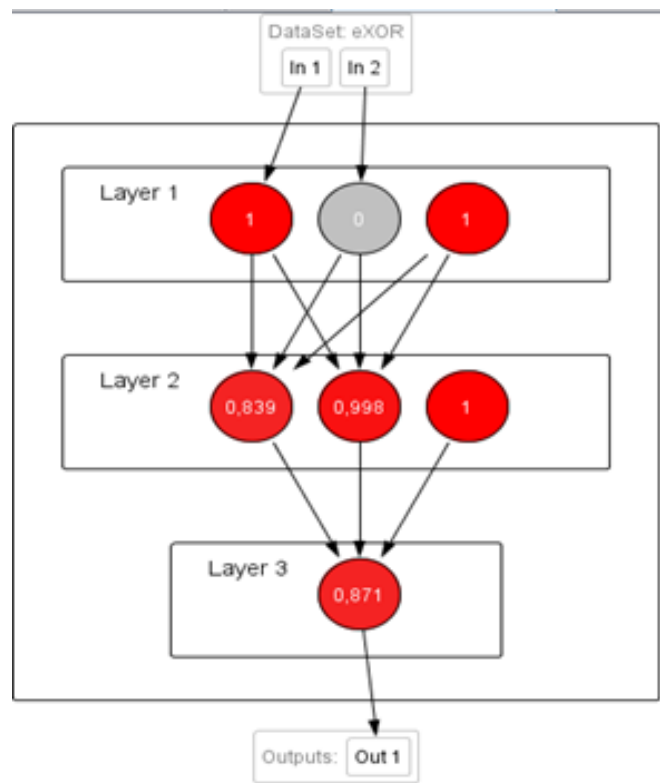


Fig. 9. Results for input - 1;0

We make a comparison between the graph in the simulator and the description of graph by Petri net. To create a graph of Petri nets we must define sets S , T , M_0 . Let's first of all to stick to the first model of the architecture of the neural network which has one neuron in the hidden layer. In this case the set S is consists of the input neuron, the neuron of the hidden layer and the output neuron. Especially for values in the set S we take an example pairs of logic function "exclusive or", two of which we already discussed in examples for simulator NeurophStudio. So far we have two values in this set. We added also and the output values - 0 or 1 for each example, as well as the value of a hidden neuron. The value of the hidden neuron is rather rounded. The set T consists of three connections between neurons - between the two input neurons and neurons of the hidden layer (2) and the connection between neurons of the hidden layer and the output neuron (1). The set M_0 is the initial initialization of the neural network weights. To build a graph of Petri nets we consider also sets P . These sets contain specific examples and actually reflect the positive examples of T . This set contains a different number of elements for each example and includes input and output values which are equal to 1. This is actually 1 of the set T (excluding the value of the hidden neuron), which is subsequently recognized as a marker in the graph of Petri nets.

For the four cases of logical function can be built eight Petri nets:

- The first of them: $S - 0; 0; 0$
- M_0 - random initial weights of links
- T - Weights of links to specific iteration
- Set P_1 - 0 elements.

Second: S - 0; 1; 0; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P2 - 1 element.
Third: S - 1; 0; 0; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P3 - 1 element.
Fourth: S - 1; 1; 0; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P4 - 2 elements.
Fifth: S - 0; 0; 1; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P5 - 1 element.
Sixth: S - 0; 1; 1; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P6 - 2 elements.
Seventh: S - 1; 0; 1; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P7 - 2 elements.
Eighth: S - 1; 1; 1; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P8 - 3 elements.

Making a comparison between the two architectures of both established neural networks. Let us now discuss the case with the second architecture of the neural network. Here we have two neurons in the hidden layer. This means that the set S is increased by 1 unit, and multitudes T and M₀ increased double. Therefore, there could be built 16 major Petri nets.

The first of them: S - 0; 0; 0; 0; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P1 - 0 elements.
Second: S - 0; 1; 0; 0; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P2 - 1 element.
Third: S - 1; 0; 0; 0; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P3 - 1 element.
Fourth: S - 1, 1, 0, 0, 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P4 - 2 elements.
Fifth: S - 0; 0; 1; 0; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P5 - 1 element.
Sixth: S - 0; 1; 1; 0; 1
M_o - random initial weights of links
T - Weights of links to specific iteration

Set P6 - 2 elements.
Seventh: S - 1; 0; 1; 0; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P7 - 2 elements.
Eighth: S - 1; 1; 1; 0; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P8 - 3 elements.
Ninth: S - 0; 0; 0; 1; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P9 - 0 elements.
Tenth: S - 0; 1; 0; 1; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P10 - 1 element.
Eleven: S - 1; 0; 0; 1; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P11 - 1 element.
Twelve: S - 1; 1; 0; 1; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P12 - 2 elements.
Thirteen: S - 0; 0; 1; 1; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P13 - 1 element.
Fourteen: S - 0; 1; 1; 1; 1
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P14 - 2 elements.
Fifteen: S - 1; 0; 1; 1; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P15 - 2 elements.
Sixteenth: S - 1; 1; 1; 1; 0
M_o - random initial weights of links
T - Weights of links to specific iteration
Set P16 - 3 elements.

In the case when the plurality of T have different values (0 and 1, 1 and 0) is obtained activation of the neuron of the output layer and the output is 1. In other cases, the output of the network is 0. This is actually case when neural network implements logical "exclusive or" function.

The sets T and M_o in Petri nets could give suggestion for the convergence of the network: how fast neural network will be trained, whether the training set is appropriate, how many test to be used, etc.

We can build a graph of the petri nets to the first example of architecture of the neural network.

We use different sample input data to describe the various states of the neural network in the column of the petri nets. In this case it appears that determining is set S. From state P1 in which we have zero values, we move in the states P2 and P3. In these conditions, already one of the input values is 1. This

determines the placement of a marker in these fields. From states P2 and P3 we already go to states P4 and P5. In fourth position, both input values are 1. We put two markers in the box. In the next state output value is 1, so we put a marker in the box. In the following two conditions P6 and P7 we have two markers in the fields. In them one input value and the output are 1. In the last state both inputs and outputs are 1. So we have three markers in this field.

We remind that this is research of the first neural network. In the same way can be examined case of the second neural network with two neurons in the hidden layer. Since it is identical we shall not dwell on it here in detail.

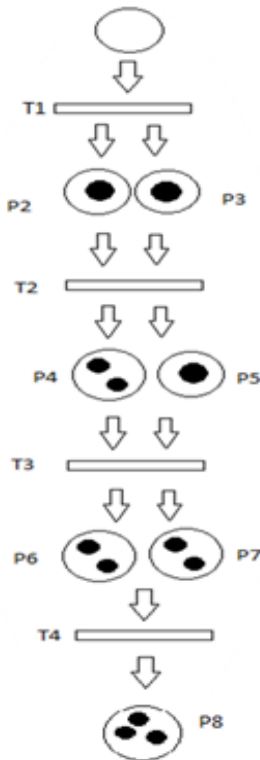


Fig. 10. Petri net for the first architecture of the neural network

Thus created graph of Petri nets can be very useful. Here the ability of Petri nets of analysis may be used. The research of fields can give many details on the neural network. Could be analyzed the truth in the result of the neural network. Neural network gives correct result where in the fields is missing marker or has two markers. In the study neural network we have the correct outcome in Example 1, 4, 6, 7 (here we look at sample values in set S). After building Petri nets based on these examples, we building a graph of the Petri net. And here we see that the fields corresponding to the correct result have two markers. It is possible in general there is no marker. So just looking the markers of graph of the petri nets can be seen how much truth there is neural network.

We see the ability the graph of the petri nets to be used to predict the accuracy of neural network. So using it can be selected at appropriate architecture of the neural network. It can be seen that Petri nets are very useful tool not only for the representation of neural networks, but also for their study.

III. CONCLUSION

The petri nets can be useful in determining the possible activations in the neural network and achievable conditions. The graph of Petri nets can follow all possible input examples of neural network. It can be seen where the neural network A has a correct result and where - not. Thus, by displaying the authenticity of the result in the neural network could be found ways to improving it. There is the possibility of conducting research on different architectures of neural networks. The petri nets could help to find a suitable architecture of the neural network. The results can be very useful in training of the neural networks. By imaging the neural network through graph of Petri nets could be found on the appropriate input examples with which to be trained network. So can be significantly reduced training time. Just should be selected input examples with two markers in the graph (or without markers) of the Petri net. Training neural networks can be much facilitated. The results can be applied in lectures and education on neural networks. The study of specific architecture of the neural network can be examined with Petri nets. Here it can be determined which is the appropriate neural network for the specific subject area and the specific problem. Thus can predict which architecture of the neural network will be most suitable (how many layers, how many neurons).

Research in this area can be extended. Until now research has not included the algorithm for training the neural network. Remains to be seen how it can affect in Petri nets. What will show such inclusion, how it will be implemented and what is the results of it, is the subject of a future researches.

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Content-Based Image Retrieval Using Texture Color Shape and Region

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Abstract—Interests to accurately retrieve required images from databases of digital images are growing day by day. Images are represented by certain features to facilitate accurate retrieval of the required images. These features include Texture, Color, Shape and Region. It is a hot research area and researchers have developed many techniques to use these feature for accurate retrieval of required images from the databases. In this paper we present a literature survey of the Content Based Image Retrieval (CBIR) techniques based on Texture, Color, Shape and Region. We also review some of the state of the art tools developed for CBIR.

Keywords—CBIR; Color Space; Relevance Feedback; Texture Features; Shape; Color

I. INTRODUCTION

Content based image retrieval (CBIR) has been an active research area since 1970. It applications has increased many fold with availability of low price disk storages and high speeds processors. Image databases containing millions of images are now cost effective to create and maintain. Image databases have significant uses in many fields including medicines, biometric security and satellite image processing. Accurate image retrieval is a key requirement for these domains. Researchers have developed several techniques for processing of images databases [1]. These include techniques for; sorting, searching, browsing and retrieval of images. Traditional image retrieval approach interprets image by text and then use textual information to retrieve images from text-based database management system. This method has several drawbacks; it uses keywords associated with images to retrieve visual information. It is very tedious and time consuming. It is hard to describe the contents of different types of images with textual representation. Keywords due to their subjective natures fail to bridge the semantic gap between the retrieval system and the user demands; consequently the accuracy of the retrieval system is questioned. The keyword for describing images becomes inadequate in large databases. It is not scalable.

Content Based Image Retrieval (CBIR) is a powerful tool. It uses the visual cues to search images databases and retrieve the required images. It uses several approaches and techniques for this purpose. The visual contents of images, such as color [2], texture [3]–[5], shape [6] and region [7], are extensively explored for indexing and representation of the image contents. These low level features of an image are directly related to the contents of the image. These image contents could be extracted from image and could be used for measuring the similarity amid the queried image and images in the database using different statistical methods. In content-based retrieval systems different features of an image query are exploited to search for analogous images features in the database [8]–[10].

Various techniques based on texture features have been proposed in the literature. These include both statistical approaches and spectral approaches. Mostly these techniques are not able to capture accurate information. Color is most reliable feature which is easier to implement for retrieval of image. Color is easier to implement because it is robust to background compilation. It is free of image size and its orientation. The most common approach for color features extraction of images is histogram. Color histogram illustrates the color distribution in image and it entails low computational cost. Color is also insensitive to trivial deviations in the assembly of image. The main shortcoming of color histogram is that they cannot fully consider spatial information and they are not exclusive [11].

Different images having same color distribution yield almost similar histograms. Besides, in diverse lighting conditions analogous images having same point of view generate dissimilar histograms. Despite of using the information extracted from image, most of the CBIR systems yield imprecise outcomes. Because it is challenging to relate the low-level features with the high-level user semantics. This problem is known as semantic gap [12]. To over-come the problem of semantic gap, relevance feedback methods are used in [12], [13]. Relevance feedback method provides a

mechanism for CBIR system to allow the system to learn about the features best serve the user's interests. This method enable user to assess the images retrieved by the current query and assign them values which indicates their relevance.

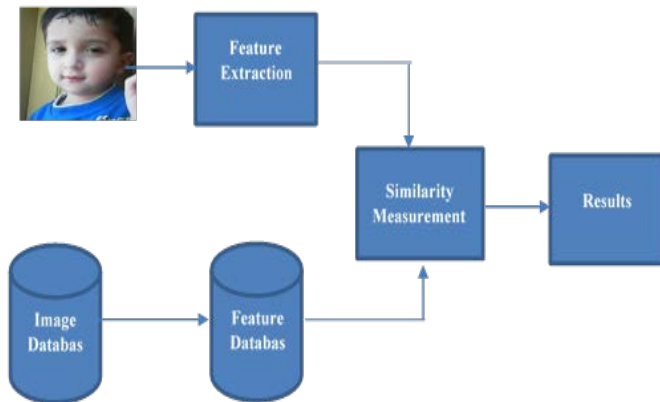


Fig. 1. CBIR system overview

II. RELATED WORK

Research on CBIR could be bifurcated into two groups on the basis of the features used to retrieve the required image. Early approaches used a single feature out of the available features namely shape, texture, color and region for retrieval of the required image. Results of single feature based retrieval systems were not satisfactory because generally image contains several visual features. The current approaches use different combination of the visual features to retrieve the required image [1], [5], [14], [15].

The shape descriptor also provides dominant information in image retrieval because shape is the only source through which humans can recognize objects. The shape feature can be retrieved by two methods boundary based shape feature extraction and region based shape extraction. The boundary based technique is based on outer boundary while the region based technique is depending on the whole region [16].

An efficient CBIR system with better performance is presented by using the wavelets decomposition of image; they have generated the composite sub-band gradient and the energy distribution pattern string from the sub images of are generated by means of wavelet decomposition to the input image [1]. For filtering out the undesired images a technique based on energy distribution pattern strings fuzzy matching is used. The resultant images are compared with query image after filtering. The system is tested on the database of 2400 images.

Texture has no formal definition but intuitively it provides measure of properties such as coarseness, regularity and smoothness. It plays a role in human visual perception and interpretation. Texture describes in three approaches, namely, structural, statistical and spectral. In structural approach the texture is formed of small texture elements called 'texels' by following placement rules. The statistical method assumes texture by means of statistical grey level features of image pixels. The spectral approach is based on filtering theory in frequency domain and power density function. The structural approach is not so prevalent because majority of the natural structures contain asymmetrical shapes [3]. In CBIR the

conventional Gabor representation and the features extracted by it has demonstrated poor performance especially in retrieval of the rotated and scaled texture images. Most of the existing retrieval systems rely on several stages of transformations for making scaling and orientation at the cost of too much computational complexity and degradation in performance.

In [5] experimentations on large data set of textured images are performed. The authors have shown that the performance and efficiency of Gabor wavelet performs better as compared to the conventional orthogonal wavelet based features. The conventional Gabor transform have some limitations. The orthogonal based sets are not formed by Gabor function that is why its representation is not solid and it consumes more computation cycles and more memory. In [17] a CBIR system is presented. It extract textural and color features of image. To extract color features color histogram is used. Gabor filter is used for the extraction of texture features. The image phenomenon of image retrieval is carried out on the basis of linear combination using texture, structural and color histogram values. Image contents are retrieved on the basis of three features and a selection technique for feature extraction is also used to select optimal features for the enhancement of the detection rate and to minimize the computational complexity of the image retrieval.

A color distribution based system is proposed in [2]. It computes the probability of the incidence of identical pixel colors for each pixel and its neighbor pixels in the image. The difference among pixels is calculated and applied to the entire image. All pixels in the image are grouped by using K-means Clustering algorithm. The technique is applied to 3 databases and better results are reported. The technique is not suffering from image displacement and rotation. Analyzing images in real-time dynamic environment is not feasible. A system is proposed in [18] that analyze images in real-time environment. It extracts shape and color to represents image contents. The system uses C-Means clustering algorithm to segment the images and extract its boundaries. Fast Fourier Transform is applied to provide an array of vector corresponding to a region. For color features HSI color model is used. Similarity Matching Algorithm using distance measures among the feature vector is applied to match the queried images.

The capability of a CBIR approaches is fully dependent on the features retrieved from the image. Frequently it is observed that there is a semantic gap between the visual features and semantic content of an image. The semantic gap could be decreased by extraction of more effective features. This is a challenging area in CBIR research. To overcome the semantic gap different machine learning techniques are used. In [19] and [20] SVM is used to extract the image features accurately and retrieve the desired image efficiently. In [21] hierarchical methodology is used to retrieve an iris image is presented. This technique is based on an innovative indexing method for an iris database. Two distinct features are exploited to extract the contents of iris image i.e. iris color is used to form the basis for the indexing of image while texture of image is used to retrieve image from the indexed iris database. The undesired images are filtered out by using the color feature; the images having no similarity with the query image color are filtered out. The proposed technique is evaluated over noisy iris images. It

produced better results. A novel image retrieval scheme (ICTEDCT-CBIR) based on curvelet transform is presented in [11], this model integrates Curvelet multi-scale ridglets with region-based vector codebook sub band clustering to extract dominant color feature and texture analysis. The advantage of Multi-scales of Curvelet transform is that it restores sparsity by curtailing the redundancy across multi-scales. The Curvelet based technique is better than the previously discussed techniques because along with Gabor filter, the wavelets are mostly used to retrieve image contents. The wavelets perform well while presenting point singularities but it ignores the geometric properties of shapes. It also ignores the use the symmetry of image edges.

Curvelet based technique is proposed in [22]. It uses Curvelet transform along with vocabulary tree for feature extraction and image retrieval. The Gabor wavelet transform is used in conjunction with Curvelet transform. For feature vector generation Histogram with vocabulary tree is used. The proposed algorithm is compared with Gabor transform and wavelet transform. The proposed algorithm showed better results because of capturing more accurate edge information.

In [6] a new approach based on three popular algorithms that are: color histogram, texture and moments invariants is proposed. The three algorithms are used to ensure capturing of the regularity of image edges. They have extracted the texture features by using Gabor filter. Shape features of an image are retrieved by using the moment invariants and the color features of the image. The proposed method is evaluated by using the typical precision and recall measure. The main drawback of surveillance images is that their quality is low. That makes the detection, recognition and retrieval process difficult. Environmental factors and use of low resolution cameras are the main reasons of low quality of surveillance images. Fundamental contributors in environmental factors are fog, rainfall and snow which can affect the quality of image. These factors affect the image details very badly and generate noise that makes the detection of objects in image and retrieval of image information more difficult. On the basis most significant features selection a CBIR model is presented in [23]. In this model Gabor filter and 3D histogram are combined together for feature extraction. Gabor filter is used to extract texture feature while 3D histogram is used for color feature extraction. They have used Genetic algorithm to get finest boundaries of the intervals. The model has reduced the retrieval time by introducing new method for feature selection. Because the traditional methods are based on the curse of dimensionality this causes the degradation in performance and increases computations. The effectiveness of CBIR depends on image descriptors that are being used to retrieve contents of image [24] like distance functions, color descriptor, texture descriptor and shape descriptor.

In [25] a new approach to extract color and texture features of image for CBIR is proposed. They identified the low level features of color and texture for CBIR by using two color histogram function and their comparison. It combined texture, color and shape descriptors for image retrieval. The paper focuses on feature extraction and representation. In [26] an image retrieval approach based on fuzzy KNN classifier is proposed. It allocates an initial semantic label to database

images. In this method the assigned labels to the images are modified steadily by relevance feedback. Several measures based on similarity were observed for three kinds of visual features. Genetic algorithm was used to assign optimum weights for each type of feature and to find out their components. For the residual 800 images of the image database, the proposed approach assigned outputs classifier as initial weights to the associated links of the network. During the image contents retrieval session, the network weights are turned in to the relevance feedback from the users.

The semantic gap between low level visual features and high level semantics is reduced by means of instantaneous feature selection and variations [27]. They used orthogonal wavelets for texture retrieval and color histogram to extract the color feature. The proposed method is applied on 1000 images database and the method produced better results. Texture, color and shape descriptor are used for image retrieval in [28]. The image is determined by utilizing color quantization algorithm along with the cluster merging technique. The approach retrieves texture by using steerable filter decomposition and achieved the shape descriptor by using Zernike moments of image moments. Zernike moment is more robust to noise.

In [29] Neural networks are exploited for CBIR. The images for classification are divided into foreground images and background images. Region based segmentation technique is used to get the region of the objects. Texture feature are extracted by using wavelet transform to retrieve shape based. They used neural network for back propagation learning.

In [30] a new descriptor for the retrieval of image is called micro-structure descriptor (MSD) is proposed. The MSD is defined on the basis of edge orientation similarity. This technique based on color, it extracts features and effectively combines them with shape, color, texture and color layout features entirely for image retrieval. The proposed method is verified on the database of 15000 images. The results are compared with Gabor features and multi-textons histograms. It produced better results. Feature selection is a common problem in CBIR. Image characterization with fewer numbers of features decreases the computational cost. Edge is also a robust feature for image characterization. A robust method for edge detection and extraction is proposed in [31]. They computed global features by means of using gray level and the shape information. This method does not require preprocessing and image segmentation. A CBIR System which first performs image segmentation by dividing image into various regions is proposed in [32]. The regions are used to retrieve image. The image searching is performed on the basis of image regions that have similar association with the regions that exists in image query. For image retrieval color and texture features are used in [32-33]. An image retrieval system based on the idea of motif co-occurrence matrix is presented (MCCM) in [34]. This technique differentiates pixels and then converts them into basic graphic. Probability of occurrence of the neighboring areas in the image is calculated to acquire the color variance concerning the adjacent pixels.

In [2] a novel CBIR system based on three methods: Initially co-occurrence matrix (CCM) is computed for color feature. The CCM matrix is used to analyze probability of

occurrence of pixels having same color and the adjacent pixels in the image. Second the difference between pixels of scan patterns (BDPSP) is computed to find out the variance among all pixels of scan patterns. The third method is color distribution for K-mean algorithm. It is based on color histogram in which each color pixel is substituted by any color that is utmost related to the existing color. The K-mean algorithm divides all the pixels into k clusters. A CBIR system based on three algorithms viz. feature extraction, image mining and the rule based is proposed in [35]. The first algorithm globally extracted the color and texture features from the image. It is considered that these features are invariable to the image transform and could be used for the detection of the objects. The second algorithm uses the image mining method that implies clustering algorithm to retrieve hidden knowledge from the image. The third algorithm uses the rules based on relevance feedback to filter the results and to improve the clusters.

A. Texture

Texture is mostly used in CBIR. Texture classification and texture based image segmentation are the challenging tasks [1], [3] [4] [5]. Texture is based on neighborhood. The texture feature provides vital information for the classification of image because it is useful in describing the contents of numerous real world images such as fruit, skin, bricks, trees, clouds in the sky and fabrics. Texture helps in describing the high level semantics for image retrieval. The main problem in texture retrieval is the scale selection; most of the literature work has neglected the importance of scale selection for the computation of texture. Statistical methods are used for the analysis of image grey level spatial distribution. The probability of the co-occurrence of gray values in distinct orientations and at different distances is calculated by these methods. Histogram is used for statistical information generation. The co-occurrence matrix is used in [2], [34] for texture identification. Texture is examined by means of texture primitives for this purpose Geometrical approach is adopted. This analysis is performed by taking the geometrical characteristics of the primitives, such as: size, shape, area, and length. In [34] [40] grids are extracted from related vectors which join the primitives' centroid.

This type of analysis is challenging for various natural textures, due to the presence of the irregularity in primitives. For example, it is to describe a wall of bricks by means of brick primitive and placement rule for this can be very simple as compared to describe the clouds in sky. Because clouds have a lot of variations in shape, size and position. Model based approaches depend on the assembly of image models. These models can be used to define and produce textures. In several systems, the image texture feature is extracted on basis of the pixels texture property or small blocks present in a small region [49]. They have calculated the mean values of the texture of all the 4*4 blocks and used it as a region feature.

Such features suffer from problems because they cannot define the texture property of the whole region. Image retrieval system proposed in [50] relies on texture features only. The texture feature similarity is extracted by using wavelets and gradient vector decomposition. The system used two characteristics of each image namely the image detail features

and the coarse features. The coarse feature is used to discard the undesired images and the detail features of image are used for the related image retrieval.

B. Color

Color points create color space. Various color spaces based on the perceptual concepts are used for color illustration. There is no universally accepted criterion to judge color space. The desire features of a color space are its completeness, compactness, uniformity, and user orientedness. Completeness shows that it must comprise of all the perceptible colors. Uniformity indicates the closeness among the different colors. It is directly connected with the psychological association among the colors. Various color description are used in [36]. The color spaces are more related to human perception and widely used in CBIR. That is comprised of RGB, LUV, HSV, YCrCb and LAB [37]–[39].

Color plays vital role in extraction of information from images. Color histograms are frequently used in CBIR systems [6], [17], [22], [40]. Histogram is useful though its global characterization is weak. Histograms are not efficient in handling noise because they are very sparse. In order to overcome limitations of the histogram other color features such as color moments and color sets representations are applied for image retrieval [6]. The color feature selection depends on the result of image segmentation. If the segmentation provides objects having heterogeneous color, then average color will not be a good option for image retrieval. Most of the CBIR systems function on the color images that are not preprocessed. The image capturing devices introduced noise in the images. If these color images are preprocessed then image retrieval accuracy of CBIR system will be enhanced. Noise from color images could be removed by filters. Several noise removing filters are proposed in the literature [41-43].

Literature survey shows three methods for color analysis. The Global approach focus on the image color information globally without applying any segmentation or preprocessing technique during feature extraction of the image. This approach is efficient but do not provide any information associated to the spatial distribution of colors. It generates color histogram as feature vector [44], [45]. Equal size regions approach decomposes the image into small portions of fixed size on the basis of the small regions separately. This approach provides more spatial information. It uses local color histogram to generate feature vectors [6], [46]. The third approach is the segmentation based approach that divides the image into different regions of different sizes. Clustering algorithm is responsible for the division that causes an additional complexity to feature extraction process [47], [48].

C. Shape and region

Shape can be defined as the feature surface configuration of an outline, object or contour. The shape feature is used to separate objects from the background and surrounding by its outline representation. It can be categorized into two broad categories. Region based and boundary based image retrieval. Regional properties of images can be utilized, and segmentation can be used for color, shape feature extraction and spatial position of the regions. In [7] a technique based on region is used for image retrieval. They used image

segmentation in small regions. Local properties of different regions are helpful in matching objects of the images. This makes CBIR effective. An approach based on region matching, that uses combination of different features like location, color, and shape is developed within the MPEG-7 framework. They have used these integrated features (shape, location and color) for the indexing of significant regions within each image. It also indexes various arrangements of regions. Similar images are clustered and Hash structure is used to capture resulted indices and Meta data. The images are retrieved on the basis of color, shape or location and their combination. Results indicate that CBIR can be made more effective and efficient by using segmentation to retrieve the images. In [51] specific regions of the objects of most interest are computed by using low level features.

The results indicate that the simple threshold technique is more accurate and produce better results than the methods which are more complex and based on statistics. There are three methods for the detection of salient objects in an image; the non-parametric measure [52], the threshold based measure in which threshold is manually selected and the modified Hubert index [53]. For the classification of objects in images, image can be divided into foreground and background regions [54]. Objects regions are extracted by using region segmentation techniques. Shape-based texture features are mined from images by using wavelets transform. The method is used to automatically classify objects of images. Neural networks are also used for the classification of objects. The selective visual attention can play a key role in human understanding of image by intuitively focusing on some salient parts. The interest of CBIR systems users is confined to selective portions of image and the rest of image is irrelevant.

A selective visual attention model is proposed in [55]. In most of the region based approaches, the shape descriptors use the pixel information within the region of the shape. Such techniques are applied for general applications. The region based techniques use moment descriptors to define shape [56], [57]. These moment descriptors are consisting of Legendre Moments, Geometric Moments, Zernike Moments and Pseudo-Zernike Moments. The region based shape descriptors are retrieved by means of spatial domain and most of them are suffering from noise and shape variations. Generic Fourier descriptor is applied to general applications. The Generic Fourier descriptor is extracted by using spectral domain. Shape finer features in both circular and radial directions are captured by using 2-D Fourier transform [58]. A region based retrieval method is presented in [59]. This method combines motif co-occurrence with spatial relationships. Local information is extracted by Local space filling curves to calculate motif co-occurrence matrix for texture description. It decomposes each image into coherent segments.

The motif co-occurrence matrix is computed for all regions. Spatial relationships between these regions are calculated. During the process of image retrieval each of the regions is allocated its own MCM and it is used as a feature vector. It is used for satellite images.

Semantic based CBIR has become very popular and attracted interest in current decade. In [60] [61] region based image retrieval system with high level semantics learning is presented. This method is more flexible because supports both query by region of interest and query by keyword. It divides image into diverse regions and for each region low level features are extracted. Decision tree based learning method (DT-ST) is used to achieve high level concepts.

TABLE I. COLOR, TEXTURE AND SHAPE FEATURES USED IN CBIR SYSTEMS

Low Level Features	Features	Advantages	Limitations	CBIR System using the feature
COLOR	Color Moments	They are sensitive to spatial information	Compactness may cause low power of discrimination	Assert
	Color Space	It represents each pixel in 3D space used for image display	Unable to separate the brightness information sensitive to human eye.	Image Rover
	Color Coherence Vector	Due to additive spatial information it is more efficient	Very much complex because of its high dimensionality	Nil
	Color Histogram	Extract both local and global color features.	It reduces the performance if the numbers of bins are increased.	CBIRD
Texture	Gabor Filter	Used for detection edge, line and different orientation	Only effective for manmade objects	NETRA
	Wavelet	Efficient for image retrieval with salient point feature	Its general selection is dangerous for investigation	MIR
	Tamura	It is very effective when used in combination with color histogram	Very complex	QBIC, Photobook
Shape	Fourier Descriptor	FFT can be used for efficient results	DC component is dependent on position of the image	Draw Search
	Bounding Box			NETRA

A set of images whose concept matches with the query image is retrieved. This approach has significantly enhanced the image retrieval process as compared to the conventional CBIR systems. The proposed DT-ST decision tree induction method from image semantics learning is better because it uses the semantic template to discretize continuous valued regions feature. This algorithm has improved classification performance of the tree and it has outclassed the well-organized ID3 and C4.5 algorithms

III. SIMILARITY MEASURES

An image may contain visual information or semantic information. The visual information can be represented in form of shape, color, texture and spatial relations. The extracted visual features are considered as feature vector that are kept in feature database. The retrieval process is performed by different similarity measuring techniques that match the resemblance between the query image and the stored image in the database. The result of the query may not be a single image. It is a series of images categorized by the similarity of retrieved image with the query image. Retrieval performance of the CBIR systems is affected by similarity measures. Similarity measures are used to find the correspondence of the query image with the image stored in the database. This section contains some similarity measures that are used in the literature. We denote I=Query image J=Image in database $f_i(I)$ = represent the number of pixels in, i^{th} bin of query image.

A. Minkowski-Form Distance

The Minkowski-form distance L_a is apt for the computation of distance between two images when each of the image feature vectors having equal importance and are independent from each other. It is defined as:

$$D(I, J) = \sqrt[\alpha]{\sum_{i=1}^n |f_i(I) - f_i(J)|^\alpha} \quad (1)$$

If $\alpha=1, 2, \dots, n$ $D(I, J)$ is L_1, L_2, \dots, L_n then this is called Euclidean distance. The minkowski-form distance for color and shape feature is used in Netra [62].

B. Histogram Intersection

Histogram intersection is the special case of Euclidian distance. Many CBIR systems used it for computation of the similarity among color images. Histogram intersection of image I and image J can be measured as:

$$S(I, J) = \frac{\sum_{i=1}^n \min(f_i(I), f_i(J))}{\sum_{i=1}^n f_i(I)} \quad (2)$$

It has been revealed that histogram intersection is reasonably tactless to variations in image resolution, histogram size, occlusion, depth, and viewing point.

C. Quadratic Form Distance

The problem in Minkowski distance is that it treats all the bins independently. It ignores the similarity between certain pairs of bins having same features. This problem is addressed by Quadratic Form Distance.

$$D(I, J) = \sqrt{((F(I) - F(J))^T A (F(I) - F(J)))} \quad (3)$$

Where $A = [a_{ij}]$ denote the similarity among i and j bins. Vectors F_i and F_j enlist each of the items in $f_i(I)$ and $f_i(J)$. Many CBIR systems use quadratic form distance [63]. It can

produce better retrieval results as compared to histogram intersection and Euclidean distance methods.

D. Mahalanobis Distance

Another metric suitable for all dimensions of image feature vector is defined as.

$$D(I, J) = \sqrt{(F(I) - F(J))^T C^{-1} (F(I) - F(J))} \quad (4)$$

The C represents co variance matrix of the feature vectors. In case the feature dimensions are independent it will become so simple. In such type of situation we need only variance of each feature component.

IV. RELEVANCE FEEDBACK IN CBIR

Currently in CBIR systems an effective tool relevance feedback is used to bridge the semantic gap and improve the retrieval rate. It is the supervised learning approach. The focus of this approach is on interaction between user and the search engine. The user is responsible for the labeling of semantically positive or negative feedbacks.

There are three categories of relevance feedback techniques namely; query point movement method [64], Reweighting method [65] and Expectation Maximization method [66]. The query point movement technique focus on the improvement of the approximation of the ideal query point. It diverts it to the positive examples and far from the negative examples. In reweighing method the weight of each feature element is modified. Most of the work done in this area has used SVM (Support Vector Machine) for the pattern classification. it is a small and simple learning method [67]. SVM is considered as a strong candidate for classification because of several reasons: its generalization ability, fast learning, flexibility and assessment of relevance feedback.

Active SVM learning method used in [65] split the image space. In this method the positive samples are considered furthest from the boundary of classifier on the positive side. The most informative samples are adjacent to the boundary for user labeling. Relevance feedback is an approach that reflects human interaction to rectify high level concepts defined by low level features. It is basically used in text based retrieval [8]. In the application of using relevance feedback first query image is selected then user marks the relevant images from retrieved result and modifies the query.

In [68] the authors automatically refined high level concepts on the basis of relevance feedback from the user. SVM is utilized to create a classifier with minimized Vapnik Chervonenkis (VC) dimension [69] based on structural risk minimization principle. The work on CBIR systems are widely discovered in the past decade [70]. The semantic gap existing between the high level and low level concepts is handled with Relevance feedback technique [71]. SVM has been considered as the most powerful tool to enhance the retrieval rate of image retrieval systems. SVM is considered is also considered as a leading technique for classification in relevance feedback systems [72-73]. The learning problem in SVM-based relevance feedback approaches is considered as a strict binary classification task. In real world relevance feedback applications this assumption is incorrect.

TABLE II. CBIR SYSTEMS AND QUALITATIVE COMPARISON

SYSTEM	Texture				Color						Shape					Key Words	
	Wavelet, Gabor, Fourier	Atomic texture features	Random Fields	Edge statistics			Color moments	Global Histogram	Color Coherence Vector	Correlation Histogram	Others	Region Histogram	Bounding box	Edge Direction Histogram	Template Matching		Elementary Descriptor
QBIC		√					√	√							√	√	
Photobook				√						√							√
NETRA	√										√						√
Jacob		√		√			√										
QuickBook 2	√	√	√			√	√	√	√			√	√				
ADL	√																√
Blobworld		√								√					√		
CANDID		√												√			
Cbird				√			√								√		
Visual Seek												√					
WEBSEEK							√										
Pic hunter																	
Assert						√											
Image Miner				√						√							

V. CBIR TOOLS COMPARISON

Most CBIR systems are the artifacts of research, therefore focuses on a single aspect of CBIR. Two types of CBIR systems exist, one is research based CBIR systems and the other is commercial based CBIR systems. The commercial versions display more standard searching capabilities and are typically less advanced. Various CBIR systems offer user interface with flexibility that enables more powerful query design. That is valuable in the demo based systems. For example if the database only contains images of airplane and user want to search cartoon then the system will only be able to always retrieve images of airplanes. For such type of small databases powerful sketching user interface is useless. The majority of CBIR systems are based on color and texture features. Few of the retrieval systems are practice shape features and very rare systems practice the layout feature. The retrieval on color feature is easy to implement.

The systems based on color feature usually retrieve images having similar color. The problem in texture based retrieval is that they produce in-accurate retrieval results. It does not always produce images that have noticeably the same texture, the texture may be effective only when the database comprises of many images with a prevailing texture. Searching for shape feature yields frequently best outcomes. Superficially for matching purpose shape features are not the best ones. Indexing data structures are not applicable for small databases. Linear search is the better alternative of indexing structure for such kind of databases. Simple matching of hundreds of images in efficient manner is not a problem for modern computers In terms of efficiency, effectiveness, and flexibility,

it is very hard to decide that how effective are the CBIR systems in terms of precision, accuracy and recall. Accuracy is defined as the ratio of appropriate images to the total retrieved images from database and recall is defined as the percentage of appropriate images to all relevant images. Numerous articles about CBIR systems contain figures showing the precision and recall. Most of these results are decent, but it is difficult to validate them. It is extensively accepted that most current CBIR systems are using the low level image features (color, texture, shape) for image retrieval. The high level semantics will be the main driver of the future CBIR systems. One way to attain this is to make the systems capable of identifying objects and scenes. This is a challenging task at implementation level but it should be practicable for applications used in specific domains [74].

VI. CONCLUSION

The content based image retrieval techniques are still under research. Various algorithms are proposed to improve the image retrieval capability of CBIR systems by using content based approaches. Majority of these approaches are based on a single algorithm and ignore the existence of others. Approaches which are relying on a single specific algorithm like color, texture or shape, such type of approaches can work successfully on specific images but when varied types of images are taken into account their performance is degraded.

In this paper we have presented different techniques used for image retrieval. Some of these approaches were single feature based and some of them were the combinations of these features (Texture, Color, Shape and region). Most of the studied techniques lack accuracy and are unable to overcome

the semantic gap between the user and the CBIR system. This area of CBIR systems is a hot research area with huge potential for research and development of better image retrieval techniques.

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Sdaa: Towards Service Discovery Anywhere Anytime Mobile Based Application

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Abstract—Providing on-demand service based on customers' current location is an urgent need for many societies and individuals. Specially, for woman, elderly people, single mother, sick people, etc. Considering the need of providing localized services, this paper proposes a mobile application framework that allows an individual to receive services from his neighborhood peers anywhere anytime. The application allows an individual to find and select reliable service providers near his location. The application will provide an opportunity to the interested individuals to use their free time for providing services to the community and earn some extra money. This application will benefit many stakeholders like elderly people, women at home, a person while traveling in an unknown place, etc. A prototype application is developed and empirical evaluation is considered to find the qualitative measures of the users' acceptability and satisfaction of the application. It is observed that users' satisfaction is high.

Keywords—mobile application; service discovery; mobile services; software engineering

I. INTRODUCTION

With the fast advancement of mobile computing technology current society is growing so fast having the ability to keep in touch with families and friends, search information in the internet, shop online without going physically to the shopping mall, learn using e-learning system, receive health services using e-health system, and search for daily needed searches, in variety ways using mobile devices and applications. Advances in the mobile technology has made our society more dependable on mobile and smart devices [1] and we cannot function without these devices. Mobile devices and applications play central rule to bring together the service providers and service users anywhere anytime. Over the past years, mobile application users have increased rapidly to look for various services from a mobile device.

Diverse necessities and demands of various services by citizens in nearly all aspects of daily life have fueled the use of Internet and mobile applications. Through the applications a consumer can shop, book appointment, communicate with peers, reserve tickets, etc. sitting at home, workplace, or while travelling with a computer or mobile devices. However, facility to look for on-demand service providers for our daily needed services (e.g., personal care, home service, food prepare and delivery, drop and pickup services, etc.) around the vicinity to our present location is still unavailable through mobile and

internet applications. Still, social contact is needed to find reliable service providers. In our daily life we need diverse services. Service may include home cleaning, buying groceries, delivery of foods, giving a ride, fixing a car on demand, etc.



Fig. 1. The location based service providing scenario

For example, Mr. John is driving in a new city. Suddenly the car has a problem while driving. If there exists any application that provides a list of car service servicers nearby his current location then he can easily get help from the service providers even he is new in the city. This is called location based service. The scenario is presented in Fig. 1.

Location Based Service(LBS) is gaining popularity from the services of mobile devices that is dependent on the user's geographical location for providing needed services to the clients. The location based service (LBS) uses the present physical location information of users. This physical location is used as a "filter" for providing services to the users and delivering important information about a particular area related to the users' position. There are mainly two types of LBS [2, 3]. Type 1 is a pushed based services. In pushed base service an application service provider triggers services when a user reaches in a particular location. Example of these services include push business advertisements when a user enters in a market or show weather information when enters into a city, etc. In type 2, pull based services, services are triggered when a user requests an specific service.

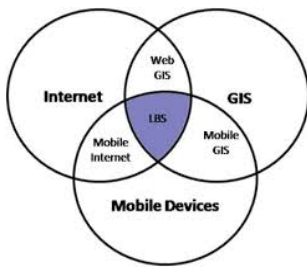


Fig. 2. Local based service architecture [6]

LBS provides services based on present location and time as well as based on the users' demands at a specific location [18]. Also the LBS system is aware of the needs of users and is able to map it to the location of the service requester. Since the mobile devices are becoming popular day by day, Mobile Location Based Service (MLBS) on the devices are expected to be a better profitable chance to the mobile service providers [19].

Exploiting mobile applications in portable devices, like smart phone, PDAs, tablet, etc. and LBS, daily needed service discovery around users vicinity can be provided to the service requesters even though the requesters move. Moreover, advances in information and communication technology have enabled service providers and requesters in the community to connect each other virtually wherever the users are located. Mobile apps help to create link between service providers and service requesters in the community.

The main goal of this paper is to propose a mobile application framework that identifies services and service providers using location based services. The need of service requesters is the central in developing the application that provides access to services when and where they need them. In the following the main goals of the proposed application are presented:

- Identify service providers and skillful persons in the neighborhood.
- Share skills among the people in the community instead of hiring people from company.
- Develop economy condition of jobless person however have skills

The rest of the paper is organized as follows. Section II discusses related works. Section III discusses the proposed application model. Section IV shows evaluation results and paper is concluded in Section V.

II. LITERATURE REVIEW

The conventional business model is changing gradually with the advancement of technology. Technological advancement facilitating the business model to reach to more markets and becoming more convenient and flexible. An example of this is the car rental company "Car2Go" [9]. The company provides its members a mobile application. Members use their Smartphone, tablet or computer to find nearby access to individual mobility through a large quantity of rental cars that are distributed across cities.

Peerby [10] is a web-based system that matches people that are in temporary need of a specific thing, with those that are in possession of the thing and are willing to lend it. Through Peerby people share (for free) their possessed items online. Members that are in need of a specific item, such as a ladder, post their item request and location through the system. The member who possesses the item and is willing to lend or share the item, and is located in a close enough vicinity of the person in need, can contact through the system. Both members living in the same neighborhood can now agree on the manner of transfer of the item.

Shareyourmeal [11] is an online application that provides "homemade foods" with the opportunity to sell their home made meals to the interested neighbors. Members of Shareyourmeal can find out that their neighbor, a talented home cook, is interested to sell homemade food that the member is interested to take. This helps community people to share cooking takeaway meals each other. The company has developed an online platform through which home cooks, willing to sell portions of meals, and neighbors in need of healthy takeaway food, can get in touch. It not only share foods, but by having neighbors visit each other for picking up takeaway food, it also helps to create trust between strangers in a neighborhood.

Sorted[12] is an online platform through which individual can sell his services to the interested parties. A sorter (service provider) can indicate the tasks he/she can conduct, hourly rate and the distance he/she is willing to travel for conducting the service [13]. The platform matches individuals that are able to provide a certain service with parties that are in need of such a service. The platform was first demand driven. Individuals could post a service request on the online platform, and wait for sorters with the relevant skills to reply.

Fixura [14] is an online peer-to-peer platform for lending and borrowing money. It provides platform to lenders and borrowers the way the clients want to lend out or borrow money. The system also matches lenders and borrowers with each other. The system provides facility to both lenders and borrowers to set the specific criteria against they want to respectively lend or borrow money. In order to lend money a borrower indicates the amount of money he/she wants to lend, the duration of the loan and the interest rate. The borrower also provide some contact information. To lend money lenders search for loan requests and assemble their investment portfolio.

Airbnb[15] is an online platform through which peers can list, discover and book unique spaces (accommodation), all made available by other peers [11].

TaskRabbit [16] is an online platform through which both peers and companies can post task they would like to "outsource" to peers that are willing and able to do these. The person that posts the task can set the price he/she is willing to pay and choose a suitable TaskRabbit from the people who bid on the task. It mainly connects neighbors in order to get all sorts of tasks and errant done, was born. The platform works similarly like Sorted, with the essential difference that this one is demand-driven, whereas Sorted is supply-driven.

This demand-driven orientation is depicted by the illustration above, which shows that the person who posts a task is in charge (e.g. of price setting, selecting a TaskRabbit and describing the service to be delivered).

Most of these companies function through an online platform or marketplace that connects consumers that own certain assets and skills with consumers in temporary need of those. These companies can facilitate peer-to-peer markets for potentially all product or service owned by consumers. This business model might become particularly disruptive to conventional rental solutions for mobility, accommodation, catering and other sorts of services, because it is able to serve the same needs at a significantly lower price. Moreover, it empowers consumers to capitalize on their property and skills, providing them with an opportunity for micro-entrepreneurship and lowering total cost of ownership.

III. SYSTEM MODEL

This section discusses the architecture of the mobile application "SDAA". The framework is shown in Fig. 3. In order to request a service or to provide a service users have to register into the system. Once a user registers he/she can post a request, see the requested services, and bid a service request. Both requester and provider can indicate the specific criteria against which they want to respectively consider or provide service. Through the system, a provider can indicate the amount of money he/she wants for the service, the duration of the service delivery to the provider. A requester can also see all the service providers' bids and select the desired service provider by analyzing the providers' portfolio. The system allows an user to search and receive services at anywhere and anytime from the user's current location. The system also allow individuals to receive services with disabilities, woman at home, individual at work, family in travel, friends in amusement parks, etc. The important services may include:

- **Personal Assistant (PA):** Delivers assistance to personal care, household jobs, and patient care with permission of a doctor. PAs are selected by individual customers. The individual will select PAs through checking service provider profiles, reputation, experience and remuneration issues.
- **Home Delivered Foods:** Provided to individuals who cannot go outside to buy food or prepare food at home. With this service an individual can ask for services to deliver food on this request.
- **Home Delivered Shopping:** An individual can ask for a service to shopping on behalf of him. The individual may be at home or at work does not have time to go for shopping or is sick.
- **Drop-Pickup:** An individual may get service for drop and pickup to office, school, market, airport, etc. if the person does not have car.

Service providing mechanism using the application is described elaborately in the following.

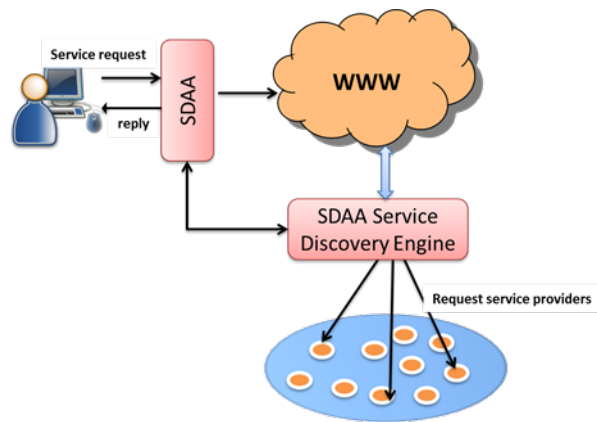


Fig. 3. SDAA Architecture

A. How are services provided?

An individual may receive service through the following steps:

1) Register to the system with the details of personal information. Mobile no is the key information to identify the current location of the individual and to find nearby service providers.

2) The client must post his service request by selecting the category of service request.

3) The client must mention the following constraints to receive better service.

- Time duration to deliver service from the time of request.
- Pickup from service provider or delivery to the individual
- Payment method. (Cash, Bank transfer, etc.)
- Payment amount.

4) A service provider may be individual or a company to provide services to a service requester.

5) A service provider must register to the system to provide services. Service providers must provide personal information to attract the service requesters and to verify the identity of the person.

6) After becoming a member the service provider can see the list of services with the following criteria.

- Distance from present location
- Types of service
- Service amount

B. How a service request is submitted and select a suitable service provider?

B.1 Online service Request

- 1) A service requester first login into the system.
- 2) Select type of service needed.



Fig. 4. Car service request scenario

- 3) Define the constraint of the service request.
- 4) Find service providers for the requested service over the map.
- 5) Select a service provider.
- 6) Make connection with the service provider for service execution.
- 7) Give feedback of the service.

B.2 Offline Service Request

- 1) A service requester first login into the system.
- 2) Select type of service needed.
- 3) Define the constraint of the service request.
- 4) Search for service providers for the requested service over the map.

5) If no service provider is available then the request is posted in the service request pool. In this case the service requester must put the details of the service and the constraint.

6) The request will be removed from the service request pool after the defined time.

B.3 Select a Service Provider.

- 1) Make connection with the service provider for service execution.
- 2) Give feedback of the service.

C. An Example Scenario

Consider a scenario of car fixing problem. Mr. John is driving in a new city. Suddenly the car has a problem while driving. Mr. John opens the application and post a request for car fixing. The system automatically detects his present location and broadcast his request to the car service providers near his current location. The application scenario is presented in Fig. 4. The system allows to search for service providers with the distance and time range. Car service providers within the range of 3 KM. Fig. 5 shows the service providers names within the range of 3 KM and Fig. 4 shows the names of the service providers with the time range. When John selects any service provider he can find the details of the service provider. After reviewing the profile of a service providers John can make a direct contact with the provider for solving the problem.



Fig. 5. Car service providers within the range of 3 KM Car service providers within the range of time limit

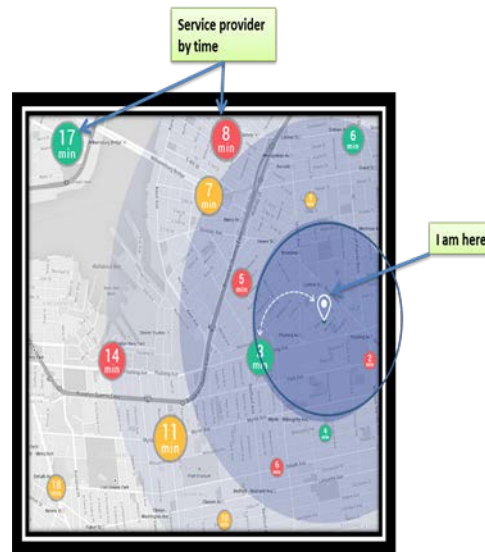


Fig. 6. Car service providers within the range of time limit

IV. IMPLEMENTATION AND RESULT

This section discusses the prototype implementation of the proposed mobile application and User acceptability evaluation of the system. For user acceptability evaluation user acceptability is measured using the Ubiquitous Computing Acceptance Model (UC-AM) [17, 4]. The obtained results show the acceptability of the proposed application. It is observed that the users' satisfaction is high. Users' experience is the most important target for any application. Therefore, understanding the requirements and characteristics of the end users is necessary. Table I summarizes questionnaires with regards to users' acceptance. The table also shows result returned by using the application. The evaluation is completed in order to better understand users' perception for qualitative analysis of the results.

It is observed that users have subsequent perception of acceptability, consistency and correctness. In order to compare with the different measurement variables, Table II is produced to show the mean response time of the application's end users. It is found that the request with more precise constraint have greater perceived acceptability, consistency and control than request with less constraints.

TABLE I. QUESTIONNAIRES WITH REGARDS TO USERS' ACCEPTANCE

No.	Measurement Variable	Questionnaire
1	Perceived consistency	I find the application provides most often returns relevant providers of the service.
2	Perceived acceptability	I observe that the application returns relevant service providers considering the contrarian that I put in looking for a service
3	Perceived correctness	I find the application returns correct or relevant service providers for my requests
4	Perceived risk	I found that in an emergency case, the application is able to provide results without much irrelevant service providers

TABLE II. MATRIX FOR USER ACCEPTANCE EVALUATION QUESTIONNAIRES WITH REGARDS TO USERS' ACCEPTANCE

		Perceived consistency	Perceived acceptability	Perceived correctness	Perceived risk
1	Perceived consistency	2.94	2.22	1.42	4.83
2	Perceived acceptability	4.2	4.25	4.27	2.85
3	Perceived correctness	4.3	4.37	4.10	3.1
4	Perceived risk				

TABLE III. QUESTIONNAIRES WITH REGARDS TO USERS' FEEDBACK

No	Question	Average Score (5:Excellent, 4:Good, 3:Neutral, 2:Poor, 1:Bad)
1	How do you evaluate the overall experience of the application?	4.2
2	How do you feel the usefulness and usability of the application?	4.6
3	How do you feel the importance of this application?	4.9
4	How do like the matchmaking between service requester and service provider?	3.8
5	How do you feel to recommend your friends to use this application?	4.4

Table 2 shows the scores generated by the end users of the application. The lower score of mean of control and perceived acceptability shows users' poor acceptance of the application. The moderate score of perceived consistency value reflects that application is potential to use even the end users have given small score in other variables. The reason is that users want some extra result with the context that can provide some benefit to them. Users' satisfaction level of the application is measured using PACMAD[5] usability model. The PACMAD usability model uses ISO standard attributes and the model of Nielsen. PACMAD also includes the cognitive load attribute that is particularly important to mobile applications. The usability model mainly considers three factors (Task, User, and Context of use) in designing mobile applications to find usability of mobile applications. These factors have impact on the final design of the user interfaces of the application. The PACMAD model also considers seven important attributes that are considered to define metrics to evaluate application's usability. However, for the proposed application only two factors: effectiveness and satisfaction to evaluate the usability of the application are considered. Table III shows the questions and the corresponding scores of each question to evaluate satisfaction and effectiveness of the application. There

were 22 users who use the prototype application. From the result it shows that users are mostly satisfied with application experience, usefulness and usability, feel that this kind of application is very important for the community service.

V. CONCLUSION AND FUTURE WORK

Location-based service providing applications demand is now increasing with the advent of GPS mobile devices. Mobile applications play central rule to bring together the service providers and service users anywhere anytime. The need for looking and obtaining daily needed services while staying at home, workplace, and anywhere anytime is highly demanding. Considering this need this paper presented a mobile application framework that allows an individual to look for services providers around his vicinity. The application provides an individual to find reliable service providers around his vicinity and to know the service providers for various needed. This application will also provide an opportunity to individuals to use their free time for providing services to his community and earn some extra money. It is proved that this application will benefit many stakeholders like elderly people, women at home, a person while traveling in an unknown place, etc. At present the prototype has been developed for the application and users' acceptability is evaluated. The future plan is to develop the application and make available for the users.

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Expectation-Maximization Algorithms for Obtaining Estimations of Generalized Failure Intensity Parameters

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Abstract—This paper presents several iterative methods based on Stochastic Expectation-Maximization (EM) methodology in order to estimate parametric reliability models for randomly lifetime data. The methodology is related to Maximum Likelihood Estimates (MLE) in the case of missing data. A bathtub form of failure intensity formulation of a repairable system reliability is presented where the estimation of its parameters is considered through EM algorithm. Field of failures data from industrial site are used to fit the model. Finally, the interval estimation basing on large-sample in literature is discussed and the examination of the actual coverage probabilities of these confidence intervals is presented using Monte Carlo simulation method.

Keywords—Repairable systems reliability; bathtub failure intensity; EM algorithm; estimation; likelihood; Monte Carlo simulation

I. INTRODUCTION

There is an ongoing effort in the industrial fields to reach more reliability and efficiency of their systems. The major risks in certainty are mainly safety, availability, costs and especially those of maintenance and lifetime. Near the industrial companies, we can go over these risks by the competitiveness and the safety which became a temptation responsible for the management of maintenance to improve the reliability objectives. The majority of the approaches of the maintenance are based on reliability such as Peña and al., 2007, Y. Dijoux (2009), L. Doyen (2012). However, the reliability of the industrial systems depend closely on the efficiency of these maintenance actions and the effective management of the maintenance policy which requires a realistic modeling of their effects. on the other hand, when a maintenance program is chosen, its efficiency and its impact on the system operation are unknown. Then appears the idea in this paper to model the system lifespan and to quantify its degradation state or its failure to realize the impact of a maintenance action on system behavior.

The most important characteristics is the evaluation of the system failure's intensity, and the discovery of its degradation at the appropriate time. And in order to optimize the maintenance programs by reducing the costs we use the Maintenance Optimization by reliability (MOR) as presented in Dewan and Dijoux (2015).

First of all stochastic models of failures process and repairs of various systems are builded. Secondly, the statistical

methods are implemented to exploit the failures and maintenances data raised by experts to evaluate the performance of these systems. In this context, different methods as MLE, moment estimation, and EM algorithm are presented in Doyen (2012). Sethuraman and Hollander (2009) developed a non-parametric Bayes estimator for a general imperfect repair model including Brown-Proschan model. Doyen (2011) generalized this approach and considered the maximum likelihood estimation. The performance of the Brown-Proschan model when repair effects are unknown as resulting in the work of Krit and Rebai (2012) and Krit (2014). Babykina and Couallier (2012) used EM algorithm to estimate the parameters of a generalization of this model which allows first-order dependency between two consecutive repair effects, they assumed that only some repair effects were unknown. Lim and Lie (2000) proposed another method based on Bayesian analysis: they assumed a prior beta distribution for parameter p . Langseth and Lindqvist (2003) generalized the Brown-Proschan model for imperfect preventive maintenance, and they proposed to estimate the parameters of the model with the likelihood function. Franco and al. (2011) study the classification of the aging properties of generalized mixtures of two or three weibull distributions in terms of the mixing weights, scale parameters and a common shape parameter, which extends the cases of exponential distributions.

Formerly, some work has been done on the estimation of the three-parameter log-normal distribution based on complete and censored samples. Basak and al. (2009) developed inferential methods based on progressively censored samples from a three-parameter log-normal distribution. In particular, they use the EM algorithm as well as some other numerical methods to determine MLE of parameters. The asymptotic variances and covariances of the MLE from the EM algorithm are computed by using the missing information principle.

The purpose of this paper is to formulate a general and realistic model, in order to identify the behavior evolution of repairable system during all its lifetime.

The paper is organized as follows; Section 2 presents the characteristics of the failures process. Section 3 analysis the various models with mixing corrective and preventive maintenance. An application to real data on the quality of the model parameters estimators is completed in section 4. Finally, conclusions are presented in section 5.

II. CHARACTERISTICS OF THE FAILURE PROCESS

In this section, the failure intensity in bathtub form is presented to formulate pace of such intensity on the three phases of the system life, Krit and Rebai (2013). Two forms are distinguished one from the other by a small change over the service life time. In the first form, as it indicates the hereafter form, the failures process is modeled by superposition of three Poisson processes; the first and the third non-homogeneous and the second is homogeneous, of which the intensity is selected in following way:

It declines up one instant noted by γ_0 , according to the function of the form $\frac{1}{\eta_0} + \frac{\beta_1}{\eta_1^{\beta_1}} (t^{\beta_1-1} - \gamma_0^{\beta_1-1})$, translating the system improvement state in time course. After that, it's constant on a level $\frac{1}{\eta_0}$ (there will not be an advance of system degradation in this phase) up to an instant γ_1 which is beyond the intensity increases in accordance with the form function $\frac{1}{\eta_0} + \frac{\beta_2}{\eta_2} \left(\frac{t-\gamma_1}{\eta_2}\right)^{\beta_2-1}$, realizing a degradation case. This idea is originally proposed by Mudholkar-Srivastava (1993) in the context of non-reparable system. It is proved that this degradation modeling comprises two terms ; one found in Weibull process wich is proceeded by admitting the assumption of perfect corrective maintenance stated in Bertholon and *al.* (2004) like an alternative against Weibull law. The waiting duration of next failure can be written by the form $\mathcal{X} = \min(\mathcal{Y}, \mathcal{Z}, \mathcal{W})$, where:

- \mathcal{Y} a random variable, independent of \mathcal{Z} and \mathcal{W} , of Weibull law having as form the first expression, with a shift parameter equal to zero.
- \mathcal{Z} a random variable, independent of \mathcal{Y} and \mathcal{W} , of exponential law with parameter η_0 , which corresponds to constant failure intensity equalizes to $\frac{1}{\eta_0}$.
- \mathcal{W} a random variable of Weibull law having a shift parameter equal to γ_1 .

Our proposal, with the help of system behavior modeling, characterizes the failures process by intensity which is formulated as follows:

$$\lambda(t) = \begin{cases} \frac{1}{\eta_0} + \frac{\beta_1}{\eta_1^{\beta_1}} (t^{\beta_1-1} - \gamma_0^{\beta_1-1}) & \text{if } 0 < t < \gamma_0 \\ \frac{1}{\eta_0} & \text{if } \gamma_0 \leq t \leq \gamma_1 \\ \frac{1}{\eta_0} + \frac{\beta_2}{\eta_2} \left(\frac{t-\gamma_1}{\eta_2}\right)^{\beta_2-1} & \text{if } t > \gamma_1 \end{cases} \quad (1)$$

Knowing this intensity, the implicitly of system reliability can be removed, by using the following relation:

$$\mathcal{R}(t) = \exp \left\{ - \int_0^t \lambda(u) du \right\} \quad (2)$$

The failure number until the instant t, noted \mathcal{N}_t , follows formally a Poisson law with parameter $\Lambda(t) = \int_0^t \lambda(u) du$. For the present model,

$$\Lambda(t) = \left[\left(\frac{t}{\eta_1}\right)^{\beta_1} - \left(\frac{\gamma_0}{\eta_1}\right)^{\beta_1-1} t \right] \cdot \mathbf{1}_{[0, \gamma_0[}(t) + \frac{1}{\eta_0} t + \left(\frac{t-\gamma_1}{\eta_2}\right)^{\beta_2} \cdot \mathbf{1}_{[\gamma_1, +\infty[}(t) \quad (3)$$

Let's announce first that all that times inter-failures are not independent. In this case, the function $\mathcal{F}_{\mathcal{T}_{n+1}/\mathcal{T}_1=t_1, \dots, \mathcal{T}_n=t_n}$ have a conditional law of the next failure instant \mathcal{T}_{n+1} such as:

$$\mathcal{F}_{\mathcal{T}_{n+1}/\mathcal{T}_1=t_1, \dots, \mathcal{T}_n=t_n}(t) = 1 - \exp\{-[\Lambda(t) - \Lambda(t_n)]\} \quad (4)$$

III. ALGORITHME EM

A. Basic Theory of the EM Algorithm

The EM Algorithm, proposed by Dempster and *al.* (1977), is an algorithm largely used to find a solution of the likelihood equation in the situations of the incomplete data. A suitable formulation is needed to facilitate the application of EM algorithm in our context. We present initially the algorithm in its general information.

We note \mathcal{T} the random vector corresponding to the data observed t . The probability distribution of \mathcal{T} is $f(t; \theta)$, where θ is the vectorial parameter of the statistical model. Moreover, t^c the vector of complete data with the distribution function $f^c(t^c; \theta)$, and u indicate the vector of the missing data, then $t^c = (t, u)$. McLachlan and Krishnan (1997) presented an extension of EM algorithm so that the vector of data observed is foreseen according to complete data t^c , of which a relation is resulted as follows:

$$f(t; \theta) = \int_{\mathcal{X}(t)} f^c(t^c; \theta) dt^c \quad (5)$$

When in two spaces \mathcal{X} and t , we examine the vector of incomplete data $t = t(x)$ in t instead of examining the vector of complete data t^c in \mathcal{X} . Moreover, there are several traces of \mathcal{X} surrounding to t . We note:

- $\mathcal{L}(\theta; t)$ and $l(\theta; t)$, respectively the likelihood and log-likelihood of data observed;
- $\mathcal{L}^c(\theta; t^c)$ and $l^c(\theta; t^c)$, respectively the likelihood and log-likelihood of de complete data;
- $Q(\theta/\theta^{(h)}) = \mathbb{E}[l^c(\theta; t^c)/t, \theta^{(h)}]$

where $\theta^{(h)}$ is the current estimate of the parameter.

In the EM algorithm, the objective is not to maximize $l(\theta; t)$ directly in seen to obtain the *MLE*. Nevertheless, we maximize repeatedly $l^c(\theta; t^c)$ in average on all the possible values of the missing data u . In fact, it is the objective function $Q(\theta/\theta^{(h)})$ which is to be maximized repeatedly.

B. The use of the EM algorithm

n instants of failure are considered on an obviously reparable system, noted $= (t_1, \dots, t_n)$. In the example, certain observations can be censured on the right. The vector of the parameters θ of the model is $(\eta_0, \eta_1, \eta_2, \beta_1, \beta_2)$, γ_0 and γ_1 are fixed at two unspecified values, checking $\gamma_0 < \gamma_1$.

In order to simplify the formulas, we note:

- $\lambda_H(t) = \frac{1}{\eta_0}$ the failure intensity of the Homogeneous Poisson Process (HPP);
- $\lambda_{NH_1}(t) = \frac{\beta_1}{\eta_1^{\beta_1}} (t^{\beta_1-1} - \gamma_0^{\beta_1-1}) \cdot \mathbf{1}_{[0, \gamma_0[}(t)$ the failure intensity of the Non-Homogeneous Poisson Process (NHPP) I;

- $\lambda_{NH_{II}}(t) = \frac{\beta_2}{\eta_2} \left(\frac{t-\gamma_1}{\eta_2}\right)^{\beta_2-1} \cdot \mathbf{1}_{[\gamma_1, +\infty)}(t)$ the failure intensity of the NHPP II.

In this case, the missing data are the indicators $u_i = (u_i^H, u_i^{NH_I}, u_i^{NH_{II}})$ that a failure is accidental ($u_i^H = 1$), or it is due to degradation ($u_i^{NH_I} = 1, u_i^{NH_{II}} = 1$), respectively either to the youth period, or to the marked degradation period.

We note then that the data observed t_i are achievements of the random variable $\min(\mathcal{H}_i, \mathcal{W}_i^I, \mathcal{W}_i^{II}, \mathcal{C}_i)$, while the vector \mathcal{H} (respectively $\mathcal{W}^I, \mathcal{W}^{II}$) is a HPP (respectively two forms of Weibull) and \mathcal{C}_i is the censure instant of the i^{th} observation.

Thus, the complete data are $t_i^c = (t_i, u_i)$. The complete log-likelihood is written:

$$l^c(\theta; t^c) = \ln \left[\prod_{i=1}^n (\lambda_{NH_I}(t_i))^{u_i^{NH_I}} (\lambda_H(t_i))^{u_i^H} (\lambda_{NH_{II}}(t_i))^{u_i^{NH_{II}}} \cdot e^{-\frac{1}{\eta_0} t_n - \left(\frac{t_n - \gamma_1}{\eta_2}\right)^{\beta_2}} \right]$$

$$l^c(\theta; t^c) = \sum_{i=1}^n [u_i^{NH_I} \ln(\lambda_{NH_I}(t_i)) + u_i^H \ln(\lambda_H(t_i)) + u_i^{NH_{II}} \ln(\lambda_{NH_{II}}(t_i))] - \left(\frac{\gamma_0}{\eta_1}\right)^{\beta_1} - \frac{1}{\eta_0} t_n - \left(\frac{t_n - \gamma_1}{\eta_2}\right)^{\beta_2}$$

Thereafter, we calculate the conditional expectation $Q(\theta/\theta^{(h)})$.

$$Q(\theta/\theta^{(h)}) = \mathbb{E}[l^c(\theta; t^c)/t, \theta^{(h)}] = \sum_{i=1}^n \mathbb{E}(u_i^{NH_I}/t, \theta^{(h)}) \ln(\lambda_{NH_I}(t_i)) + \mathbb{E}(u_i^H/t, \theta^{(h)}) \ln(\lambda_H(t_i)) + \mathbb{E}(u_i^{NH_{II}}/t, \theta^{(h)}) \ln(\lambda_{NH_{II}}(t_i))] - \left(\frac{\gamma_0}{\eta_1}\right)^{\beta_1} - \frac{1}{\eta_0} t_n - \left(\frac{t_n - \gamma_1}{\eta_2}\right)^{\beta_2}$$

$$= \left[\sum_{i=1}^n \tilde{\phi}_{NH_I}(t_i) \ln(\lambda_{NH_I}(t_i)) + \tilde{\phi}_H(t_i) \ln(\lambda_H(t_i)) + \tilde{\phi}_{NH_{II}}(t_i) \ln(\lambda_{NH_{II}}(t_i)) \right] - \left(\frac{\gamma_0}{\eta_1}\right)^{\beta_1} - \frac{1}{\eta_0} t_n - \left(\frac{t_n - \gamma_1}{\eta_2}\right)^{\beta_2}$$

with regard to

$$\tilde{\phi}_H(t_i) = \mathbb{E}(u_i^H/t, \theta^{(h)}) = \begin{cases} 0 & \text{if } t_i \text{ is a censure} \\ \Pr(u_i^H = 1/t_i, \theta^{(h)}) = \Pr(\mathcal{H}_i \leq \mathcal{W}_i^I, \mathcal{H}_i \leq \mathcal{W}_i^{II}/t_i, \theta^{(h)}) & \text{if not} \\ = \frac{\lambda_H(t_i)}{\lambda_{NH_I}(t_i) + \lambda_H(t_i) + \lambda_{NH_{II}}(t_i)} & \end{cases}$$

It is similarly

$$\tilde{\phi}_{NH}(t_i) = \mathbb{E}(u_i^{NH}/t, \theta^{(h)}) = \begin{cases} 0 & \text{if } t_i \text{ is a censure} \\ \Pr(u_i^{NH_I} = 1/t_i, \theta^{(h)}) = \Pr(\mathcal{W}_i^I \leq \mathcal{H}_i, \mathcal{W}_i^I \leq \mathcal{W}_i^{II}/t_i, \theta^{(h)}) & \text{if not} \\ = \frac{\lambda_{NH_I}(t_i)}{\lambda_{NH_I}(t_i) + \lambda_H(t_i) + \lambda_{NH_{II}}(t_i)} & \end{cases}$$

Subsequently, the stage of maximization is approved by the decomposition of $Q(\theta/\theta^{(h)})$. Indeed, it was carried out while maximizing separately $Q_H(\theta/\theta^{(h)})$, $Q_{NH_I}(\theta/\theta^{(h)})$ and

$Q_{NH_{II}}(\theta/\theta^{(h)})$, since the first component utilizes only the parameter η_0 . The second component, parameters η_1, β_1 , and the third component utilizes only the parameters η_2, β_2 .

Formally, the iteration $h + 1$ of the EM algorithm requires the following calculations:

1. $\eta_0^{(h+1)} = \frac{t_n}{\sum_{i=1}^n \tilde{\phi}_H(t_i)}$
2. $\beta_1^{(h+1)} = \frac{\sum_{i \in \Omega} \tilde{\phi}_{NH_I}(t_i) [\ln(\sum_{i \in \Omega} \tilde{\phi}_{NH_I}(t_i)) - 1]}{2 \sum_{i \in \Omega} [\tilde{\phi}_{NH_I}(t_i) \ln(\frac{\gamma_0}{t_i})]}$
3. $\eta_1^{(h+1)} = \frac{\gamma_0}{[\sum_{i \in \Omega} \tilde{\phi}_{NH_I}(t_i)]^{\frac{1}{\beta_1^{(h+1)}}$
4. $\beta_2^{(h+1)} = \frac{\sum_{i \in \Phi} \tilde{\phi}_{NH_{II}}(t_i)}{\sum_{i \in \Phi} [\tilde{\phi}_{NH_{II}}(t_i) \ln(\frac{t_n - \gamma_1}{t_i - \gamma_1})]}$
5. $\eta_2^{(h+1)} = \frac{t_n - \gamma_1}{[\sum_{i \in \Phi} \tilde{\phi}_{NH_{II}}(t_i)]^{\frac{1}{\beta_2^{(h+1)}}$

IV. NUMERICAL EXAMPLES

The data analysis is based on real example concerning reparable system (hydraulic pump) about nuclear sector of France which was used in Bertholon and *al.* (2004). The studied system retains a hydraulic pump on which the observation of 6 successive failures are used (18 months, 30, 82, 113, 121, 126). The estimation of model's parameters using the EM algorithm gives the following results:

- γ_0 and γ_1 , respective instants of improvement end and degradation beginning are estimated to 26.685 and 101.412.
- The reverse of accidental failure rate η_0 is estimated by $\hat{\eta}_0 = 43.855$.
- The scale parameters η_1 and η_2 are estimated respectively by $\hat{\eta}_1 = 4.409$ and $\hat{\eta}_2 = 4.361$.
- The shape parameters β_1 and β_2 are estimated respectively by $\hat{\beta}_1 = 1.098$ and $\hat{\beta}_2 = 3.000$.

In order to obtain a solid numerical results, a Monte-Carlo simulation is employed, allowing to compare the estimation of our model by MLE and EM algorithms. Two different cases are presented as follow:

- The first case retains 100 simulations of 50 size sample of our model with parameters $\eta_0 = 1, \eta_1 = 1, \beta_1 = 0.5, \eta_2 = 1, \beta_2 = 2, \gamma_0 = 30, \gamma_1 = 100$.
- The second case retains 100 simulations of 50 size sample of our model with the same parameters except for $\beta_2 = 3$.

The results are stated in form of mean and a 95% confidence interval. The next table presents the estimation results:

TABLE I. SIMULATION RESULTS

		First case ($\beta_2 = 2$)		Second case ($\beta_2 = 3$)	
		Mean	C I	Mean	C I
MLE	$\hat{\eta}_0$	1.705	[1.411, 1.998]	1.331	[1.096, 1.565]
	$\hat{\eta}_1$	0.850	[0.659, 1.041]	0.968	[0.810, 1.127]
	$\hat{\eta}_2$	0.880	[0.656, 1.104]	1.117	[0.935, 1.299]
	$\hat{\gamma}_0$	30.454	[25.130, 35.778]	35.085	[49.625, 60.546]
	$\hat{\gamma}_1$	104.72	[99.153, 110.29]	109.695	[104.52, 114.863]
EM	$\hat{\eta}_0$	1.881	[1.527, 2.234]	1.156	[0.985, 1.326]
	$\hat{\eta}_1$	0.801	[0.592, 1.008]	0.999	[0.875, 1.123]
	$\hat{\beta}_1$	0.345	[0.184, 0.506]	0.454	[0.411, 0.496]
	$\hat{\eta}_2$	0.799	[0.559, 1.038]	1.094	[0.959, 1.230]
	$\hat{\beta}_2$	1.776	[1.464, 2.089]	2.783	[2.602, 2.963]
	$\hat{\gamma}_0$	31.645	[27.091, 36.198]	32.560	[27.493, 37.627]
	$\hat{\gamma}_1$	105.63	[100.03, 111.25]	109.229	[104.459, 113.998]

V. DISCUSSION

In the long run, following the results of preceding tests, the failures process is a NHPP. The empirical cumulative distribution function of real data is evolved in the same direction as the simulated one. This process is then managed by our reliability model. Consequently, the effects of estimation show the way that there is an improvement of the system until the second failure (during 2.2 years of operation) and degradation starts from the fourth failure (beyond 8.5 years of operation). Considering the same unit of data over the improvement period and of degradation, the scale parameters η_1 and η_2 over these two periods do not have a significant difference. This can be easily guaranteed with skew of an averages difference traditional test. The estimate value of β_2 is higher than 2, the failure intensity is increasing and convex announcing a marginal progress in degradation state. At the same time, β_1 takes an estimate value very near to 1 indicates that the intensity is practically constant. as a result, the failures are rather accidental and cannot be due to youth diseases. This purified model of improvement period, which is presented in Bertholon and *al.* (2004), remains able independently to concretize the hydraulic pump behavior. In light of simulations, we state the following criticisms:

- The $\hat{\eta}_j$ ($j = 0, 1$ ou 2) have the best behavior to one side for the first case where $\hat{\eta}_0$ appears to degrade.
- The $\hat{\gamma}_j$ ($j = 0$ ou 1) are all acceptable.
- The variability of $\hat{\beta}_j$ ($j = 1$ ou 2) is significant enough.

As a final point, the EM procedures offer better estimators for the second case. The values of β_2 are rather higher than 2, then the curve is convex over the degradation period as it is presented in our model. A potential limitation of our model is that it involves seven parameters. In fact, it is difficult to estimate these parameters for small-sized and/or censored samples. For this reason even, the MLE appear more reliable for industrial applications.

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Detecting Distributed Denial of Service Attacks Using Data Mining Techniques

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Abstract—Users and organizations find it continuously challenging to deal with distributed denial of service (DDoS) attacks. The security engineer works to keep a service available at all times by dealing with intruder attacks. The intrusion-detection system (IDS) is one of the solutions to detecting and classifying any anomalous behavior. The IDS system should always be updated with the latest intruder attack deterrents to preserve the confidentiality, integrity and availability of the service. In this paper, a new dataset is collected because there were no common data sets that contain modern DDoS attacks in different network layers, such as (SIDDoS, HTTP Flood). This work incorporates three well-known classification techniques: Multilayer Perceptron (MLP), Naïve Bayes and Random Forest. The experimental results show that MLP achieved the highest accuracy rate (98.63%).

Keywords—DDoS; IDS; MLP; Naïve Bayes; Random Forest

I. INTRODUCTION

Network security has become of utmost importance in all areas of business and industry, including bank transactions, Email, social media and university eServices, etc. Recently, web and network services have suffered from intruder attacks. Hackers are continually generating new types of Distributed Denial of Service (DDoS) which work on the application layer as well as the network layer. The vulnerabilities in the above mentioned areas allow hackers to deny access to web services and slow down access to network resources.

The IDS system is one of the most common solutions to dealing with the problem of DDoS and preserving the confidentiality, integrity and availability of web services and computer network resources. IDS system uses machine learning techniques to detect and classify types of DDoS in an intelligent way and will eliminate intrusion without referral to the System Security Officer (SSO), however achieving one hundred percent accuracy in detecting and classifying all new types of attacks is hard to achieve. Naïve

Many types of DDoS attacks are already known, such as a Smurf attack, which sends large numbers of Internet controlled message protocol packets to the intended victims. Another type of DDoS is R-U-Dead-Yet (RUDY), which simply consumes

all available sessions of a web application which means sessions will never end. In other words, the web service will be unavailable for any new request from new users. One of the most up-to-date DDoS types is HTTP POST/GET, where attackers send a completely legitimate posted messages at a very slow rate, such as (1 byte/240 second), into a web server that is hosting a web application. This type of DDoS will have a harmful effect on a web service and cause it to slow down temporarily and interrupting the service. Another type of modern DDoS attack is an SQL Injection Dos (SIDDoS) where attackers insert a malicious SQL statement as a string that will pass to a website's database as an equation (e.g., through the input values in the website form), then illegally allowing access to the resources or to the stored data on servers.

Unfortunately, most of the common open access data sets have duplicated and redundant instances, which make the detection and classification of the DDoS unrealistic and ineffectual. There are also no available data sets (e.g. KDD 99) which include new DDoS types, such as HTTP flood and SIDDOS. In this research, we collected a completely new dataset in a controlled environment, which includes four harmful types of attack namely: UDP flood, Smurf, HTTP Flood and SIDDOS.

Machine learning is used to detect and classify network traffic based on some features (average packet size, inter arrival time, packet size, packet rate, bit rate, etc.) that are used to measure and determine whether the network traffic is normal or is a type of DDoS. DDoS attacks mostly have the same average packet size. The number of packets will increase in the attacked packet rather than the normal packet; also, the inter arrival time will be too small to allow attackers to consume resources rapidly. DDoS packets always have a high bit rate for network layer attack. Attackers focus on any attributes that help them to consume resources and make the service unavailable to end users.

In this work, we make the following contributions:

1) A new dataset has been collected including modern types of attack, which were not been used (to the best of our knowledge) in previous research. The dataset contains 27

features and five classes. A network simulator (NS2) is used in this work, because NS2 can be used with high confidence due to its capability of producing valid results that reflect a real environment.

2) The collected data has been recorded for different types of attack that target the most critical network layers (Application and network). It should be known that this data has never been collected before.

3) Three machine learning algorithms are applied using the collected dataset to classify the DDoS types of attack.

The remainder of the paper is organized as follows: Section II presents the related work and provides a brief discussion of machine learning classifiers in the relevant area. In Section III, we present the architecture of intrusion detection system techniques. Section IV describes the possibility of an attack in the application layer and network layer, followed by details about the dataset collection in Section V. Section VI describes the IDS classifiers used in this paper. In section VII, we describe the evaluation metrics used to assess the performance of the classifiers used; the section also discusses the experiments and the achieved results. Finally, Section VIII concludes the the paper and describes the future work.

II. RELATED WORK

Early investigation of IDS was carried out by Georgios Loukas and Glay [1], who began their work with the time-line significant DOS. In September 1996, a SYN Flood attack was discovered, a smurf attack began in January 1998 and a HTTP flood was the modern DOS that began in 2004. Specialists in the area suggest complete protection architecture through detection which can either be anomaly-based or signature based, or a hybrid of these two methods. Classifications such as neural networks, radial basis functions and genetic algorithms are increasingly used in DoS detection because of the automatic classification they can offer. The protection system either drops the attacking packets in a timely fashion or renders them inoperable. Traffic rate, SYN and URG flags, as well as some specific ranges of ports, are the most significant for the identification of a DoS attack, as some investigators have tested in their surveys.

Guiomar et al. [2] have implemented a Network Intrusion Detection System (NIDS) using OPNET simulation. This was based on misuse detection and network traffic was imported using an ACE module into OPNET. A NMAP port scanner was used to simulate a flood attack, and the proposed IDS was tested in a controlled environment; the result was satisfactory with around 93% accuracy rate.

Different techniques have been used for IDS. Chandrika Palagiri showed that a modelling network can achieve a realistic result to demonstrate a Neural Network, especially for an individual attack. They also used a support vector machine (SVMs) as a clustering approach with MLP which is a very useful technique to present a uniform or clustering group. Researchers often focus on a Neural Network that can make decisions quickly and for real-time detection [3].

Sujay Apale, Rupesh Kamble and others [4] on the layer seven DDoS flooding attacks because such attacks are becoming more severe and are increasing in occurrence.

Different machine learning techniques were used for the intrusion detection system. Some of these techniques achieved an acceptable detection rate, while others achieved only a poor detection rate. Moreover, the Naïve Bayes (NB) algorithm enables a faster learning/training speed than other machine learning algorithms, and it has a greater accuracy rate in classification and detection of a layer seven attack.

Kejie et al. [5] proposed a framework to detect DDoS attacks and identify attack packets efficiently. The purpose of the framework is to exploit spatial and temporal correlation of DDoS attack traffic. Such techniques can accurately detect DDoS attacks and identify attack packets without modifying existing IP forwarding mechanisms at the routers. This work achieved 97% for detection probability using the proposed framework.

Further, a hybrid Neural Network technique was used by Wei Pan and Weihua Li, who proposed a hybrid Neural Network consisting of a self-organizing map (SOM) and radial basis functions to detect and classify DDoS attacks. The proposed technique achieved a satisfactory accuracy rate result for detecting and classifying DDoS attacks [6].

Norouzian et al. [7] presented a most effective classification technique for detecting and classifying attacks into two groups normal or threat. They proposed a new approach to IDS based on a MultiLayer Perceptron Neural Network to detect and classify data into 6 groups. They implemented their MLP design with two hidden layers of neurons and achieved 90.78% accuracy rate.

A NIDS using a 2-layered, feed-forward neural network was proposed by Haddadi, F, Sara Khanchi and others [8]. The proposed system classified normal connections and attacks. Different types of attacks were determined, and they focused on using training function, data validation and a preprocess dataset that caused less memory usage, minimum resource consumption and faster training. After implementing the proposed system on a KDD cup 99 dataset, the result was very satisfactory, both on accuracy rate and performance.

Zheng et al. [9] implemented a Hierarchical Intrusion Detection (HIDE) system which can detect attacks using preprocessing statistical values and a Neural Network. They tested five classifiers: Perceptron, Backpropagation (BP), Perceptron-backpropagation-hybrid (PBH), Fuzzy ARTMAP, and Radial-based Function. They stated that BP and PBH achieved the highest accuracy rate for detecting and classifying network attacks.

Reyhaneh Karimazad and Ahmad Faraahi [10] proposed an anomaly-based DDoS detection approach using an analysis of network traffic. A radial-based function (RBF) Neural Network was used in this approach, and they tested their method on a UCLA dataset, achieving 96% accuracy rate for a DDoS attack.

Other work on preventing/ avoiding attacks by means of, for example, fuzzy clustering, genetic algorithm, and artificial neural network (ANN) has been conducted. Ms. Jawale and Prof. Bhusari presented research on ANN that achieved the highest accuracy rate. They proposed a system that uses multilayer perceptrons, back propagation and a support vector

machine, consisting of multi modules such as packet collection and preprocessing data. This system achieved 90.78% detection rate [11].

An overview and broad classification of IDS was presented by Mohammed Alenezi and Martin J Reed. The difficulties and characteristics of DoS/DDoS attacks are discussed in this research. Three different classifications were chosen. They focus on general DoS and flooding attacks. The CUSUM approach has many advantages over statistical techniques, which this research effectively demonstrates [12].

Mouhammd Alkasassbeh and Mo Adda [13] showed in their work that mobile agent technology combined with statistical methods based on the Wiener filter can effectively provide the ability to detect network intrusion attempts. Wiener statistical filtering takes advantage of the correlation matrix between the input management information base (MIB) variables and cross correlation with the desired MIB variables to detect abnormal traffic. The MIB variables give a clear image when an abrupt change in network traffic occurs, for example, when a number of PCs start flooding the server with requests. This excessive number of ftp requests stresses the server, which leads to it crashing.

In [14], Dimitris Gorillas and Evangelos Dermatas presented and evaluated a Radial-basis-function (RFB) Neural Network for DDoS attacks dependent on statistical vectors through short-time window analysis. The proposed method was tested and evaluated in a controlled environment with an accuracy rate of 98% of DDoS detection.

III. INTRUSION DETECTION SYSTEM (IDS)

An intrusion detection system is one of the solutions which can be used to prevent an intruder from launching a DDoS attack in a protected network. An effective IDS can detect a new DDoS in a short time without human intervention. An IDS system can be categorized into two types as follows.

- Host Intrusion Detection System (HIDS): this type of IDS can be implemented in network devices or workstations. HIDS techniques can be used to prevent a DDoS attack on a selected device, but it does not support monitor of a whole network.
- Network Intrusion Detection System (NIDS): this type of IDS can be implemented as a security strategy within a protected network, and can be used to detect and classify all network traffic from all devices. In this research, we will apply an NIDS security strategy in a network [12]. Figure 1 shows a NIDS and HIDS structure.

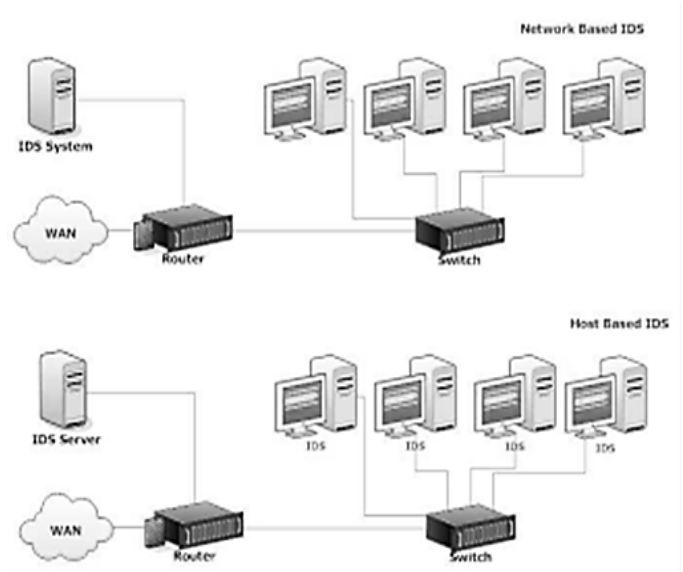


Fig. 1. NIDS and HIDS Structure

IDS can be applied, either anomaly-based or signature based, to detect and classify network traffic. The anomaly based is used to compare network traffic behavior with historical baseline data and so it requires training data to work logically. A signature-based focus is placed on each independent packet, and is then compared with the stored signature or known intruder attack. Detection time for the signature based is faster than the anomaly-based, as the anomaly-based requires training data, while the signature-based requires a stored signature. Figure 2 shows general IDS classification.

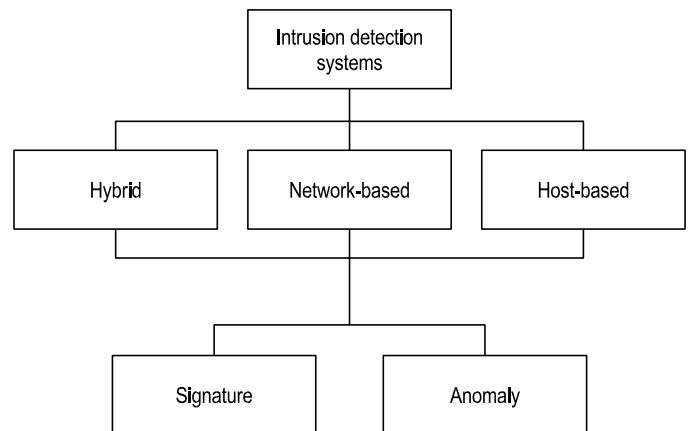


Fig. 2. IDS Classification

IV. POSSIBILITIES OF ATTACKS

DDoS attacks; depending on the target protocol, different types of DDoS attack can be implemented on OSI layers; Figure 3 presents the seven layers of OSI. In this paper, we describe the testing of the most recent attacks on the application layer and the network layer.

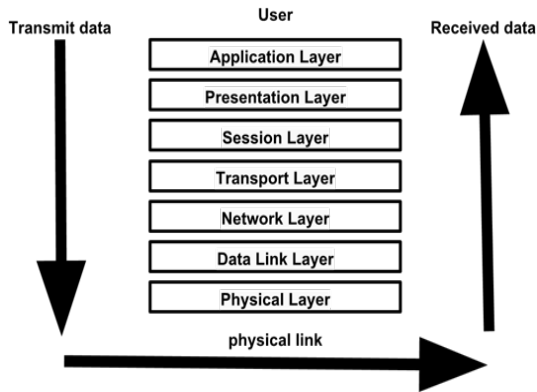


Fig. 3. OSI Seven Layers

A. Network Attack Layer

Attackers implement their DDoS on the victim servers by consuming the servers resources until the service becomes unavailable to all users. A Smurf attack and User Datagram Protocol (UDP) flood attack are part of the network layer attack where an attacker uses malicious (TCP / UDP) network traffic to deny access to the server service. Attackers implement a Smurf attack on malicious network traffic by spoofing IP addresses in a network for sending a large number of Internet Control Message Protocol (ICMP) floods to the victim machine. A large number of ICMPs causes denial of access to the server's services. Figure 4 shows how an attacker implements a Smurf attack on a network.

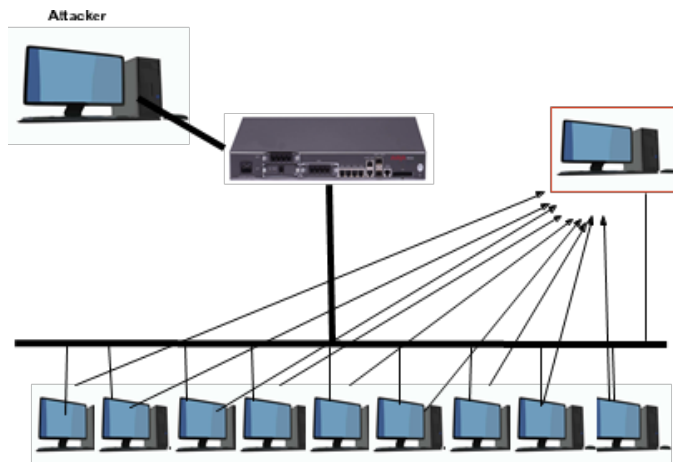


Fig. 4. Smurf Attack Structure

Another type of network layer attack is a UDP flood, where the UDP is a connectionless protocol. An attacker sends a command to the slave machine to use the network workstation as part of the attack. All the workstations send a great amount of UDP traffic to the victim server [15]. Figure 5 indicates how attackers implement a UDP flood on the network.

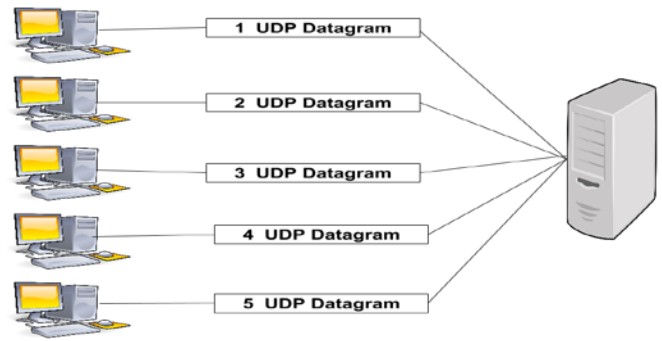


Fig. 5. UDP Flood Structure

B. Application Attack Layer

One of the most common protocols supported by the application layer is HTTP protocol. Any web applications implemented over HTTP are accessible from anywhere, so this makes them difficult to detect and classifying application layer attacks is difficult to prevent. In most situations, intruders send the completely legitimate request for service, and the attackers act as a normal user request service from the server. Figure 6 shows web applications on the World Wide Web.

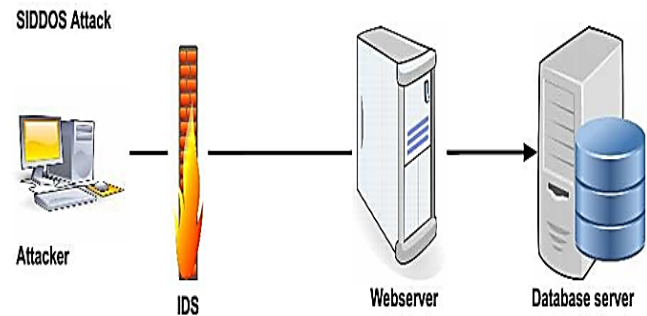


Fig. 6. Web Application Structure

Moreover, most modern DDoS application layer attacks are SQL Injection Distributed Denial of Service (SIDDOS), where Attackers start from the client side, for example the browsers, by inserting a malicious code, and forwarding it to the server side (e.g., input inbox values in a web form) [16]. A SIDDOS attack consumes the servers resources if the malicious code is then forwarded to the servers execution indefinitely. On the other hand, a SIDDOS attack can also be used to steal personal information and make the service unavailable for clients by changing their personal information.

V. DATASET COLLECTION

A new dataset was collected in this work because there is no existing data sets that contain a modern DDoS attack such as (SIDDOS, HTTP Flood), and furthermore, other available data sets may include a great deal of duplicate and redundant records, and that may result in an ultimate unrealistic outcome. Our collected dataset contains four types of DDoS attack as follows: (HTTP Flood, SIDDOS, UDP Flood, and Smurf) without redundant and duplicate records. Table 1 lists number of records of these types of attack. Table 2 shows the dataset features we dealt with.

TABLE I. DISTRIBUTION OF ATTACKS

Attack Name	No. of Records
Smurf	12590
UDP Flood	201344
SIDDOS	6665
HTTP Flood	4110

TABLE II. EXTRACTED DATA SET FEATURES

Variable No	Description	Type
1	SRC ADD	continuous
2	DES ADD	Continuous
3	PKT ID	Continuous
4	FROM NODE	Continuous
5	TO NODE	Continuous
6	PKT TYPE	Continuous
7	PKT SIZE	Continuous
8	FLAGS	Symbolic
9	FID	continuous
10	SEQ NUMBER	continuous
11	NUMBER OF PKT	continuous
12	NUMBER OF BYTE	continuous
13	NODE NAME FROM	Symbolic
14	NODE NAME TO	Symbolic
15	PKT IN	continuous
16	PKTOUT	continuous
17	PKTR	continuous
18	PKT DELAY NODE	continuous
19	PKTRATE	continuous
20	BYTE RATE	continuous
21	PKT AVG SIZE	continuous
22	UTILIZATION	continuous
23	PKT DELAY	continuous
24	PKT SEND TIME	continuous
25	PKT RESEVED TIME	continuous
26	FIRST PKT SENT	continuous
27	LAST PKT RESEVED	continuous

The proposed system that is used to collect a new dataset is organized as illustrated in Figure 7. The selected classifiers were tested in 734,627 records which they were fully randomized to obtain realistic results.

- Collection and auditing: in this step, all network traffic is collected and audited from NIDS.
- Preprocessing file format: redundant and duplicate records are removed.
- Feature extraction: extracts features parameters from the collected network traffic and assigns each feature to the first column; these will be used as a vector in the new dataset.
- Statistical measurements: in this step additional features are calculated using statistical equations.

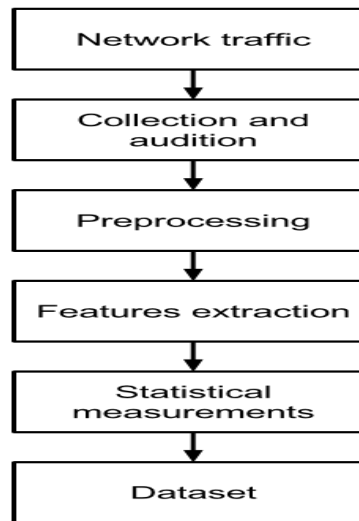


Fig. 7. Dataset Collection Steps

VI. IDS CLASSIFIERS

In this work, we investigated and tested three different classifiers based on the dataset collected and described in the previous section. The models are the MLP, Random Forest and Naïve Bayes classifiers. The models are described as follows.

A. MLP-ANN

The most common and well-known Feedforward Neural Network (FFNN) model is called MLP. MLP has been successfully applied in a number of applications, including regression, classification, or time series prediction problems using simple auto-regressive models [17] [18] [19]. The structure of a simple MLP network is clarified in Figure 8.

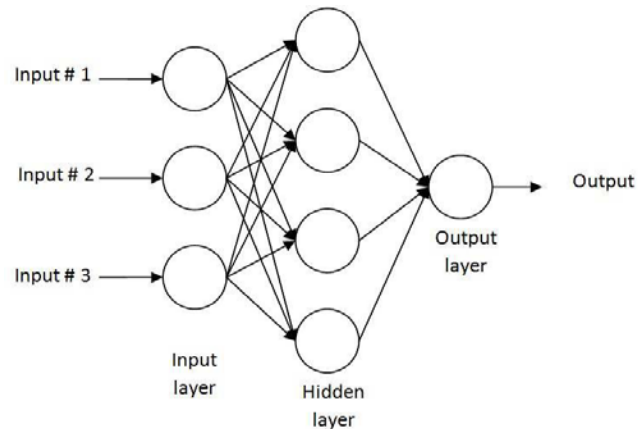


Fig. 8. MLP Structure

MLP permits the data flow to travel one way, from input to output. There is no feedback; it tends to be straight-forward networks that companion inputs with outputs. According to [20] [21], any MLP network can be distinguished by a number of performance characteristics, which can be summarized in three points:

- Neural Network Architecture: Overall, MLP architecture can be clarified as the pattern of connections between the neurons in different layers.

The architecture consists of three layers: input layer, hidden layers, and output layer. Two nodes of each end-to-end layer are connected. Furthermore, MLP is always fully connected. Each link has a weight which is limited based on the training algorithm. More complex architectures have more layers.

- Training Algorithm: The method of selecting one model from a set of models, which determines the weights of the connections.
- Transfer Function: Transfer Function is applied by each neuron to its net input to determine its output signal. This function is usually non-linear.

Sigmoid Function is one of the most commonly used transfer functions. The use of the sigmoid function has an advantage in neural networks trained by a backpropagation learning algorithm. The sigmoid function and other common transfer functions are listed in Table 3.

TABLE III. EXAMPLES OF SOME COMMON TRANSFER FUNCTIONS

Transfer Function	Detention
Linear	$f(x) = x$
Segmoid	$f(x) = \frac{1}{1 + e^{-x}}$
Hyperbolic	$f(x) = \frac{(e^x) - (e^{-x})}{1 + e^{-x}}$
Hard Limit	$f(x) = \begin{cases} 0, & x < 0 \\ 1, & x \geq 0 \end{cases}$
Symmetric Hard Limit	$f(x) = \begin{cases} -1, & x < 0 \\ 1, & x \geq 0 \end{cases}$

In order to understand the algorithm of the learning process on MLP, suppose that a given MLP has N neurons in the input layer and M neurons in the hidden layers, and one output neuron. The learning process can be divided into a number of stages and is described as follows:

- Hidden layer stage: Given a number of inputs φ_i and a set of corresponding weights between the input and hidden neurons w_{ij} , the outputs of all neurons in the hidden layer are calculated by (1) and (2):

$$O_i = \sum_{i=0}^n w_{ij} \varphi_i \quad (1)$$

$$y_i = z(O_j) \quad (2)$$

Where $i = 1, 2, \dots, N$ and $j = 1, 2, \dots, M$. z and y_j are the activation function and output of the j th node in the hidden layer, respectively. z is usually a sigmoid function given in 3.

$$z(x) = \frac{1}{1 + e^{-x}} \quad (3)$$

- Output stage: The outputs of all neurons in the output layer are given in 4. l is defined as the number of neurons in the output layer. For simplicity, l is one.

$$Y^* = f\left(X \sum_{j=0}^m w_{jl} y_j^H\right) \quad (4)$$

f_0 is the activation function of the output layer which is

usually a linear function. Y^* is the neural network output from the single neuron in the output layer. The MLP network is attempting to minimize the Error via the classical Back-propagation (BP) Training algorithm. BP learning starts with all weights initialized randomly, then weights are modified as the algorithm progresses until steady state values are reached.

- Error validation stage: ANN continues the learning process until the error minimization criteria are reached. Assuming that the desired output is Y and the produced ANN output is Y^* , the learning process should stop when the error difference given in Equation 5 is minimal. T is the total number of observations used to build the ANN model during training. Another set of data must be used to validate the developed ANN model performance.

$$Error = \frac{1}{T} \sum_{i=1}^T (Y_i - Y^*_i)^2 \quad (5)$$

In MLP, all the network weights and bias values are assigned with random values initially, and the goal of the training is to find the set of network weights that cause the output of the network to match the real values as closely as possible. However, we cannot forget that the MLP always takes the longest time for training [8].

B. Random Forest

Random forest classifiers were developed by LEO Breiman and Adele Cutler. They combine tree classifiers to predict new unlabeled data, the predictor depends on the number of as that are represented by the number of trees in the forest, the attributes are selected randomly, each number of trees represents a single forest and each forest represents a predation class for new unlabeled data [4]. In this algorithm, random features selection will be selected for each individual tree. A random forest classifier ensemble learning algorithm is used for classification and prediction of the outputs based on an individual number of trees [22]. Using random forest classifiers, many classification trees will be generated, and each individual tree is constructed by a different part of the general dataset. After each tree is classified in an unlabeled class, a new object will be implemented under each tree vote for decision. The forest chosen as the winner is based on the highest number of votes recorded. Figure 9 shows decision forest architecture and how the number of votes is calculated. Random forest algorithms:

- If you have a dataset you need to split n samples from the whole dataset, so that now we have (n samples = number of trees).
- Each dataset sample needs to be regressed or classified for each record randomly split among all predictor classes to reach an approximately optimal split. bagging can be learned as a special scenario when $m(\text{tries}) = P$ (number of predictors).
- Predict unlabeled class based on a reassembled number of aggregation predictions re number of trees.

Decision Forest

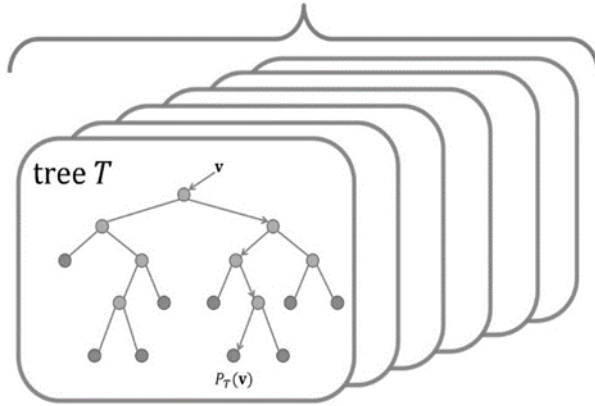


Fig. 9. Decision Forest Architecture

Random Forest Tuning parameters The accuracy rate and error rate for Random Forest (RF) classifiers can be measured by splitting a whole dataset for testing, e.g. (40%) and for training, e.g. (60%). After the random forest, a model test (40%) can be used to calculate the error rate, and the accuracy rate can be measured based on comparing correctly classified instances with incorrectly classified instances. Out of bag (OOB) is another way of calculating the error rate. In this technique, there is no need to split the dataset because calculation occurs in the training phase. The following parameters need to be adjusted correctly in order to achieve the highest accuracy rate with minimum error rate:

- Number of trees.
- Number of descriptors that occurs randomly for present candidates $m(\text{tries})$.

After analysis and studying many cases, 500 trees are needed within the descriptor. Even if there is a great number of trees that will not achieve the highest accuracy rate and will only waste training time and resources [23], so that random forest tuning parameters are a vital research area that needs to be fine-tuned.

C. Naïve Bayes

Naïve Bayes [24] is a simple probabilistic classifier that returns $p(y|x)$, Naïve Bayes calculates probabilities for each class in a dataset and determines discriminative learning to predict values of the new class. The main formulation for Naïve Bayes may be found in [9]. A Naïve classifier link the dataset attributes $x \in X$ that are used as inputs to the class labels $Z \in \{1,2,,C\}$, where X is attributes space and Z is a class space. Let $X = \text{IRD}$ where D is a real number. Naïve classifier may be used with discrete and continuous attributes. This model is called a multi-label problem. The learning function that directly computes the class is called the discriminates model, the main purpose of which is to learn the conditional class that is used for nonlinear and multi-label problems. We will use the 6,7,8,9,10:

$$p(y|x) = \frac{p(x,y)}{p(x)} = \frac{p(\frac{x}{y})p(y)}{\sum_{y'=1}^c \frac{p(x/y')}{p(y')}} \quad (6)$$

Naïve classifier achieve outputs based on argument max function as follows:

$$f(x) = y'(x) = \text{arg}\{(\max)\left(p\left(\frac{y}{x}\right)\right)\} \quad (7)$$

Probabilistic classifiers have the following advantages:

- Option to reject: this is used when we are uncertain of the prediction result and so the prediction result can be ignored since human effort is present.
- Allow learn function to be changed: combinations of probability function can be used to achieve the best performance of the main issues if we use the direct learning function $p(\frac{y}{x})$ and the probability function is changed, so there is no need to re calculate $p(x,y)$.
- Balanced classes : some parts of the collected dataset have unbalanced classes which means that if we have one million records of normal network traffic where there is only 1 abnormal for 1000 records, that means we can directly train an unbalanced training dataset and easily achieve 99.9% accuracy rate by just using class always = normal. To deal with this problem, balanced classes are used.

$$p_{bal}((y|x)\alpha P(x,y)P_{bal}(y) \quad (8)$$

$$P \text{ true}(yx) \propto P \text{ true}(x,y)P \text{ true}(y) \propto \frac{p(\frac{y}{x})}{P_{bal}(y)} P \text{ true}(y) \quad (9)$$

- Models combinations it is very useful when the collected dataset contains a mix of feature types, such as if there is a collected dataset and each feature vector represents distinguished data types (text, images, numbers, etc.) Two or more kinds of attributes using model combinations means that we can build two or more classifiers, such as and so on [25]. To combine two different information sources the following formula is used:

$$P(x1, x2 | xy) = P(x1/y) P(x2/y) \quad (10)$$

VII. EXPERIMENTS AND RESULTS

In this work, the experiments were performed on Ubuntu 13.10 platform, Intel, Xeon (R) CPU E5-2680 @ 2.70 GHZ x 4, 12.0 GB RAM. A machine learning tool called WEKA version 3.7.12 was used for the application of the classification techniques.. To measure the efficiency of the algorithms, each algorithm was trained on our dataset using 66% of the collected data and the 34% were used as a test data. In fact, we tried the ten-fold validation but our previous partitioning worked better.

A. Evaluation Metrics

To evaluate the performance of the classifiers applied in this work, we use primary performance indicators based on confusion matrix shown in Table 4. This matrix contains information about real and predicted classifications carried out by the classification models. Performance metrics of such systems are commonly evaluated using the information in the matrix [26]. We use MLP, Random Forest and Naïve Bayes classifiers based on the dataset collected in this study. MLP

parameters are tuned as listed in Table 5. While for Random forest the ideal number of trees is empirically set to 50.

TABLE IV. CONFUSION MATRIX

		Predicted	
		Positive	Negative
True	Positive	TP	FN
	Negative	FP	TN

TABLE V. MLP SETTINGS

Parameter	value
Number of neurons in hidden layer	16
Learning rate	0.3
Momentum	0.2
Maximum number of epochs	500

- Accuracy - measures the rate of the correctly classified attack instances of both classes.

$$Accuracy = \frac{TP+TN}{TP+FN+FP+TN} \quad (11)$$

- Precision: It is the ratio of the number of relevant attacks retrieved to the total number of irrelevant and relevant attacks retrieved. It is also called positive predictive, which can be calculated by the following equation.

$$Precision = \frac{TP}{TP+FP} \quad (12)$$

- Recall: It is the ratio of the number of relevant attacks retrieved to the total number of relevant attacks. It is also called positive sensitivity value, which can be calculated by the following equation.

$$Recall = \frac{TP}{TP+FN} \quad (13)$$

B. Result Discussion

The classifiers were evaluated and assessed using the confusion matrix based on the evaluation metrics listed in section VII (A). The resultant confusion matrices for MLP, Random Forest and Naïve Bayes are shown in Tables 6, 7 and 8, respectively. From these confusion matrices we calculated the accuracy, precision and recall of the models. The overall accuracy was 98.63%, 98.02% and 96.91% for MLP, Random Forest and Naïve Bayes, correspondingly.

However, taking into consideration only the accuracy rate is not sufficient, especially when the data are imbalanced as in our case, where the number of instances in the normal class was much higher than the other classes. Therefore, the precision and recall were calculated for each class: Normal, UDP-Flood, Smurf, SIDDOS and the HTTP-FLOOD.

Figures 10 and 11 depict a comparison between the obtained precision and recall values of the developed models

for each class. According to the Figures, it can be noted that all classifiers achieved high precision and recall rates for the normal class. However, their performance varies regarding the other four classes of attacks.

MLP achieved high precision and recall results for the minority classes, while Naïve Bayes was the worst. It can be noted also that the Smurf class was the most challenging for all classifiers. This is due to the nature of Smurf type of attack, which aims to send large number of ICMP echo packet requests that are hard to be classified as normal or abnormal traffic. It should be noted that ICMP complements the missing flow control and traffic management of IP4. As a result, Random Forest and Naïve Bayes failed to show good rates for the Smurf class, while, on the other hand, MLP achieved a high precision rate. In general, it can be concluded that MLP is the best classifier for detecting DDoS with promising performance results.

TABLE VI. CONFUSION MATRIX FOR MLP

	Normal	UDP-Flood	Smurf	SIDDOS	HTTP-FLOOD
Normal	657961	0	0	70	20
UDP-Flood	6767	61765	0	0	0
Smurf	2817	0	1396	132	10
SIDDOS	115	0	0	2136	0
HTTP-FLOOD	0	0	0	86	1352

TABLE VII. CONFUSION MATRIX FOR RANDOM FOREST

	Normal	UDP-Flood	Smurf	SIDDOS	HTTP-FLOOD
Normal	653545	3117	1258	112	19
UDP-Flood	6728	61794	10	0	0
Smurf	2798	10	1414	121	12
SIDDOS	160	0	75	1972	44
HTTP-FLOOD	8	0	11	77	1342

TABLE VIII. CONFUSION MATRIX OF NAÏVE BAYES

	Normal	UDP-Flood	Smurf	SIDDOS	HTTP-FLOOD
Normal	646612	0	10375	981	83
UDP-Flood	6705	61765	59	3	0
Smurf	2717	0	92	140	1406
SIDDOS	115	0	16	2120	0
HTTP-FLOOD	0	0	0	86	1352

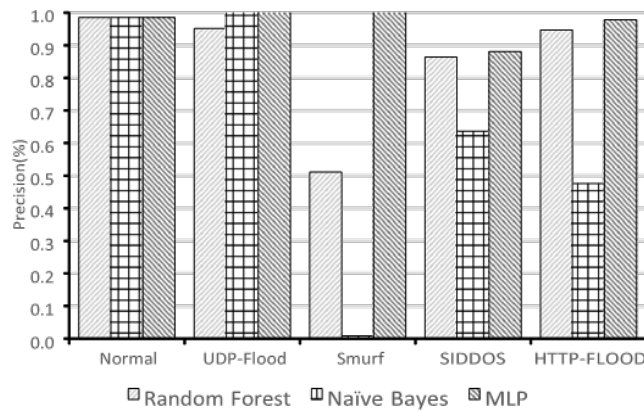


Fig. 10. Precision Results

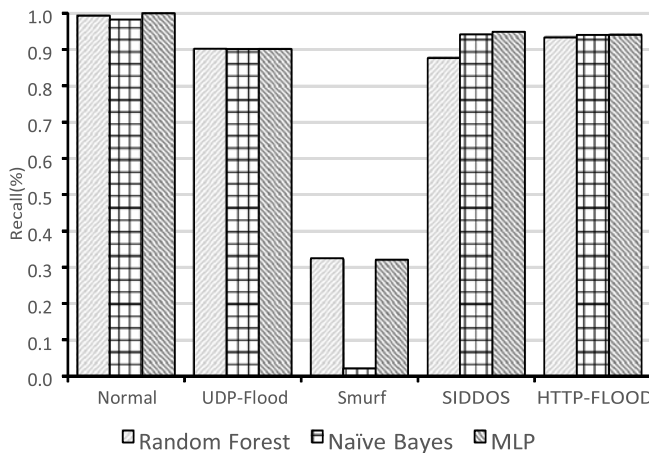


Fig. 11. Recall Results

VIII. CONCLUSIONS

In this paper, we collected a new dataset that includes modern types of attack, which were not been used in previous research. The dataset contains 27 features and five classes. A network simulator (NS2) was used in this work, because NS2 can be used with high confidence due to its capability of producing valid results that reflect a real environment. The collected data has been recorded for different types of attack that target the Application and network layers. Three machine learning algorithms (MLP, Random Forest, and Naïve Bayes) were applied on the collected dataset to classify the DDoS types of attack namely: Smurf, UDP-Flood, HTTP-Flood and SIDDOS. The MLP classifier achieved the highest accuracy rate.

The future work is to include more types of modern attacks in different OSI layers, such as the transport layer. In addition, we will examine different features selection techniques to choose the appropriate features.

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Investigating the Effect of Different Kernel Functions on the Performance of SVM for Recognizing Arabic Characters

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Abstract—A considerable progress in the recognition techniques of Latin and Chinese characters has been achieved. By contrast, Arabic Optical character Recognition is still lagging in spite that the interest and research in this area is becoming more intensive than before. This is because the Arabic is a cursive language, written from right to left, each character has two to four different forms according to its position in the word, and several characters are associated with complementary parts above, below, or inside the character. Support Vector Machines (SVMs) are used successfully for recognizing Latin, and Chinese characters. This paper studies the effect of different kernel functions on the performance of SVMs for recognizing Arabic characters. Eleven different kernel functions are used throughout this study. The objective is to specify which type of kernel functions gives the best recognition rate. The resulting kernel functions can be considered as base for future studies aiming at enhancing their performance. The obtained results show that Exponential and Laplacian Kernels give excellent performance, while others, like multi-quadratic kernel, fail to recognize the characters, specially with increased level of noise.

Keywords—SVM; Kernel Functions; Arabic Character Recognition

I. INTRODUCTION

Optical character recognition systems improve human-machine interaction and are urgently required for many governmental and commercial organizations. A considerable progress in the recognition techniques of Latin and Chinese characters has been achieved. By contrast, Arabic Optical character Recognition (AOCR) is still lagging because of the special characteristics of the Arabic Language. It is a cursive language, written from right to left, each character has two to four different forms according to its position in the word, and most characters are associated with complementary parts above, below, or inside the character. However, the interest and research in this area is becoming more intensive than before.

SVMs is a powerful tool compared to other supervised classification techniques. It is based on statistical learning theory developed by Vladimir Naumovich Vapnik [1] back in 1963 and since then, his original ideas have been perfected by a series of new techniques and algorithms including that of Olusayo D. Fenwa et al. [2], who evaluated the Performance Of PSO-Based Kernel Support Vector Machine in Offline

Digit Recognition. One of the most important design choices for SVMs is the kernel-parameter, which implicitly defines the structure of the high dimensional feature space where a maximal margin hyperplane will be found. However, before this stage is reached in the use of SVMs, the actual kernel must be chosen, as different kernels may exhibit different performance.

This paper investigates the effect of eleven different kernels on the performance of SVMs in recognizing Arabic characters. The objective is to extract the kernels giving the best performance. Future work can then be elaborated for enhancing the performance of these kernels. The rest of the paper is organized as follows. Section II reviews the literature. Section III describes the used algorithm. The analysis and discussion of the obtained results are summarized in Section IV. Conclusions of this work with some future lines of research are presented in the last section.

II. LITERATURE SURVEY

Support Vector Machine (SVM) is a state-of-the-art classification method that belongs to the general category of kernel methods. A kernel method is an algorithm that depends on the data only through dot-products. When this is the case, the dot product can be replaced by a kernel function which computes a dot product in some possibly high-dimensional feature space. This approach has two advantages: First, the ability to generate nonlinear decision boundaries using methods designed for linear classifiers. Second, the use of kernel functions allows the user to apply a classifier to data that have no obvious fixed-dimensional vector space representation.

In recent years, Kernel methods have received major attention, particularly due to the increased popularity of the Support Vector Machines. Kernel functions can be used in many applications as they provide a simple bridge from linearity to non-linearity for algorithms which can be expressed in terms of dot products [3].

A linear support vector machine is composed of a set of given support vectors \mathbf{y} and a set of weights \mathbf{w} . The computation for the output of a given SVM with N support vectors y_1, y_2, \dots, y_N and weights w_1, w_2, \dots, w_N is then given by:

$$F(x) = \sum_{i=1}^N w_i(y_i, x) + b \quad (1)$$

Using kernels, the original formulation for the SVM given SVM with support vectors y_1, y_2, \dots, y_N and weights w_1, w_2, \dots, w_N is now given by:

$$F(x) = \sum_{i=1}^N w_i k(y_i, x) + b \quad (2)$$

This work investigates the effect of eleven different kernels on the performance of SVM for recognizing Arabic characters. The most popular kernels for real-valued vector inputs are shown below [4]:

1) *Linear (trivial) Kernel:*

Linear kernel, the simplest kernel function, is given by the inner product $\langle x, y \rangle$ plus an optional constant c . Algorithms using a linear kernel are often equivalent to their non-kernel counterparts.

$$k(x, y) = x^T y + c \quad (3)$$

2) *Quadratic Kernel:*

$$k(x, y) = (S(x, y) + c)^2 \quad (4)$$

where, c and S are kernel-specific parameters

3) *Rational Quadratic Kernel*

$$k(x, y) = 1 - \frac{\|x-y\|^2}{\|x-y\|^2 + c} \quad (5)$$

4) *Multiquadric Kernel*

The Multiquadric kernel can be used in the same situations as the Rational Quadratic kernel.

$$k(x, y) = \sqrt{\|x-y\|^2 + c^2} \quad (6)$$

5) *Inverse Multiquadric Kernel*

$$k(x, y) = \frac{1}{\sqrt{\|x-y\|^2 + c^2}} \quad (7)$$

6) *Polynomial Kernel*

The Polynomial kernel is a non-stationary kernel that is well suited for problems where all the training data is normalized.

$$k(x, y) = (\alpha x^T y + c)^d \quad (8)$$

Adjustable parameters are the slope α , the constant term c and the polynomial degree d .

7) *Gaussian Kernel*

The Gaussian kernel is an example of radial basis function kernel.

$$k(x, y) = \exp\left(-\frac{\|x-y\|^2}{2\sigma^2}\right) \quad (9)$$

The adjustable parameter σ plays a major role in the performance of the kernel, and should be carefully tuned to the problem at hand.

8) *Exponential Kernel*

The exponential kernel is closely related to the Gaussian kernel, with only the square of the norm left out. It is also a radial basis function kernel.

$$k(x, y) = \exp\left(-\frac{\|x-y\|}{2\sigma^2}\right) \quad (10)$$

9) *Laplacian Kernel*

The Laplace Kernel is completely equivalent to the exponential kernel, except for being less sensitive for changes in the sigma parameter. Being equivalent, it is also a radial basis function kernel.

$$k(x, y) = \exp\left(-\frac{\|x-y\|}{\sigma}\right) \quad (11)$$

It is important to note that the observations made about the sigma parameter for the Gaussian kernel also apply to the Exponential and Laplacian kernels.

10) *Hyperbolic Tangent (Sigmoid) Kernel*

The Hyperbolic Tangent Kernel is also known as the Sigmoid Kernel and as the Multilayer Perceptron (MLP) kernel.

$$k(x, y) = \tanh(\alpha x^T y + c) \quad (12)$$

It is interesting to note that a SVM model using a sigmoid kernel function is equivalent to a two-layer perceptron neural network. This kernel was quite popular for support vector machines due to its origin from neural network theory. There are two adjustable parameters in the sigmoid kernel, the slope **alpha** and the intercept constant c . A common value for alpha is $1/N$, where N is the data dimension.

11) *Multi-Layer Perceptron:*

The long established MLP, with a single hidden layer, also has a valid kernel representation.

$$k(x, y) = \tanh((x, y) + c) \quad (13)$$

Many authors tried the investigation of using SVMs and similar tools for recognizing Arabic characters and categorizing Arabic text. Mahmoud Zennaki et al., in [5], presented a comparative study of SVM models for learning handwritten Arabic Characters. Eugen-Dumitru Tautu and Flrin Leon, [6] examined the effect of 4 kernels on the performance of SVM for recognizing English handwritten characters. The examined kernels are: linear, RBF, polynomial, and sigmoid functions. They found that the type of the kernel function affects the recognition accuracy. Behjat Siddiquie et al, [7] tried combining multiple Kernels for efficient image classification. S.F. Bahgat et al, [8] proposed a Hybrid Technique for Recognizing Arabic Characters.

III. PROPOSED APPROACH

The process starts with creating the database of the Arabic alphabetic character images used throughout the study. The database contains the character images, the feature vectors of noise-free images, as well as the feature vectors of character images corrupted by additional salt-and-pepper noise with levels ranging from 0.1 to 0.9. The SVM is first trained using the noise-free feature vectors for each used kernel. The SVM is then used for the classification of noisy character set for each kernel. A comparative analysis is then carried out to conclude which kernels are suitable and which are not. A detailed description of the used algorithm is as shown below.

Algorithm for determining the effect of different kernels on the performance of SVMs for recognizing Arabic characters:

INPUT: Noise free database for the Arabic 28 character images.

OUTPUT: Recognition rate of Arabic characters using 11 different kernels.

1. Initialize: clear all data from the workspace.
2. Calculate the statistical features of the noise free characters and train the SVM.
for $l=1:11$ (No. of kernels)
for $i=1:28$ (No. of characters)
for $j = 1:100$ (repeat 100 times)
Generate 100 statistical feature vector matrices
end for
Train the SVM for all characters using this kernel
end for
Repeat training for each kernel
end for
3. Calculate noisy statistical feature vectors for all characters.
for $k=1:9$ (Salt & pepper level noise from 0.1 – 0.9)
for $i=28$ (No. of characters)
Generate noisy database.
Calculate noisy statistical features for all characters.
end for
Repeat for all noise levels.
end for
4. Calculate the performance of SVMs for recognizing Arabic characters for each kernel.
for $l=1:11$ (no. of kernels).
for $m=1:10$ (Take the average of 10 repetitions).
for $k = 1:9$
SVM classification for different noise levels
Store results.
end for
Repeat 10 times.
end for
Calculate the average recognition rate for each kernel.
end for
5. Exit.

IV. RESULTS ANALYSIS AND DISCUSSION

This section investigates the performance of SVMs in recognizing the Arabic characters, corrupted by salt and pepper noise with levels starting from 0.1 to 0.9, using eleven different kernels. Samples of the free and noisy Arabic character images used in this study are shown in Fig. 1.

As shown in TABLE I and Fig. 2., it is clear that Laplacian, Exponential, Gaussian, Inverse Multiquadric, and Rational Quadratic Kernels give excellent results for all noise levels. On the other hand, the multi-quadric kernel gives very bad results for all noise levels. Linear, Quadratic, Polynomial, Hyperbolic Tangent, and Multilayer Perceptron Kernels, have monotonically decreasing performance with increasing the noise level.

Excluding the Multiquadric kernel, TABLE II and Fig. 3. show that there are two groups of kernels according to their performance. Group 1, which has the best performance for all noise levels, includes Laplacian, Exponential, Gaussian, Inverse multiquadric, and Rational Quadratic Kernels. Group 2, which has a monotonically decreasing performance with increasing noise levels, includes Linear, Quadratic, Polynomial, Hyperbolic Tangent, and Multilayer Perceptron Kernels.

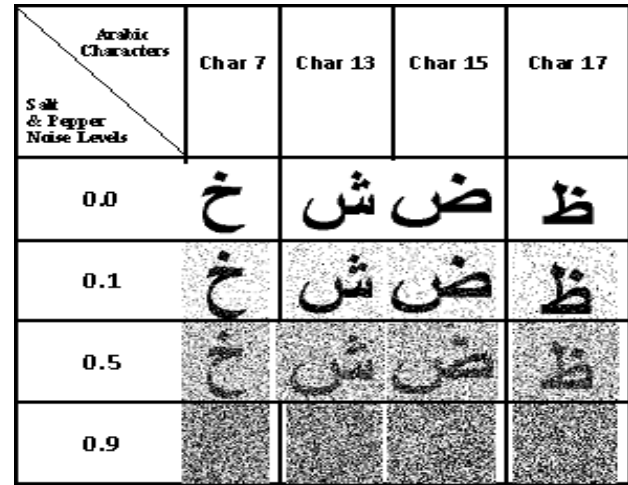


Fig. 1. Samples of the used database

TABLE I. RECOGNITION RATE OF DIFFERENT KERNELS (%) IN THE PRESENCE OF SALT AND PEPPER NOISE

Salt & Pepper Noise Level	Linear Kernel	Quadratic Kernel	Rational Quadratic Kernel	Multiquadric Kernel	Inverse Quadratic Kernel	Polynomial Kernel	Gaussian Kernel	Exponential Kernel	Laplacian Kernel	Hyperbolic Tangent Kernel	Multilayer Perceptron Kernel
0.1	83.7	90.6	96.5	9.17	96.4	84.8	96.4	96.4	96.4	83.1	87.7
0.2	74.7	86.9	96.3	9.20	96.1	74.6	96.4	96.4	96.4	76.1	82.7
0.3	68.8	84.3	96.0	8.55	95.9	66.7	96.2	96.4	96.4	73.4	76.6
0.4	65.9	81.7	95.3	8.23	95.3	60.8	96.0	96.4	96.1	73.2	67.2
0.5	62.7	79.6	95.1	8.17	95.1	55.9	95.6	96.4	96.1	73.0	67.2
0.6	59.8	76.2	94.8	7.76	94.8	50.8	95.2	96.1	96.0	72.7	65.9
0.7	58.1	72.8	94.5	7.69	94.5	46.5	94.9	96.0	96.0	72.3	64.8
0.8	57.3	70.6	94.4	7.16	94.3	44.1	94.6	95.9	95.9	71.3	63.4
0.9	56.5	70.0	94.2	6.98	94.2	44.1	94.4	95.8	95.8	71.2	60.9

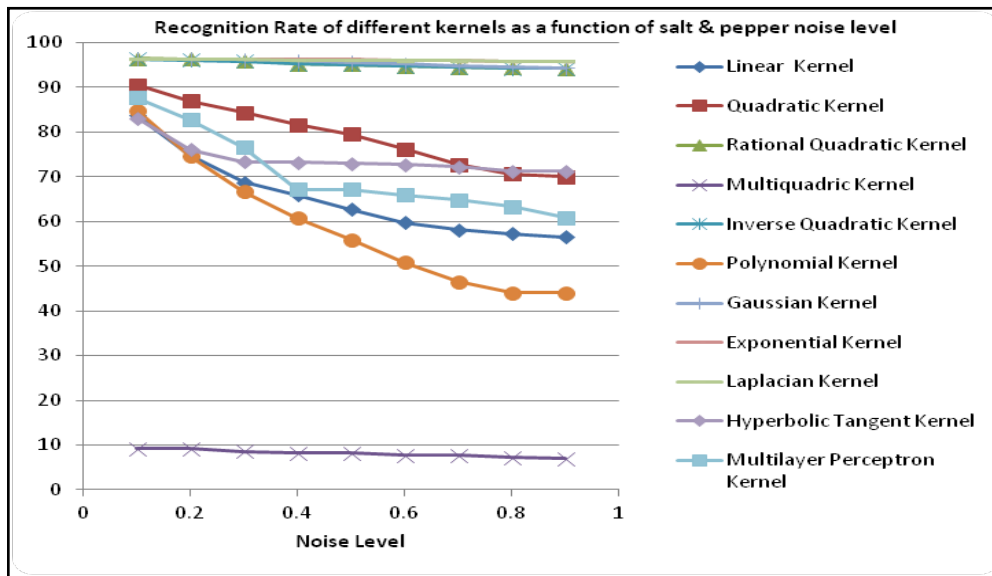


Fig. 2. Recognition rate of Arabic characters using SVM with different kernels as a function of salt & pepper noise

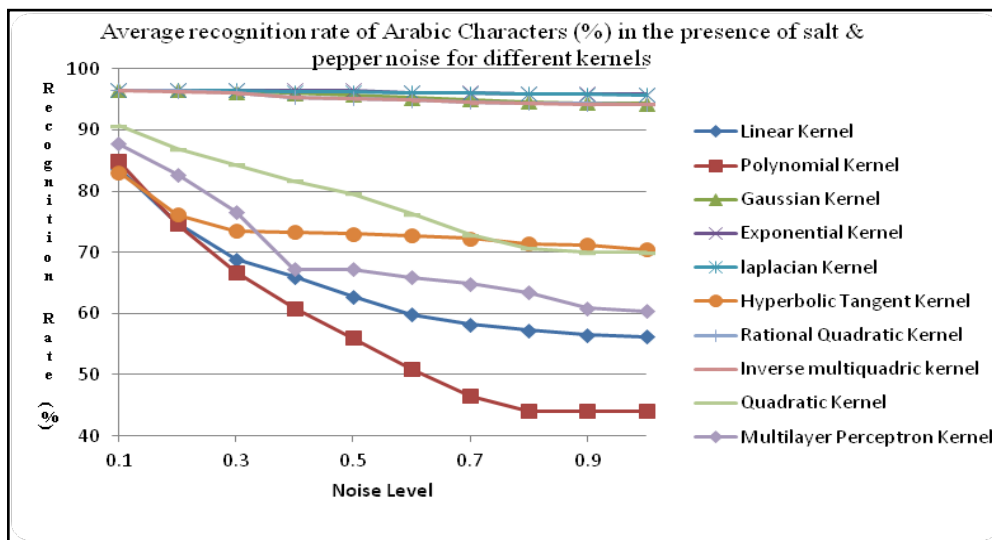


Fig. 3. Recognition rate of Arabic characters after excluding the multi-quadric kernel

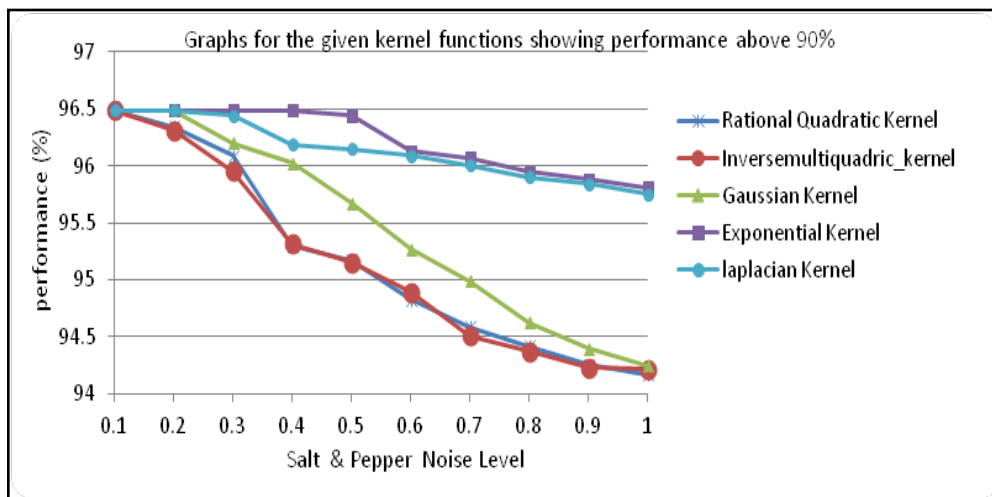


Fig. 4. Performance comparison for the five best kernels

TABLE II. RECOGNITION RATE OF ARABIC CHARACTERS AFTER EXCLUDING THE MULTI-QUADRIC KERNEL

Salt & Pepper noise level	Linear Kernel	Polynomial Kernel	Gaussian Kernel	Exponential Kernel	laplacian Kernel	Hyperbolic Tangent Kernel	Rational Quadratic Kernel	Inverse multiquadric kernel	Quadratic Kernel	Multilayer Perceptron Kernel
0.1	83.7	84.9	96.5	96.5	96.5	83.1	96.5	96.5	90.7	87.7
0.2	74.7	74.7	96.5	96.5	96.5	76.2	96.3	96.3	86.9	82.7
0.3	68.8	66.8	96.2	96.5	96.4	73.5	96.1	96.0	84.3	76.6
0.4	66.0	60.9	96.0	96.5	96.2	73.3	95.3	95.3	81.7	67.3
0.5	62.8	56.0	95.7	96.4	96.2	73.0	95.2	95.2	79.6	67.2
0.6	59.9	50.9	95.3	96.1	96.1	72.8	94.8	94.9	76.2	65.9
0.7	58.2	46.6	95.0	96.1	96.0	72.3	94.6	94.5	72.9	64.9
0.8	57.3	44.1	94.6	96.0	95.9	71.4	94.4	94.4	70.6	63.5
0.9	56.5	44.1	94.4	95.9	95.8	71.3	94.3	94.2	70.0	60.9

Focusing on the group having the best performance, TABLE III and Fig. 4., show that the exponential and Laplacian kernels give the best performance in the set of the examined kernels. However, there are another 14 kernels that will be examined to extract the final result.

V. CONCLUSION

SVMs are used as a classification tool for the recognition of Arabic characters. However, before this stage is reached in the use of SVMs, the actual kernel must be chosen, as different kernels may exhibit different performance. This paper studies the effect of eleven different kernel functions on the performance of SVMs for recognizing Arabic characters.

The obtained results show that Laplacian, Exponential, Gaussian, Inverse Multiquadric, and Rational Quadratic Kernels give excellent results for all noise levels. On the other hand, the multi-quadric kernel gives very bad results for all noise levels. Linear, Quadratic, Polynomial, Hyperbolic Tangent, and Multilayer Perceptron Kernels, have monotonically decreasing performance with increasing the noise level. Further investigation showed that Exponential and Laplacian kernels give the best performance. Future work will stress on another set of kernels to extract the most suitable kernel functions for recognizing Arabic characters.

TABLE III. KERNEL FUNCTIONS SHOWING PERFORMANCE ABOVE 90%

Salt & Pepper noise value	Gaussian Kernel	Exponential Kernel	laplacian Kernel	Rational Quadratic Kernel	Inverse multiquadric kernel
0.1	96.5	96.5	96.5	96.5	96.5
0.2	96.5	96.5	96.5	96.3	96.3
0.3	96.2	96.5	96.4	96.1	95.9
0.4	96.0	96.5	96.2	95.3	95.3
0.5	95.6	96.4	96.2	95.2	95.1
0.6	95.3	96.1	96.1	94.8	94.9
0.7	94.9	96.1	96.0	94.6	94.5
0.8	94.6	95.9	95.9	94.4	94.4
0.9	94.4	95.8	95.8	94.3	94.2

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Cosine Based Latent Factor Model for Precision Oriented Recommendation

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Abstract—Recommender systems suggest a list of interesting items to users based on their prior purchase or browsing behaviour on e-commerce platforms. The continuing research in recommender systems have primarily focused on developing algorithms for rating prediction task. However, most e-commerce platforms provide ‘top-k’ list of interesting items for every user. In line with this idea, the paper proposes a novel machine learning algorithm to predict a list of ‘top-k’ items by optimizing the latent factors of users and items with the mapped scores from ratings. The basic idea is to learn latent factors based on the cosine similarity between the users and items latent features which is then used to predict the scores for unseen items for every user. Comprehensive empirical evaluations on publicly available benchmark datasets reveal that the proposed model outperforms the state-of-the-art algorithms in recommending good items to a user.

Keywords—collaborative filtering; recommender systems; precision; e-commerce; machine learning

I. INTRODUCTION

In the age of ‘internet of things’, there is a growing importance of personalized recommender systems (RS). RS are typical software solutions used in E-commerce for personalized services [1]. It helps customers to find interesting items by providing recommendations based on their prior preferences viz., amazon.com suggests a list of items based on the purchase history of a user. It also benefits E-commerce portals that offer millions of products for sale by targeting the right customer for the right product [2]. Due to growing significance of RS, several techniques for developing the recommendation systems have been studied. These include content-based filtering (CBF), collaborative filtering (CF), and hybrid based recommender system. Among them, the CF technique has been widely used due to its simplicity and effectiveness, and has also proven to be useful in many practices [3].

Recommender Systems (RS) collect information on the preferences of its users for a set of items. The information can be collected explicitly and/or implicitly. RS may also use demographics of users such as age, location, gender. There is a growing trend of utilizing social information like followers, followed, tweets for personalized recommendation [4].

The fundamental assumption of CF is that “if users X and Y rate n items similarly, or have similar behaviors (e.g.,

buying, watching, listening), and hence will rate or act on other items similarly” [5]. CF techniques use a database of preference for items (e.g., movies, songs, books, travel destination) by users to predict additional items that may be of interest to users. In a typical CF database, there is a list of users (say m users) and items (say n items) where each user either explicitly (typically, by extracting users’ preferences in form of star rating) or implicitly (typically, by monitoring purchase history, browsing history or even mouse clicks) indicate their preferences corresponding to items [6]. Since, every user cannot look into every item when there are millions of items in E-commerce setup therefore; the preferences are also not available for most of user-item pair. In order to generate recommendation list, an active user can be recommended items with help of other users who have indicated similar preferences for items in CF database.

CF is classified into two types: memory based CF and model based CF. Memory based CF are those which generates recommendation list based on similarity measures either between user-user or between item-item. The similarity measures generally employed are cosine similarity or Pearson correlation similarities which are quite effective. Model based CF learns parameters of models using data mining, machine learning algorithm on training data. The learned parameters are used to predict real data. Latent factor models, Bayesian networks, latent dirichlet allocation(LDA) and Markov decision process based models are frequently researched models in model based CF[5].

Latent factor models, such as Singular value decomposition (SVD), has been quite popular in research of RS as it has been regarded to be the best single method in improving the accuracy in Netflix prize[7]. SVD transforms both items and users to the same latent factor space, thus making them directly comparable. The SVD model as used in Netflix prize learns item bias and user bias which are independent of the features being used for characterisation for items and users.

The previous works in RS literature have focussed more to solve rating prediction task; however the prime objective of RS is to present ‘top-k’ good items in the recommendation lists for every user. Therefore, this work focuses on ‘top-k’ recommendation instead of rating prediction. This means that formulation of the problem has to be transformed as

classification problem, where the task is to classify the good and the bad items. Based on the above arguments, this paper has proposed an innovative algorithm which is a fusion of similarity concepts and latent factor model. The latent factors of user and items are learnt based on the degree of similarity between user and item. The assumption of the proposed model is, more the similarity between user and items, the probability of liking the item by the user would be higher and vice-versa. Experiments for validating the effectiveness of our approach were conducted using benchmark datasets in RS.

The rest of this paper is organized as follows. Section 2 reviews previous studies regarding latent factor models implemented in recommendation systems. Section 3 describes the problem at hand in formal manner. Section 4 describes the proposed model with pseudo code; next section explains how our approach makes a difference based on the results from experiments and also describes the implications of the experiments. Lastly, conclusion is drawn based on observations.

II. RELATED WORK

Since the user-item matrix is often sparse due to unavailability of feedback from most of the users for most of the items, it is often difficult to incorporate memory based CF techniques in RS successfully.

One of the approaches to deal with the problem of sparseness is by adapting model based approach in CF. SVD is used in model based CF which reduces dimensionality of user-item matrix and identifies latent factor in the data [8]. An application of SVD in the context of information retrieval has already been patented and is named as Latent Semantic Indexing (LSI).

Some of the early works in RS by applying SVD has been adopted with appropriate modifications which are different from applications in information retrieval. Daniel Billsus and Michael J. Pazzani in their paper [9] described CF algorithm as classification problem. At first sparse user item matrix is first converted to Boolean feature matrix for every user based on items rated by the user. Subsequent to it the Boolean feature matrix is decomposed using SVD by taking 'k' number of dimensions to be retained. Neural network is used to train the singular vectors and thereafter for prediction [9]. Since the method described is a bit complex and not scalable for real time recommendations, Sarwar, Karypis, Konstan, & Riedl in their paper describe about the methodology of SVD which is directly applied in RS [10]. The user-item matrix which is sparse has to be filled up by user average rating or item average rating. After this pre-processing step SVD is applied on the resultant filled matrix. SVD decomposes the matrix into two matrices; two of which are orthogonal matrices and one is diagonal matrix or singular matrix.

The user-item matrix has to be imputed (assigned value) at the first step before proceeding to SVD which has been criticized by researchers in the field, as imputations led to over-generalization and accuracy of the method is lost. However, the start of SVD was remarkable in the context of recommender system and it solved the problem of sparsity to an extent [11] but there arises a different set of problem as it

happens in very large data set, which often is the case of real world, the complexity and computation of user-item matrix increases exponentially with increasing user item dataset. There is also a need to update the recommendation real time in order to have the most accurate recommendation. In order to address the complexity and computation time problem of SVD can be solved by following a technique proposed by author known as folding-in in SVD [11].

However, it was only until the Netflix prize (Netflix, 2006) that the SVD approach was accepted to be the best single method in RS. Simon Funk popularized the regularized SVD method for the first time to explore the Netflix prize data in order to make accurate prediction [12]. Subsequent to this, modification to the basic regularized SVD was proposed for the Netflix prize dataset. The top prize winner in Netflix prize [13], stressed on augmenting the basic SVD with popular neighborhood based technique. The author suggested incorporating implicit feedback as well as explicit feedback in the same model for the best prediction which was being evaluated on RMSE [6].

Singular value decomposition as a method has also been incorporated along with other available feature of dataset to accurately predict the ratings in case of movie recommender system. SVD combined with demographic data is also proposed to improve the approach of collaborative filtering. The reason of using demographic data along with SVD is to supplement the collaborative filtering algorithm [14].

While deterministic latent factor models such as SVD have been successfully implemented and made popular, probabilistic latent factor models also were considered in information retrieval and subsequently in RS. Thomas Hofmann in his paper utilized the statistical base as a primary reason of using probabilistic latent semantic analysis (pLSA) [15].

Since pLSA has a drawback that exact estimating the ratings is intractable, which means that potentially slow or inaccurate approximations are required for computing the posterior distribution over hidden factors in such model [16]. Full Bayesian analysis of model was done later in 2008 by same authors and called it as probabilistic matrix factorization (pmf) to overcome the problem of inaccurate prediction. The model can be viewed as probabilistic extension of SVD. Using Markov Chain Monte Carlo (MCMC), pmf training is also done to avoid tuning of parameters manually which is required to avoid over-fitting [17].

A relatively similar approach to pLSA is Latent Dirichlet Allocation (LDA). Latent Dirichlet Allocation (LDA) is similar to pLSA in the sense that latent variables are present in a probabilistic way. While pLSA does not assume a specific prior distribution over number of dimensions in hidden variable, LDA assumes that priors have the form of the Dirichlet distribution [18]. Gibbs sampling is used to estimate the parameters in LDA model [19]. The Expectation Maximization (EM) algorithm and its variation can also be used in solving the parameters of the model.

Continuing with matrix factorization method to discover latent factor models, there are other approaches as well which

have been used in the field of RS. One more way of utilizing the matrix factorization so that sparse data can be handled more effectively is a model named Eigentaste that uses principal component analysis for optimal dimensionality reduction and then clusters users in the lower dimensional subspace. As these are model based collaborative filters, they are operated in two modes; online and offline mode. The online mode uses Eigen vectors to project new users into clusters and a lookup table to recommend appropriate items so that run time is independent of the number of users in the database [20].

Matrix factorization is not the only way to handle latent factor models. Discrete Wavelet Transform (DWT) has been used for data reduction without deterioration in signal processing and image processing earlier. Influenced by the technique in handling sparse data DWT has also been used in RS. The technique illustrated is the unique way applied in data reduction in RS to best of our knowledge. The argument presented by author based on previous research illustrates that PCA and SVD find feature combinations that model the largest contributions in a dataset, but these may not be the same features that differentiate attributes, as weaker relationships may be lost [21].

Restricted Boltzmann Machine (RBM) has also been used in order to solve the sparse and large data set such as that of Netflix [22]. RBM introduced for learning the Netflix dataset used a class of two-layer undirected graphical models, suitable for modeling tabular or count data, and presented efficient learning and inference procedures for this class of models.

This sums up the related work using latent factor models in RS. Since, the previous models based on latent factors are guided by the loss function that optimizes the actual ratings; they couldn't quite assist in 'top-k' prediction task. In order to build a model that can handle the prediction of 'top-k' good items to every user, this paper proposes a loss function based on cosine similarity between user and item latent feature. In the next section we will describe the problem at hand in formal and detailed manner.

III. PROBLEM SETTING

In a typical E-commerce setup, there are millions of users and thousands of products listed in database. The user specifically searches for products which he is willing to purchase; with each transaction of a user we can build his purchase history and behavior so forth. The building of a user's preference based on purchase history is termed as implicit feedback. Also, a user may show his explicit preference for a product by providing ratings; viz. 1 to 5 stars. The building of explicit preferences for a user-item pair is termed as explicit feedback. Based on the feedbacks user-item matrix is obtained, consisting of rows representing the users, columns representing the items and elements of matrix are ratings of user for an item.

Practically, not all users may show their preferences for all the items either implicitly or explicitly, which gives rise to sparsity in user-item matrix. This poses a challenge in the recommendation task. In order to model such practical scenarios in research we have tested our models on

MovieLens (ml100k) data set and FilmTrust dataset. The dataset is publicly available for research and has been used in many research papers dealing with recommender system [4]. The proposed model first learns the latent features of users and items using cosine similarity as loss function, and later score of the unseen items for a user is generated. Top 'k' items based on predicted scores can be recommended to a user in descending order of the predicted score.

A. Notations

For distinguishing users from items special indexing letters have been used for user and items – a user is denoted by "i", and an item is denoted by "j". A rating r_{ij} indicates the preference of a user i for item j, where high values mean stronger preference and low values mean low preference or no preference for an item i. For example in a range of "1 star" to "5 stars", "1 star" rating means lower interest by a particular user u for a given item i and "5 stars" rating means high interest by user u for a given item i. A mapped score s is obtained from the ratings by passing through a suitable function is described in next section. The parameters \mathbf{P}_i and \mathbf{Q}_j denotes the user and item features respectively and are in form of a vector. In this paper the 'bold' notations denotes the vector and the corresponding elements of the vectors are 'normal'.

IV. PROPOSED MODEL

This section will cover the model building phase of the innovative algorithm for classification of good and bad items. The algorithm is primarily build for classification task but we can also extend this to rating prediction task.

A. Cosine based latent factor model

There are a few disadvantages of using matrix factorization to learn the latent factors of users and items. One of the disadvantages is that the function is not bounded; hence there is a possibility of obtaining the predicted values out of range [16]. Since the predicted values may get out of range the predicted values are clipped [12] or passed through a bounded function such as logistic function [16]. This may not be appropriated since the mapping function of actual rating in training set of data is not mapped according to the bounded function and are generally normalized [16].

To furnish a mathematical solution to this problem this work introduces a cosine based latent factor model. The cosine function is bounded between -1 and 1 which gives an advantage to map the actual ratings in train set using cosine function and use cosine latent factor model to learn the features of users and items.

The intuition behind use of pervious latent factor models such as regularized SVD states that interaction between user and item features results in ratings of an item by a user [7]. In the proposed cosine latent factor model the intuition is the degree of similarity between user and item features defines the interest of user for an item. So if a user is highly interested in an item the similarity between the user and item features is close to 1 otherwise, the similarity is close to 0. In order to map the actual ratings $R_{ij} \in \{1 \dots r\}$ in between 0 and 1 we passed it using a function ϕ . This function has to be defined

such that the minimum rating shall be close to 0 and maximum rating shall be close to 1. One such function is defined below:

$$\emptyset = \frac{e^{(r-r_{\text{mean}})}}{e^{-(r-r_{\text{mean}})} + e^{(r-r_{\text{mean}})}}$$

where, r_{mean} is the average of $\{1 \dots r\}$. The ratings are passed through this function to obtain a mapped score s . For obtaining the latent factors of each item and user the proposed cosine latent factor model is set equivalent to obtained mapped scores (s). This leads to minimizing the following objective function.

$$f = \min_{P_i, Q_j} \sum_{(ij \in \kappa)} \left(s_{ij} - \frac{P_i Q_j^T}{\sqrt{P_i^2} \sqrt{Q_j^2}} \right)^2 + \lambda (\|P_i\|_{\text{Fro}}^2 + \|Q_j\|_{\text{Fro}}^2) \quad (1)$$

Here, $\|\cdot\|_{\text{Fro}}$ denotes the Frobenius norm.

Regularization parameter λ is introduced to make a balance between over-fitting and variance. The optimum value of the minimization function can be obtained by using stochastic gradient descent method. For every iteration, learning rate (α) is multiplied against the slope of descent of the function in order to reach local minima. The partial derivatives with respect to P_i and Q_j results in gradient of descent for this function.

$$\frac{\partial f}{\partial P_i} = -2 \left(s_{ij} - \frac{P_i Q_j^T}{\sqrt{P_i^2} \sqrt{Q_j^2}} \right) \left(\frac{\sqrt{P_i^2} Q_j^T - \left(\frac{P_i Q_j^T \sqrt{Q_j^2} P_i}{P_i^2 Q_j^2} \right)}{P_i^2 Q_j^2} \right) + 2\lambda P_i \quad (2)$$

$$\frac{\partial f}{\partial Q_j} = -2 \left(s_{ij} - \frac{P_i Q_j^T}{\sqrt{P_i^2} \sqrt{Q_j^2}} \right) \left(\frac{\sqrt{P_i^2} Q_j^2 P_i - \left(\frac{P_i Q_j^T \sqrt{P_i^2} Q_j^T}{\sqrt{Q_j^2}} \right)}{P_i^2 Q_j^2} \right) + 2\lambda Q_j \quad (3)$$

1) Model learning

Algorithm:

Input:

- R : A matrix of rating, dimension $N \times M$ (user item rating matrix)
- κ : Set of known ratings in matrix R
- P_i : An initial vector of dimension $N \times F$ (User feature vector)
- Q_j : An initial vector of dimension $M \times F$ (item feature matrix)
- F : Number of latent features to be trained
- s_{ij} : mapped scores obtained after passing through function \emptyset

Parameters:

- α_1 : learning rate

- λ : over fitting regularization parameter
- Steps : Number of iterations

Output: A matrix with scores to generate recommendation list

Method:

- 1) Initialize random values to vector Q_j, P_i
 - 2) Fix value of F, α_1 and λ .
 - 3) do till error converges [error(step-1) - error(step) < ϵ]
- error (step) =

$$\min_{P_i, Q_j} \sum_{(ij \in \kappa)} \left(s_{ij} - \frac{P_i Q_j^T}{\sqrt{P_i^2} \sqrt{Q_j^2}} \right)^2 + \lambda (\|P_i\|_{\text{Fro}}^2 + \|Q_j\|_{\text{Fro}}^2)$$

for each $R \in \kappa$

Update training parameters

$$P_i \longleftarrow P_i + \alpha_1 \left(\frac{\partial f}{\partial P_i} \right)$$

$$Q_j \longleftarrow Q_j + \alpha_1 \left(\frac{\partial f}{\partial Q_j} \right)$$

end for

- 1) Return P_i, Q_j

The obtained P_i, Q_j for each user and items are used to predict a score using the following equation.

$$\hat{s}_{ij} = \frac{P_i Q_j^T}{\sqrt{P_i^2} \sqrt{Q_j^2}} \quad (4)$$

Based on these, the scores are arranged in descending order and top 'k' items for each user can be generated in the recommendation list.

Further, to extend this model for rating prediction task, we will use the calculated similarity score between the user and unrated item, user and all other items rated by active user, based on learned latent features. The top n-nearest neighbours to the unrated items are scanned based on calculated similarity score and their average is used to predict the rating for the unrated item.

V. EXPERIMENTATION AND EVALUATION

In this section the experimental setup and evaluation protocol to test the proposed model on two publicly available datasets have been presented. The proposed model is compared with baseline and other state-of-art algorithms. The proposed algorithms are evaluated both on classification and rating prediction using appropriate performance measures.

A. Datasets

For the experimental evaluations of the proposed method, two different datasets are used. The first one is a publicly available Movie Lens dataset (ml-100k). The dataset consists of ratings of movies provided by users with corresponding user and movie IDs. There are 943 users and 1682 movies

with 100000 ratings in the dataset. Had every user would have rated every movie total ratings available should have been 1586126 (i.e. 943×1682); however only 100000 ratings are available which means that not every user has rated every movie and dataset is very sparse (93.7%). This dataset resembles an actual scenario in E-commerce, where not every user explicitly or implicitly expresses preferences for every item.

The second dataset consists of movie reviews from FilmTrust [23]. There are 1508 users and 2071 movies with only 35497 ratings. The sparsity levels (98.86%) are more than movieLens dataset.

B. Cross-Validation

The dataset is partitioned into 5 equal disjoint sets with 4 datasets used for training and one left out dataset for testing the model. The process is repeated five times, as a procedure adopted for 5-fold cross-validation. On testing dataset the accuracy measure such as RMSE, and precision is calculated and averaged over the 5-folds which is a procedure adopted to nullify the effect of biasness of partitioning the sample

C. Performance Metrics

In a classification task the performance metrics that determines the top ‘k’ as used in recommendation systems are

1) *Precision*: Precision is defined as the ratio of relevant items, Nrs, recommended to the total number of items, Ns, recommended to a user.

$$\text{Precision} = \text{Nrs} / \text{Ns}$$

In rating prediction task, the goal is to minimize the difference in ratings between predicted and actual ratings. In order to evaluate the accuracy, RMSE (Root Mean Square Error) and MAE (Mean Absolute Error) are popular metrics. The variants of RMSE and MAE such as Normalized RMSE and Normalized MAE or average RMSE and average MAE are also used. The predicted ratings (\hat{r}_{ij}) for a test set ‘ Γ ’ of user-item pairs (i, j) for which true item ratings (r_{ij}) are known, the RMSE is given by

$$\sqrt{\frac{1}{|\Gamma|} \sum_{(i,j) \in \Gamma} (\hat{r}_{ij} - r_{ij})^2}$$

$|\Gamma|$ is number of observations in test set

MAE on the other hand is given by

$$\frac{1}{|\Gamma|} \sum_{(i,j) \in \Gamma} |\hat{r}_{ij} - r_{ij}|$$

D. Evaluating the performance of models

In this section, the performances of the proposed model with already existing state-of-the-art algorithm in this field are evaluated. One of the state-of-the-art algorithms is RSVD that was designed primarily for rating prediction task. For experimentation purposes, the number of latent features (F) is varied from 10 to 100 in steps of 10 for proposed model and RSVD algorithm. Firstly, the focus is on classification task where the idea is to present ‘top-k’ items to each user. Based on the obtained predicted ratings, in case of RSVD, and

obtained scores, in proposed model, top 5 and top 10 items are presented to the user. The predicted rating and predicted scores respectively in descending order are presented to every user and then accuracy measures such as precision are obtained [24].

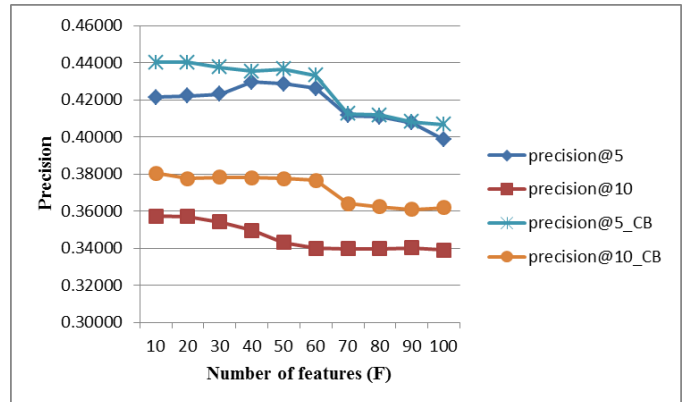


Fig. 1. precision of proposed cosine latent factor model and RSVD on ml-100k dataset

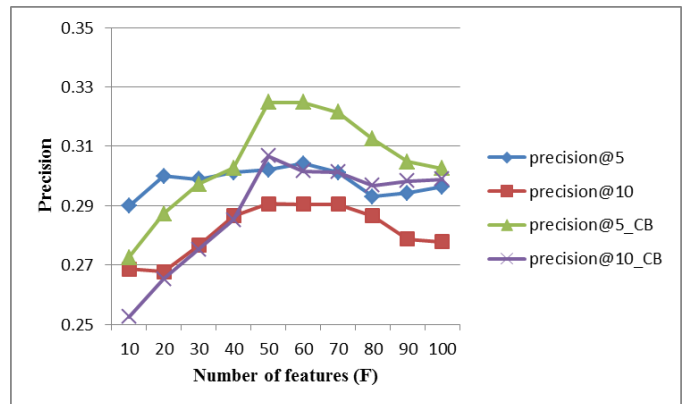


Fig. 2. precision of proposed cosine latent factor model and RSVD on FilmTrust dataset

Figure 1 and figure 2 show the precision of the proposed cosine based latent factor model and RSVD on ml-100k dataset and FilmTrust dataset respectively. Since the precision are computed for top 5 and top 10 items presented to each user Precision@5 and Precision@10 are used to denote in the figure 1 and 2. Precision@5_CB shows the precision as obtained on datasets by applying cosine based latent factor model, while precision@5 shows the precision as obtained by RSVD. In ml-100k dataset, the highest precision for cosine based latent factor model occurs when the number of features (F) for user and item are 10. Correspondingly, the highest value of precision for RSVD occurs for F=10 but the values of precision at both top 5 and top 10 items is better for the proposed cosine latent factor model than state-of-the-art RSVD. There is an improvement of 4.5% for precision@5 and 5% for precision@10 over RSVD algorithm on ml-100k dataset. In case of FilmTrust dataset, the maximum precision@5 for cosine based latent factor model and RSVD occurs when the F value is 60. For precision@10, the maximum value occurs at F=50 for both cosine based latent factor model and RSVD. Here, cosine based latent factor

model outperforms RSVD for precision@5 by approximately 6.5% and for precision@5 by approximately 5%.

In rating prediction task, the predicted ratings obtained from both RSVD and proposed cosine latent factor model with varying latent features (F) are compared with actual ratings in the test set using 5-fold cross-validation. The latent features (F) in both RSVD and proposed cosine latent factor model are varied from 10 to 100 in steps of 10, the cross-validated MAE and RMSE are obtained for both the two datasets and compared.

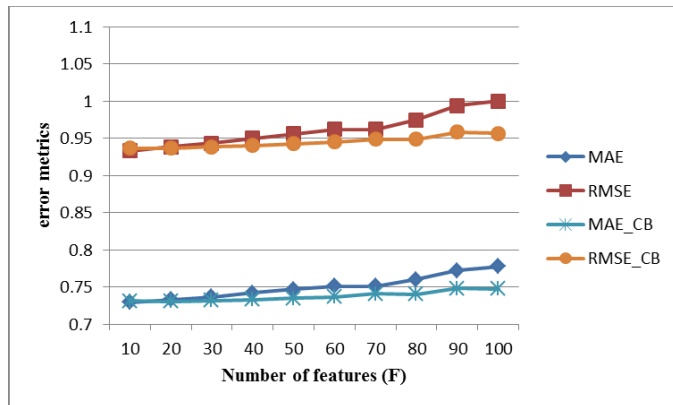


Fig. 3. Error metrics of proposed cosine latent factor model and RSVD on ml-100k dataset

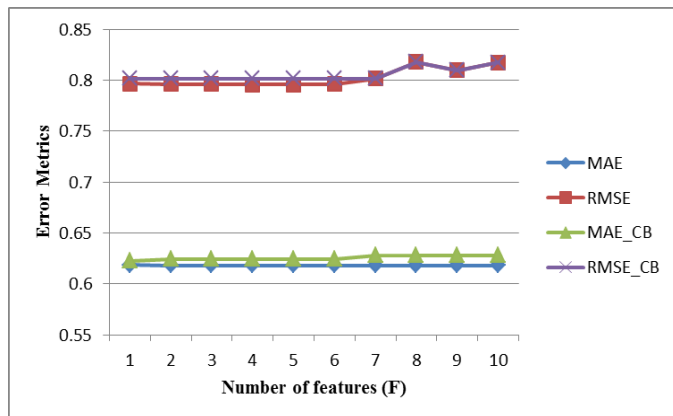


Fig. 4. Error metrics of proposed cosine latent factor model and RSVD on FilmTrust dataset

From the figure 3 and figure 4, one can see that although, MAE and RMSE for RSVD is better than cosine based latent factor model on both the datasets, the difference is negligible for the best value obtained from both these algorithms.

Thus, this work has shown through empirical experimentation that the proposed cosine latent factor model outperforms state-of-the-art algorithm in RS in terms of precision. Also, the proposed model gives comparable results in terms of MAE and RMSE for rating prediction task. In modern e-commerce retail, like amazon, alibaba, the users are presented with a set of recommended products based on their prior purchase and browsing behaviour. Therefore, our work focuses primarily on this aspect of recommending top 'k' items for every user.

VI. DISCUSSIONS AND CONCLUSIONS

In this research work, we have proposed a novel algorithm that caters to recommending top 'k' items for each user. Our work primarily falls in domain of model based RS with focus on classification of good and bad items. This work introduces the concept of cosine similarity based latent factor model which is a unique algorithm in itself. Also, the rationale behind using cosine similarity latent factor model over RSVD is theoretically sound. As mentioned in the paper that the rating prediction using RSVD often goes out of bounds as the loss function is not bounded, the loss function used in the proposed model is bounded and therefore the prediction do not go out of the bounds. One more advantage in using the proposed model is its ability to handle difficult (outlier) data points due to its inherent property of bounds which are not observed in RSVD model.

In future, we look forward to utilize the proposed cosine latent factor model in field of information retrieval and also in ranking prediction task for both recommender system and information retrieval. The learning method of optimization can also be suitably modified to learn the parameters of the proposed model a bit faster. One of the other approaches of using the proposed model for the above task can be by ensemble of weak learners generated by varying the number of latent factors of the model. The present work is primarily designed for top 'k' recommendation task but has also been extended to rating prediction task by using simple average of the ratings of most similar items are applied. The rating prediction using more sophisticated techniques like clustering of the item features can also be obtained to check any improvement. The techniques being applied has to be carefully chosen as they may increase the complexity without improving the accuracy adequately.

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Towards Building an Intelligent Call Routing System

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Abstract—This paper presents EduICR - an Intelligent Call Routing system. This system can route calls to the most appropriate agent using routing rules built by the text classifier. EduICR includes the following main components: telephone communication network; Vietnamese speech recognition; Text classifier/ Natural language processor and Vietnamese speech synthesis. To our best knowledge, this is one of the first systems in Vietnam to implement the integration mechanism of text processing and speech processing. This allows voice applications to be more intelligent, able to communicate with humans in natural language with high accuracy and reasonable speed. Having been built and tested in real environment, our system proves its accuracy attaining more than 95%.

Keywords—EduICR; spoken dialog systems; intelligent call center; voice application

I. INTRODUCTION

In recent years, Vietnamese have been seeing many remarkable publications which displayed by groups devoting to spoken Vietnamese recognition researches from Institute of Information Technology (Vietnamese Academy of Science and Technology) and University of Science, VNU-HCM. It is worth mentioning the works of Thang Vu and Mai Luong [7] as well as Quan Vu et al. [3,5]. These studies crucially concentrated on improving the efficiency of their voice recognition system, such as the Quan Vu et al. 's one which obtained the precision rate of over than 93% and this group successfully built many voice applications on this base. For example, in [5], Quan Vu et., al. successfully built VIS::DIR system, which caller can say the names of departments/offices in a university and the system will forward/redirect/route these calls to the associate agents without any help from receptionist. Nevertheless, all the applications have not been accompanied with an efficient text processing mechanism yet, which is the important mechanism in view of helping the system with understanding commands.

In this study we would like to propose some approaches to build EduICR - an Intelligent Call Routing system. This system can route calls to the most appropriate agent using routing rules built by the text classifier. In this work, we have approached two techniques: SVM (Support Vector Machine) machine learning method to classify the text commands, and DCG (Definite Clause Grammar) rule-based method [1] to deal with

the syntax and semantic analysis of the text commands. Our system also has Automatic Speech Recognition module and Speech Synthesis module. Same to our approach in [8,9,10] we deal with a Vietnamese speech recognition task by using HTK (Hidden Markov Model Toolkit) [6] and speech synthesis operations by using Unit-Selection method [2].

II. SYSTEM ARCHITECTURE

Our system is designed with the following functions: identifying the commands via telephone; classifying the commands; routing and answering the user via telephone. An inquiring session between the system and the user is described in Figure 1.

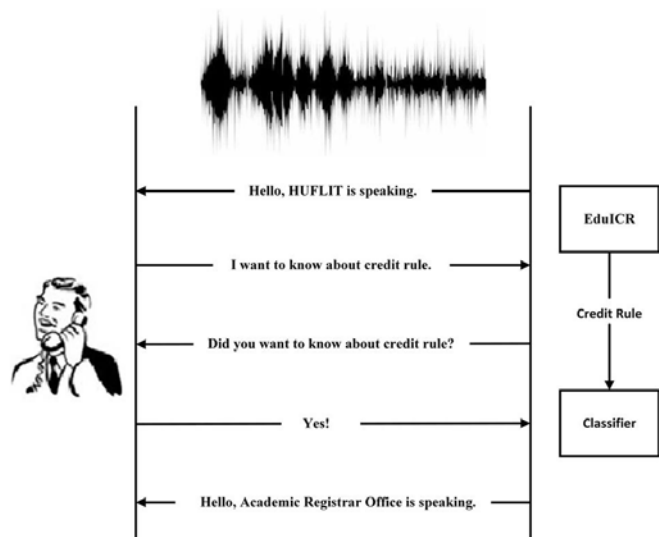


Fig. 1. An inquiring session

In order to realize the functions that given in the scenario in Figure 1, the system must consist of the following components:

- Automatic speech recognizer (ASR): to identify words that the user speaks, then convert them into text.
- Text processor: to classify the text commands to associate with agents.
- Synthesizer: to convert text to speech.

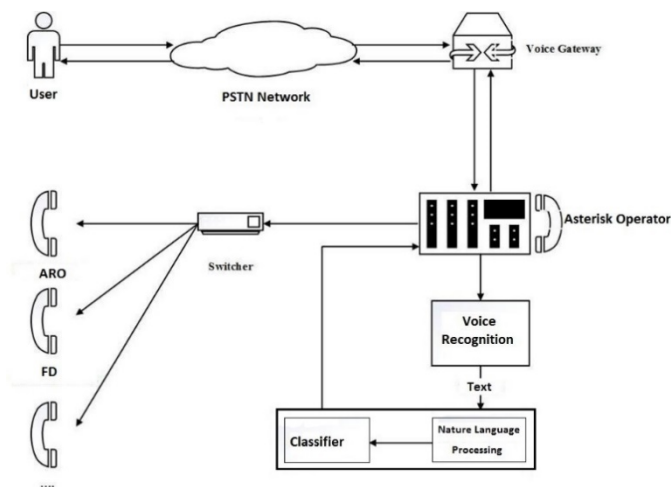


Fig. 2. Architecture of EduICR

III. THE SPEECH RECOGNITION

In EduICR system, we have used HTK [8] to build the Automatic Speech Recognition component. Employing the same approach as in [5,8,9,10] we have applied the context-dependent model based on tri-phone to recognize words. Besides, we have defined the tied rules for its grammar. Figure 3 shows some steps in order to create the Automatic Speech Recognizer.

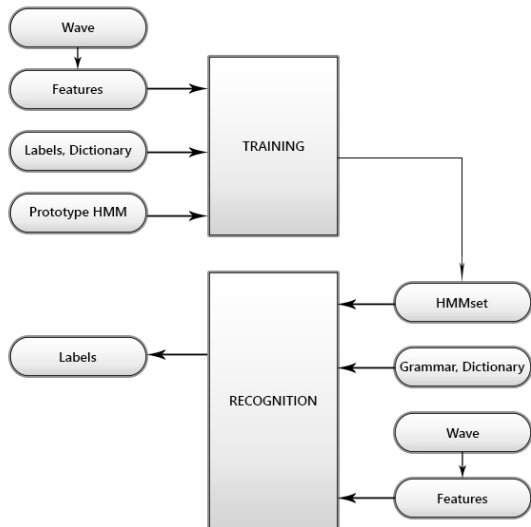


Fig. 3. Steps to build the Automatic Speech Recognizer [8]

A. Training data

There are 3,500 sentences in the speech corpus (3,500 sentences have been taken from users and they have not randomly). Total audio training covers 3 hours. All speech was sampled at 8,000Hz, 16bit by PCM format in a relatively quiet environment with 30 speakers. The lexicon comprises of 153 keywords in 3,500 sentences as being shown in Table I.

TABLE I. LIST OF KEYWORDS

bao	đào	là	nhieu	thông
bảng	đại	làm	như	thời
bảo	đăng	lạc	những	thu
bằng	điều	lại	nộp	thực
biểu	điểm	lệ	nói	thức
biết	định	liên	nợ	tiêu
bình	đôi	lich	nữa	tin
bộ	đục	lơ	ôn	tích
bổng	gấp	luận	ờ	tín
câu	giây	lưu	phân	tính
cao	gian	lũy	phí	tôi
các	giảm	luyện	phòng	tốt
cách	giáo	máy	phúc	trả
chất	giới	miễn	quốc	trẻ
chế	hai	môn	quy	trình
chi	hạn	mở	ra	trung
chi	hết	mùa	rèn	trường
chính	hè	muốn	sách	tuyển
cho	hoàn	mức	sau	và
chuẩn	học	nào	sinh	vào
chương	hỏi	năm	sĩ	vấn
chứng	huỷ	nay	tập	về
chuyên	khảo	ngành	tạo	việc
còn	khi	nghe	thành	viên
công	khoa	nghe	thạc	vong
có	khoản	nghe	thể	với
của	không	ngoài	thể	xanh
cương	khóa	ngoại	theo	xét
cứu	kiện	ngữ	thể	yêu
đầu	kỳ	nguyên	thi	
đầu	ký	nhận	thiệt	

B. Grammar Rules / Constraints

A grammar is a set of constraints defining the phrases that a speech recognition engine can use to match speech input. Moreover, we also can provide the speech recognition engine with the predefined grammar that are included custom grammar that we create.

In addition, HTK provides the grammar definition format and an associated HParse tool that is used to build this network of words automatically. We store the grammar definition in a file called gram.txt. In our application, a part of its grammar is on the following:

```

$timetable = ( LIJCH | THOWFI GIAN | THOWFI KHOSA
BIEERU )
$graduate = ( SAU DDAJI HOJC | THAJC SIX )
$sen23 = $subject2 $target TUYEERN SINH $graduate NAWM
NAY LAF BAO NHIEEU;
$sen25 = $subject2 $timetable OON THI CAO HOJC;
    
```

IV. THE SPEECH SYNTHESIZER

Speech synthesizer is a system that converts free text into the speech. Observably, this is a process that a computer reads out the text for people. The speech synthesis can be performed using Formant synthesis [2] or Unit-selection method [2]. With EduICR, we chose an integrative approach by Unit-selection methods, complying process as summarization in Figure 4.

V. VIETNAMESE TEXT PROCESSING

In our system, there are 32 types of commands, being presented in Table II.

In this report, we approached two techniques: SVM (Support Vector Machine) machine learning method to classify the text commands, and DCG (Definite Clause Grammar) rule-based method to deal with the syntax and semantic analysis of the text commands.

With the Vietnam text processing, our system can easily classify/route a command/call as one of associate agents: Academic Registrar Office (ARO); Student Service Department (SSD); Finance Department (FD); Graduate Education Department (GED) and Faculty of Information Technology (FIT).

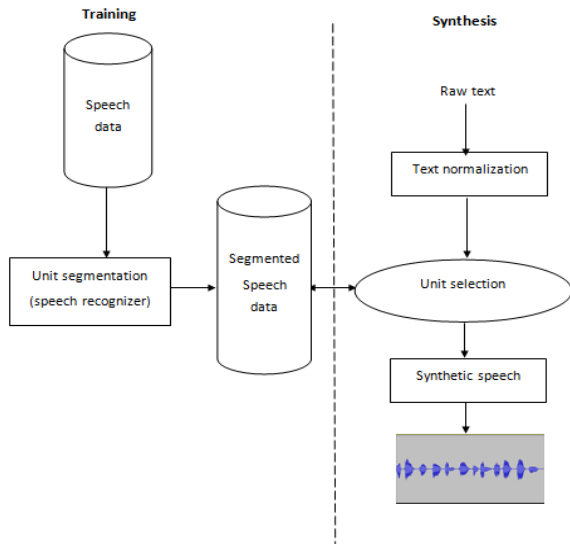


Fig. 4. Speech Synthesizer blueprint [5]

TABLE II. THE COMMAND TYPES

Type	Subject	Sentence Patterns
1	ARO	(Tôi muốn Cho Tôi) (liên lạc nối máy gặp) phòng đào tạo. (I want Please) [to] (contact connect speak) to Academic Registrar Office.
2	ARO	(Tôi muốn Cho Tôi) (biết hỏi) cách (đăng ký huỷ đổi) môn học [như thế nào?] (I want Please) [to] (know ask show me) how to (register cancel change) course.
3	ARO	Tôi muốn nhận (chứng chỉ giáo dục thể chất chứng chỉ giáo dục quốc phòng bằng tốt nghiệp bằng điểm)? I want to get (Certificate of Physical Education Certificate of National Defense Education Graduate Certificate Educational record).
4	ARO	[Tôi muốn Cho Tôi] [biết hỏi] trường (tuyển sinh xét tuyển) những ngành nào? [I want Please] [to] [know ask show me] which faculties does HUFLIT have to be applied for.
5	ARO	[Tôi muốn Cho Tôi] [biết hỏi] (điểm chuẩn chỉ tiêu) [nguyên vọng 2] tuyển sinh đại học <năm><khoa> [là bao nhiêu?] [I want Please] [to] [know ask show me] <faculty> 's second aspiration passing score in <year>.
6	ARO	[Tôi muốn Cho Tôi] [biết hỏi] [cách tính] (điểm điểm rèn luyện điểm trung). [I want Please] [to] [know ask show me] [how to calculate] (Practise score Average score).
7	ARO	(Tôi muốn Cho Tôi) (biết hỏi) điều kiện xét tốt nghiệp. (I want Please) [to] (know ask show me) the Graduate requirements.
8	SSD	(Tôi muốn Cho Tôi) (liên lạc nối máy gặp) phòng tổ chức công tác sinh viên. (I want Please) [to] (contact connect speak) to Student Service Department.
9	SSD	(Tôi muốn Cho Tôi) nhận (thẻ sinh viên giấy chứng nhận sinh viên giấy giới thiệu thực tập) (I want Please) [to] get (Student card Student certificate Practise referral form.)
10	SSD	(Tôi muốn Cho Tôi) (biết hỏi) [về các] [chương trình chính sách] học bổng. (I want Please) to (know ask show me) [about] scholarship [programs policies].
11	SSD	(Tôi muốn Cho Tôi) (biết hỏi) [về các] (chương trình mùa tình nguyện chương trình ngoại khoá). (I want Please) [to] (know ask show me) [about] (Volunteer programs Extracurricular programs).
12	SSD	(Tôi muốn Cho Tôi) (biết hỏi) điều kiện xét sinh viên 5 tốt. (I want Please) [to] (know ask show me) Top Ranking Student requirements.
13	SSD	(Tôi muốn Cho Tôi) (biết hỏi) điều kiện xin miễn giảm học phí. (I want Please) [to] (know ask show me) Tuition fee (discount free) requirements.
14	FD	(Tôi muốn Cho Tôi) (liên lạc nối máy gặp) phòng tài vụ. (I want Please) [to] (contact connect speak) to Finance Department.
15	FD	(Tôi muốn Cho Tôi) (biết hỏi) (cách thời gian) nộp học phí. (I want Please) [to] (know ask show me) (how to when to) pay tuition fee.
16	FD	(Tôi muốn Cho Tôi) (biết hỏi) khi nào hết hạn nộp học phí. (I want Please) [to] (know ask show me) Tuition fee payment's deadline.
17	FD	(Tôi muốn Cho Tôi) (biết hỏi) mức học phí của trường. (I want Please) [to] (know ask show me) Tuition fee.
18	FD	(Tôi muốn Cho Tôi) (biết hỏi) về việc (đóng học phí trễ hoàn trả nợ học phí học phí học kì 3 đóng học phí qua ngân hàng). (I want Please) [to] (know ask show me) about (Late payment Tuition fee refund Third semester tuition fee paying tuition fee through bank)

19	FD	(Tôi muốn Cho Tôi) (biết hỏi) ngoài học phí ra trường có thu các khoản phụ phí nào không. (I want Please) [to] (know ask show me) extra fee of HUFLIT.
20	GED	(Tôi muốn Cho Tôi) (liên lạc nối máy gặp) phòng sau đại học.(I want Please) [to] (contact connect speak) to Graduate Education Department
21	GED	(Tôi muốn Cho Tôi) (biết hỏi) (điểm chuẩn chỉ tiêu) tuyển sinh (sau đại học thạc sĩ) năm nay là bao nhiêu? (I want Please) [to] (know ask show me) passing score of Graduate Education admission this year?
22	GED	(Tôi muốn Cho Tôi) (biết hỏi) những quy định tuyển sinh thạc sĩ. (I want Please) [to] (know ask show me) Master Degree Admission rules.
23	GED	(Tôi muốn Cho Tôi) (biết hỏi) (lịch thời khoá biểu) ôn thi cao học. (I want Please) [to] (know ask show me) Graduate Education review schedule.
24	GED	(Tôi muốn Cho Tôi) (biết hỏi) văn bằng hai có những ngành đào tạo nào. (I want Please) [to] (know ask show me) which faculties does Secondary Bachelor Degree have?
25	GED	(Tôi muốn Cho Tôi) (biết hỏi) yêu cầu ngoại ngữ đầu vào của chương trình thạc sĩ. (I want Please) [to] (know ask show me) Foreign Language requirements of Master Degree program
26	FIT	(Tôi muốn Cho Tôi) (liên lạc nối máy gặp) khoa CNTT. (I want Please) [to] (contact connect speak) to Faculty of Information Technology.
27	FIT	(Tôi muốn Cho Tôi) (biết hỏi) (điều kiện lệ phí thời gian) phúc khảo bài thi. (I want Please) [to] (know ask show me) (conditions fee time) of Remarking test.
28	FIT	(Tôi muốn Cho Tôi) (biết hỏi) (điều kiện lệ phí thời gian) mở lớp học hè. (I want Please) [to] (know ask show me) (conditions fee time) of summer courses.
29	FIT	(Tôi muốn Cho Tôi) (biết hỏi) (điều kiện cách thức) bảo lưu chương trình [và theo học lại]. (I want Please) [to] (know ask show me) (conditions how to) [of] course reservation.
30	FIT	(Tôi muốn Cho Tôi) (biết hỏi) về (thực tập tốt nghiệp khoá luận tốt nghiệp nghiên cứu khoa học). (I want Please) [to] (know ask show me) about (Graduate Practise Graduate Thesis Science Research).
31	FIT	(Tôi muốn Cho Tôi) (biết hỏi) khoa CNTT có những (môn bộ môn chuyên ngành) nào. (I want Please) [to] (know ask show me) which courses does Faculty of Information technology have?
32	FIT	(Tôi muốn Cho Tôi) (biết hỏi) (lịch thời khoá biểu thời gian học) môn tin học đại cương. (I want Please) [to] (know ask show me) schedule of Informatic overview course.

A. DCG rule-based method

In our system, so as to perform semantic commands, we utilized DCG with 14 structures. All means of performance are presented in Table III.

TABLE III. THE SEMANTIC PRESENTATION OF QUESTIONS

ID	Semantic presentations
1	query(callto(Dept))
2	query(course(Task))
3	query(get(Cert))
4	query(have(Faculty))
5	query(admissions(Score,Faculty,Year))
6	query(calculate(Score))
7	query(graduate(Requirement))
8	query(policy(Scholarship))
9	query(program(Voluntrer))
10	query(fee(WhQues))
11	query(schedule(Revise))
12	query(require(Subject))
13	query(fit(Course))
14	query(time(Subject))

Example 1: Cho tôi hỏi cách huỷ môn học như thế nào?
(I would like to know how to drop a course)

The syntactic & semantic rules in DCG are defined as below:

```

query(query(Course))-->w_ques, np_course(Course),w_tail.
w_ques-->[cho, tôi, hỏi].
w_tail-->[như, thế, nào].
np_course(course(hury))-->[cách, hủy, môn, học].
    
```

These syntactic and semantic rules determine the semantic structure of the command as: query(course(huỷ)). The semantic structure is the Structure 2 in Table III. With above semantic structures, the system will automatically convert these to the associated SQL queries.

We have done manual tests including 100 sentences for evaluating the performance of the Vietnamese processing component. They are sentences, which are found in 14 pattern rules, built with a lot of respect to the system. The latter is capable of handling all the pattern sentences. For sentences not covered by the syntactic structure, the system will return the false parsing result. These suggest that the DCG syntax rules which topics have built and dictionaries still cannot cover all the cases. If additional dictionaries from perfect DCG syntax rules, the coverage of the system will be increased to much higher level.

B. SVM Machine learning method

We collected 2,500 demo calls corresponding to 50 persons (by survey). All data was preprocessed and manually labeled as ARO, SSD, FD, GED, FIT. Next, word segmentation and POS tagger were performed, we chose vnTokenizer [4]. Then, we removed stop-words. We also removed all features whose number of occurrences doesn't meet a threshold. In this study, the threshold was set 3. Finally, the training set was vectorized and an SVM is used to compute a separating hyperplane.

250 calls were collected randomly for testing. We evaluated how well the system can identify ARO, SSD, FD, GED, FIT from the test data using the standard Precision, Recall and F-score measures. Figure 5 shows the results of the system running on test data with threshold = 3.

Corpus statistics	Document statistics			
Annotation	Match	Prec.B/A	Rec.B/A	F1.0-s.
comment	246	0.9840	0.9840	0.9840
Macro summary		0.9840	0.9840	0.9840
Micro summary	246	0.9840	0.9840	0.9840

Fig. 5. The results of the system running on test data with threshold =3

VI. EXPERIMENTS AND EVALUATION

The first test is performed on Speech Recognition. Next, we perform experiments on the system, as well as implement the perception survey/ assessment of users on the system, including speech synthesis component.

A. Speech Recognizer

The speech recognition performance is typically evaluated in terms of Word Error Rate (WER), which can then be computed as: $WER = (S + D + I) / N \times 100\%$ [6], where N is the total number of words in the testing data, S denotes the total number of substitution errors, D is the total number of deletion errors and I is the total number of insertion errors. We make use of Word Accuracy (WA) [6] instead, which is computed as $WA = (1 - (S + D + I) / N) \times 100\%$, to report performance of the speech recognizer. The accuracy of the system is reported in Table IV.

TABLE IV. TEST RESULT BY CAPACITY OF CORPUS

Model	Descriptions	Result (accuracy)	
		Trained users	Untrained users
VNSS_C10	Train corpus of 10 speaker	99%	93%
VNSS_C20	Train corpus of 20 speakers	98%	94%
VNSS_C30	Train corpus of 30 speakers	97%	95%

B. Investigation

We have also performed the survey to 50 users using the system with the question: "The system is easy to use or not?" with 4 levels of evaluation, and the results in Table V.

TABLE V. COMFORT RESULTS

Very comfort	Fairly comfort	A bit comfort	Not comfort
28%	24%	26%	22%

C. System Experiments

Text processing: DCG rule-based method is capable of handling all the pattern sentences but for sentences not covered by the syntactic structure, the system will return the false parsing result. So we chose SVM machine learning for text classifier.

TABLE VI. EXPERIMENTAL ENVIRONMENTS

Number of Questions	100
Environment	in-door
Sampling rate	8 kHz
Quantization	16 bits
Format	PCM

The system correctly analyzes and executes 95/100 of the spoken questions in Vietnamese language. The fault cases must be remained at the speech recognition step. So, our system demonstrates its accuracy attaining 95%. About 3.4 seconds for a command is spent as the average feedback time of the system.

VII. CONCLUSION

This paper has presented the architectural model of EduICR system as well as our approach to build it. In our opinion, this is one of the first systems in Vietnam equipped with a mechanism for text processing efficiency in voice applications. This study also opens up a new direction for the construction and development of systems inquiry that can understand and communicate with Vietnamese speaking users. Our upcoming improvement is to enrich the routing rules and widen application-oriented vocabulary.

ACKNOWLEDGMENT

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A Privacy-Preserving Roaming Authentication Scheme for Ubiquitous Networks

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Abstract—A privacy-preserving roaming authentication scheme (PPRAS) for ubiquitous networks is proposed, in which a remote mobile user can obtain the service offered by a foreign agent after being authenticated. In order to protect the mobile user's privacy, the user presents an anonymous identity to the foreign agent with the assistance of his or her home agent to complete the authentication. After that, the user and the foreign agent can establish a session key using the semi-group property of Chebyshev polynomial. In this way, huge burden of key management is avoided. Furthermore, the user can update the login password and the session key between itself and the foreign agent if necessary. The correctness is proved using BAN logic, and the performance comparison against the existing schemes is given as well.

Keywords—roaming authentication; anonymous; chaotic maps; key agreement

I. INTRODUCTION

High-speed development of mobile internet has a profound influence on people's daily life. The mobile user wishes to share something or get some resources via mobile devices anytime anywhere and it should not be an issue when he or she locates in the range of the home network provider. However, when a mobile user moves to the region of a foreign network, how does he or she access the foreign network. Undoubtedly, as shown in Fig. 1., the ubiquitous networks should be equipped with authentication and session key establishment before it permits the user to access the Internet provided by itself.

Many authentication and key establishment protocols for mobile networks [1-7] have been proposed in recent years. In 2009 Chang et al. [1] proposed an efficient authentication protocol for mobile devices, which uses one-way hash functions and exclusive-or operation to reduce computation, and they claimed that their scheme can achieve perfect forward secrecy. However, their protocol cannot protect user's privacy since plaintext of real identities are used during the authentication. Later, Chang et al. [2] proposed another

enhanced authentication scheme, which uses a random number and one-way hash functions to protect the user's identity, while the scheme cannot prevent insider attack as a malicious inner user can get the real identity at ease. Li et al. [8] proposed an efficient mobile networks authentication scheme, which can protect mobile users' privacy, while it is vulnerable to the man-in-the-middle attacks. Shin et al. [9] and Wen et al. [10] proposed anonymous authentication schemes for mobile networks respectively, while Shin et al.'s [9] scheme cannot resist to the man-in-the-middle attacks, and Wen et al.'s [10] scheme will reveal the user's real identity. In 2014, Xie et al. [11] proposed a mobile roaming authentication protocol and claimed this scheme can protect users' privacy; however, its efficiency is not desirable. Mao et al. [12] proposed an anonymous authentication for global mobility networks in the same year. Recently, Farash et al. [13] proposed a light weight authentication scheme for roaming ubiquitous networks, while it is vulnerable to the replay attacks.

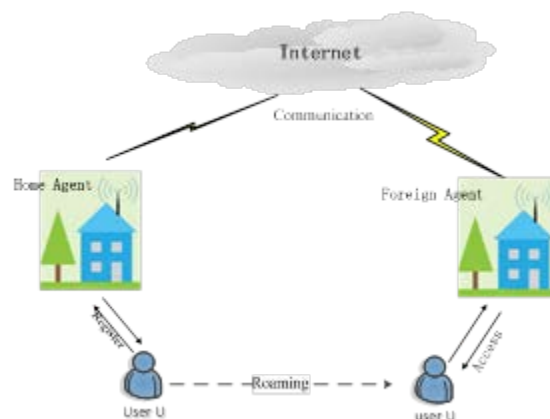


Fig. 1. The scenario of roaming authentication

To improve the security issues, some protocols [14-16] use smart card to authenticate and establish session key. In 2010,

Li et al. [15] proposed an efficient authentication protocol using smart card to make user be anonymous, which enhances the security with untraceability property. Recently, much work on Chebyshev chaotic map based authentication with smart card [17-21] have been done. Juang et al. [22] proposed an authenticated key agreement using smart card, which is privacy-preserving and time-synchronization free. However, in 2009, Sun et al. [23] pointed out that Juang et al.'s [22] scheme suffers inability of the password-changing operation and the session-key problem, hence they proposed an improved authentication protocol using smart card. In 2013, Guo et al. [21] proposed a password-authenticated using smart card. In 2015, Lin et al. [24] proposed an improved chaotic maps based authentication protocol using smart card.

As the popularity of mobile network-enabled devices, people are fond of dealing all work on those devices. However, the private information, for example user identification, may be illegally intercepted and then tracked by the potential attackers. However, the existing schemes either fail to provide privacy preserving or incur huge key management, since traditional symmetric or asymmetric encryption is employed for the handshake message. To address mobile users' privacy effectively, a privacy preserving roaming authentication and key agreement (PPRAS) is proposed in this paper. In PPRAS, the smart card together with chaotic maps is employed to improve efficiency and simplify the session key agreement and key management. In the proposed scheme, the foreign agent can authenticate the mobile user without knowing the user's real identity, then they can agree the shared session key and the temporary identification.

The rest of the article is organized as follows, some related basics are briefly reviewed in section II. The concrete construction of PPRAS is illustrated in section III. Analysis and comparison are presented in section IV. The paper is concluded in the last section.

II. PRELIMINARIES

A brief introduction of the Chebyshev maps and some related basics are given in this section.

A. Chebyshev Chaotic Maps

Definition 1: Let n be an integer, $x \in [-1,1]$, an n -order Chebyshev polynomial map $T_n(x) : [-1,1] \rightarrow [-1,1]$ is defined as follows:

$$T_n(x) = \cos(n * \cos(x))$$

According the definition, the recursive form of Chebyshev polynomial map can be produced as follows

$$T_n(x) = 2 * x * T_{n-2}(x) - T_{n-2}(x), n \geq 2,$$

$$\text{where } T_0(x) = 1, T_2(x) = x, n \geq 2.$$

The Chebyshev polynomial map follows the following two properties

1) Semi-group property

$$\begin{aligned} T_r(T_s(x)) &= \cos(r * \cos^{-1}(s * \cos^{-1}(x))) \\ &= \cos(r * s * \cos^{-1}(x)) \\ &= T_{sr}(x) = T_s(T_r(x)) \end{aligned}$$

where r, s are two integers, $x \in [-1,1]$.

2) Chaos property

When $n > 1$, a n -degree Chebyshev polynomial map $T_n(x) : [-1,1] \rightarrow [-1,1]$ has the constant measure $f^*(x) = 1 / (p\sqrt{1-x^2})$ and positive Lyapunov exponent $\lambda = \ln n > 0$.

B. The Extended Chebyshev Chaotic Maps

According to the periodicity of $y = \cos(x)$, there exist multiple x associated with the same y to make the equation hold. Zhang [25] proved that the Chebyshev polynomial map still keeps the semi-group property over the interval $(-\infty, \infty)$, and proposed the concept of the extended Chebyshev chaotic maps as follows.

$$T_n(x) = (2xT_{n-1}(x) - T_{n-2}(x)) \bmod P,$$

where $n \geq 2, x \in [-1,1]$, and P is a big prime number.

Furthermore, the following equation holds as well:

$$T_r(T_s(x)) = T_{sr}(x) = T_s(T_r(x)) \bmod P$$

Definition 2: Discrete Logarithm Problem (DL)

Given any two big integer x, y , find an integer s to satisfy the equation $T_s(x) \equiv y$.

Definition 3: Decisional Diffie-Hellman Problem (DDH)

Given $T_r(x), T_s(x), T_u(x)$, where r, s, u are unknown, determine the equation $T_{sr}(x) = T_u(x) \bmod P$ holds or not.

III. CONSTRUCTION OF PPRAS

In this section, the detailed construction of PPRAS is presented. For convenience, the descriptions of some symbols to be used are listed in TABLE I.

In PPRS, there exist three entities: the mobile user MU , the home agent HA and the foreign agent FA . When MU moves to FA 's network, FA needs to authenticate MU before giving him the permission to access the network. To finish the authentication, FA needs the assistance of HA to verify whether MU is an authorized user or not. If not, the authentication process will be terminated. The proposed scheme consists four stages: registration phase, authentication phase including session key establishment, session key update and login password update phase.

During the initialization, FA shares a session key with HA , which is securely stored locally. The authentication is launched by MU , and then proceeds as the following interactive steps.

TABLE I. DESCRIPTION OF SYMBOLS

Symbol	Description
ID_i	identification of communication entity i
SID	temporary identification of mobile user(MU)
$T_n(x)$	Chebyshev polynomial with degree n
T_s	$T_s(x)$
T_{MU}, T_{FA}	$T_{r_{MU}}(x), T_{r_{FA}}(x)$
x	the initial value of chaotic map
s	private key of the home agent
P	a big prime number
x_i, r_i	random numbers chosen by users
K_{MU}	session key shared between MU, FA and HA
k_{HF}	the shared key between FA and HA
$E(\cdot) / D(\cdot)$	symmetric encryption/decryption algorithm
t_{MU}, t_{FA}	timestamp
ΔT	threshold of interval
$H(\cdot)$	a secure one-way hash function
\oplus	XOR operation
PW_{MU}	password of mobile user
T_H	running time for hash operation
T_E	running time for encryption operation

T_D	running time for decryption operation
T_C	running time for chaotic map operation
T_M	running time for modular exponential operation

A. Registration Phase

A mobile user MU registers himself in his or her home agent HA using the following steps,

1) HA chooses two random numbers x, s and a big prime number P , then computes $T_s = T_s(x) \bmod P$, and publishes (x, T_s, P) .

2) MU chooses his PW_{MU} and a random number λ , then computes $H = h(PW_{MU}, \lambda)$, then sends $\{ID_{MU}, H\}$ to HA via a secure channel.

3) HA checks the validity of ID_{MU} and $H(PW_{MU}, \lambda)$ using $H(H(PW_{MU}, \lambda) \parallel ID_{MU})$. If yes, computes the message $IM = h(ID_{MU} \parallel s \parallel t_{reg})$ which respect the identity of MU , with his secret key s and the timestamp t_{reg} , then store the parameters $\{ID_{MU}, H, IM, x, T_s, ID_{HA}, H(\cdot), E(\cdot), T_n(\cdot), P\}$ into a smart card and send it to MU , where $T_n(\cdot)$ is a Chebyshev polynomial with degree n among them. Otherwise, MU fails to register in the system.

B. Authentication and Key Establishment Phase

MU and FA can complete the authentication and establishment by following the steps shown in Fig. 2.

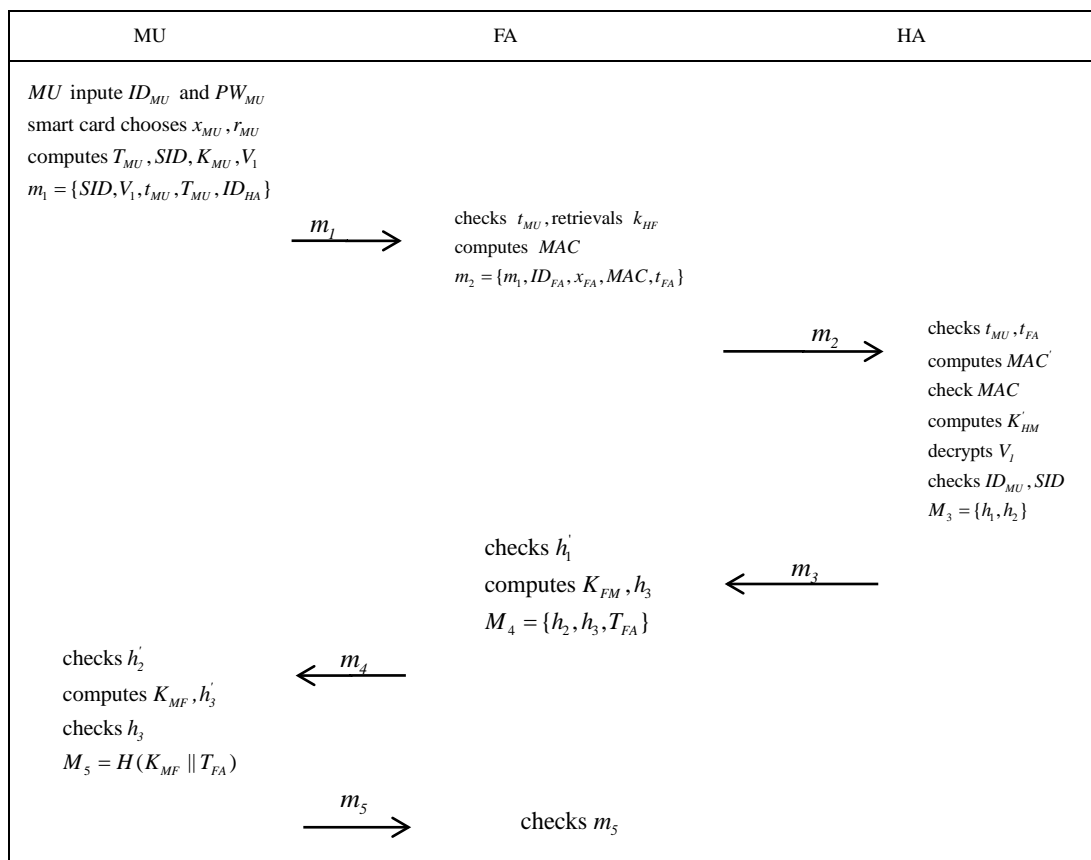


Fig. 2. The process of authenticating and key establishing

- $MU \rightarrow FA : m_1 = \{SID, V_1, T_{MU}, t_{MU}, ID_{HA}\}$

MU first inputs his real identity ID_{MU} and password PW_{MU} into the smart card, then the smart card (SC) make the decision that allowing MU to login or not by computing $H' = h(PW_{MU}, \lambda)$ and checking validity of ID_{MU} and $H' ? = H$. If yes, SC chooses two random numbers: x_{MU} , r_{MU} , then computes $T_{MU} = T_{r_{MU}}(x) \bmod P$, $K_{MH} = T_{r_{MU}}(T_s) \bmod P$ and $SID = ID_{MU} \oplus H(x_{MU})$, where SID denotes the temporary identification of MU , and K_{MH} denotes the shared session key between MU and HA . After that, SC encrypts ID_{MU} , ID_{HA} , IM , x_{MU} and the current timestamp t_{MU} using K_{MH} , that is $V_1 = E_{K_{MH}}(ID_{MU} || ID_{HA} || IM || x_{MU} || t_{MU})$. Next, MU sends $m_1 = \{SID, V_1, T_{MU}, t_{MU}, ID_{HA}\}$ to FA .

- $FA \rightarrow HA : m_2 = \{m_1, ID_{FA}, x_{FA}, MAC, t_{FA}\}$

Upon receiving m_1 from MU , FA firstly checks $|t_{FA} - t_{MU}| < \Delta T$ holds or not, where t_{FA} is the current time of FA , ΔT denotes the permissible threshold of time interval. If yes, stores SID temporarily firstly, and then searches the shared session key k_{HF} between FA and HA using ID_{HA} . Next, computes the message authentication code MAC :

$MAC = h(ID_{FA} || V_1 || x_{FA} || t_{FA} || k_{HF})$, where x_{FA} is a random number chosen by FA temporarily. At last, sends the message $m_2 = \{m_1, ID_{FA}, x_{FA}, MAC, t_{FA}\}$ to HA .

- $HA \rightarrow FA : m_3 = \{h_1, h_2\}$

After receiving m_2 from FA , HA firstly checks $|T - t_{FA}| < \Delta T$, $|T - t_{MU}| < \Delta T$ holds or not, where T denotes the timestamp of HA , ΔT denotes the permissible threshold of time interval.

If these two equation hold, HA confirms ID_{FA} and SID as follows:

Step 1. Uses ID_{FA} retrievals the shared session key k_{HF} between HA and FA , then computes $MAC' = h(ID_{FA} || V_1 || x_{FA} || t_{FA} || k_{HF})$ and checks whether $MAC' = MAC$ holds or not.

Step 2. If yes, HA confirms the identity ID_{FA} from FA , then computes $K_{HM} = T_s(T_{MU}) \bmod P$ to decrypt V_1 , and checks whether t_{MU} , ID_{HA} in V_1 are all equal to the plaintext t_{MU} , ID_{HA} in message m_1 . If yes, uses the decrypted ID_{MU} to retrieval his database to check whether $t_{MU} > t_{reg}$. If holds, computes $IM' = h(ID_{MU} || s || t_{reg})$ and $SID' = ID_{MU} \oplus H(x_{MU})$,

then checks if $IM' = IM$ and $SID' = SID$. If they all hold, confirms the anonymous identity SID is valid.

Step 3. Computes the message $m_3 = \{h_1 = h(SID || x_{FA} || k_{HF} || h_2), h_2 = h(IM || x_{MU} || k_{HM})\}$, then sends it to FA .

- $FA \rightarrow MU : m_4 = \{h_2, h_3, T_{FA}\}$

After receiving m_3 from HA , FA firstly computes $h'_1 = h(SID || x_{FA} || k_{HF} || h_2)$, then checks if $h'_1 = h_1$. If yes, confirms the temporary identification SID of MU is valid. After that, FA chooses a random number r_{FA} , then computes $T_{FA} = T_{r_{FA}}(x) \bmod P$, $K_{FM} = T_{FA}(T_{MU}) \bmod P$ and $h_3 = h(SID || k_{FM} || h_2 || T_{FA})$, where K_{FM} is the session key between FA and MU , then sends $m_4 = \{h_2, h_3, T_{FA}\}$ to MU .

- $MU \rightarrow FA : m_5 = h(K_{MF} || T_{FA})$

After receiving m_4 from FA , MU firstly computes $h'_2 = h(IM || x_{MU} || K_{HM})$, checks whether $h_2 = h'_2$ holds or not. If yes, MU confirms FA is authenticated, then computes the following values: $K_{MF} = T_{MU}(T_{FA}(x)) \bmod P$ and $h'_3 = h(SID || k_{MF} || h_2 || T_{FA})$, where K_{MF} is the session key between MU and FA , then checks if $h'_3 ? = h_3$. If yes, establishes the session key K_{MF} , then MU computes the message $m_5 = h(k_{MF} || T_{FA})$ and sends it to FA .

- Upon receiving m_5 from MU , FA firstly computes $M'_5 = h(k_{FM} || T_{FA})$, then check if $M'_5 ? = M_5$. If holds, completes establishing the session key.

C. Session Key Update Phase

In order to ensure the security, it is necessary for MU to periodically update the session key established previously between himself and FA . MU follows the following steps to update his or her session key in the i^{th} time:

Step 1. MU firstly chooses a number t_i randomly, computes $T_{vMU} = T_{t_i}(x) \bmod P$ and $m_i = \{E_{K_{MF_i}}(SID, T_{vMU}, t_{vMU}, Ch), SID, t_{vMU}\}$, then sends the message m_i to FA , where SID is the anonymous identity of MU when he establishes the session key k_{MF_i} between himself and FA at the previous time, t_{vMU} is current timestamp, Ch is the flag to denotes update query.

Step 2. Upon receiving the message m_i from MU , firstly checks $|T_i - t_{vMU}| < \Delta T$ holds or not, where T_i is the current time of FA , ΔT denotes the permissible threshold of time interval. If yes, FA uses SID to get the i^{th} session key K_{FM_i} to decrypt m_i and check whether SID is equal to the plaintext SID .

If yes, FA chooses a random number r_{i+1} , and computes $T_{FA_{i+1}} = T_{r_{i+1}}(x) \bmod P$, $k_{FM_{i+1}} = T_{FA_{i+1}}(T_{v_{MU}}(x)) \bmod P$, where $k_{FM_{i+1}}$ is the current session key. Then FA sends the message $m_{i+1} = \{E_{K_{FM_i}}(h(K_{FM_{i+1}} \| K_{FM_i}), T_{FA_{i+1}}, ID_{FA}), ID_{FA}\}$ to MU

Step 3. After receiving m_{i+1} from FA , MU firstly uses the previous session key k_{MF_i} to decrypt m_2 , computes the new session key $k_{MF_{i+1}} = T_{v_{MU}}(T_{FA_{i+1}}(x)) \bmod P$ and $h'(K_{FM_{i+1}} \| K_{FM_i})$, then checks if $h'(K_{FM_{i+1}} \| K_{FM_i}) = h(K_{FM_{i+1}} \| K_{FM_i})$ holds or not. If yes, completes session key update.

D. Login Password Update Phase

It is necessary for MU to update his or her login password dynamically to prevent someone else who knows his or her password from doing some impersonation attacks. The update of login password can be finished as follows:

- Step 1. MU puts his or her smart card into the reader, then inputs his or her real identity ID_{MU} and the password PW_{MU} , then the smart card can make the decision that allowing MU to login or not by computing $H' = h(PW_{MU}, \lambda)$ and checking validity of ID_{MU} and $H' ? = H$. If yes, MU sends update request.
- Step 2. When the smart card receives the request, it asks MU to input the new password PW'_{MU} , and a random number t' if necessary, then the smart card computes $H' = h(PW'_{MU}, t')$ and updates it.

IV. ANALYSIS OF PPRAS

A. Correctness Analysis

The Burrows–Abadi–Needham (i.e. BAN) logic [27] is useful to identify some possible weakness in the security protocols, especially for the authentication protocol, so the BAN logic is used to analyze the correctness of PPRAS. Some notations are listed in TABLE II.

TABLE II. NOTATIONS FOR BAN LOGIC

Notation	Description
$A \equiv X$	A trusts X , or A believes X
$A \triangleleft X$	A sees X , or A holds X
$A \sim X$	A has said X
$A \Rightarrow X$	A completely controls over X
Rule 1	Rule 2 comes from Rule 1
Rule 2	
$A \xrightarrow{x} B$	x is a secret key or information between A and B
$\{X\}_K$	X is encrypted by the key K

1) Idealization

According to the rules of the BAN logic, the first step is to idealize the authentication phases of PPRAS as follows:

- a) $MU \rightarrow FA$:
 $m_1 = \{SID, \{ID_{MU} \| ID_{HA} \| IM \| x_{MU} \| t_{MU}\}_{MU \xleftarrow{K_{MH}} HA}, T_{MU}, t_{MU}\}$
- b) $FA \rightarrow HA$:
 $m_2 = \{m_1, h(ID_{FA} \| V_1 \| x_{FA} \| t_{FA} \| FA \xleftarrow{K_{FH}} HA), t_{FA}\}$
- c) $HA \rightarrow FA$:
 $m_3 = \{h_1 = h(SID \| x_{FA} \| HA \xleftarrow{K_{HF}} FA \| h_2), h_2 = h(IM \| x_{MU} \| HA \xleftarrow{K_{HM}} MU)\}$
- d) $FA \rightarrow MU$:
 $m_4 = \{h_2, h_3 = h(SID \| FA \xleftarrow{K_{FM}} MU \| h_2 \| T_{FA}), T_{FA}\}$
- e) $MU \rightarrow FA$: $m_5 = h(MU \xleftarrow{K_{MF}} FA \| T_{FA})$

2) Assumptions

In PPRAS, there exist three entities: the mobile user (MU), the foreign agent (FA) and the home agent (HA). Each entity has his or her possessions and abilities. The initial assumptions are described as follows:

For MU :

- A1. $MU \triangleleft ID_{MU}$
 A2. $MU \equiv SID$
 A3. $MU \equiv ID_{HA}$
 A4. $MU \equiv r_{MU}$
 A5. $MU \equiv MU \xleftarrow{K_{MH}} HA$

A1: MU believes his or her own identity.

A2: MU believes his or her own pseudonym SID .

A3: As MU registers himself in his home agent HA to be a legitimate user, so he believes HA 's identity ID_{HA} .

A4: MU believes the number x_{MU} chosen by himself.

A5: MU believes the session key K_{MH} between himself and HA , because K_{MH} is computed using the Chebyshev polynomials with HA 's public parameter T_{HA} and T_{MU} .

For HA :

- A6: $HA \triangleleft ID_{HA}$
 A7: $HA \equiv \#(t_{MU})$
 A8: $HA \equiv \#(t_{FA})$
 A9: $HA \Rightarrow s$
 A10: $HA \equiv HA \xleftarrow{K_{HF}} FA$

A6: HA holds his own identity.

A7: HA believes t_{MU} is fresh, and has never received it before so that he can authenticate MU .

A8: HA believes t_{FA} is fresh, and has never received it before so that he can authenticate FA .

A9: As s is HA 's secret key, so HA completely controls over his secret key s .

A10: HA believes the key shared between HA and FA before authenticating.

For FA :

$$A11: FA \triangleleft ID_{FA}$$

$$A12: FA \triangleleft ID_{HA}$$

$$A13: FA \models \#(t_{MU})$$

$$A14: FA \models FA \xleftarrow{k_{HF}} HA$$

$$A15: FA \models r_{FA}$$

A11: FA holds his own identity.

A12: FA needs to authenticate MU with the help of HA , so he needs to hold HA 's identity ID_{HA} .

A13: FA believes t_{MU} is fresh so that he will be able to finish the next operation.

A14: FA believes the session key K_{FH} between himself and HA , because K_{FH} is computed using Chebyshev polynomials with HA 's public parameter T_{HA} and the value T_{FA} computed by FA himself.

3) Goals

According to the proposed scheme, MU and FA want to establish a session key with the help of HA , so our proposed scheme needs to achieve the following goals:

$$G1: HA \models SID$$

$$G2: HA \models ID_{FA}$$

$$G3: FA \models HA \models SID$$

$$G4: MU \models HA \models ID_{FA}$$

$$G5: MU \models MU \xleftarrow{K_{MF}} FA$$

$$G6: FA \models FA \xleftarrow{K_{FM}} MU$$

G1: HA believes the anonymous identity of MU .

G2: HA believes FA 's identity.

G3: FA believes that HA has verified MU 's anonymous identity SID . G4: MU believes that HA believes FA is a legitimate agent.

G5: MU believes the session key between himself and FA , that is MU has already successfully generated the session key with FA .

G6: FA believes the session key between himself and MU , that is FA has already successfully generated the session key with MU .

MU wants to establish a session key with FA without leaking his identify, he needs an anonymous identify which used to be authenticated by HA , and HA must believe FA 's

identify to enable MU to communicate with FA . After they finish the process of generating the session key, FA and MU must believe the authenticated peer holds the common session key.

4) Verification

In this section, the BAN logic is employed to check whether PPRAS is correct or not. The primary steps are shown as follows:

Theorem 1. HA believes the anonymous identity of MU and the identity of FA .

Proof :

$$V1: \frac{\frac{HA \triangleleft m_2}{\frac{HA \triangleleft h(ID_{FA} \parallel V_1 \parallel x_{FA} \parallel t_{FA} \parallel k), HA \models HA \xleftarrow{k_{HF}} FA}{HA \models FA \sim ID_{FA}, HA \models \#(h(ID_{FA} \parallel V_1 \parallel x_{FA} \parallel t_{FA} \parallel k_{HF}))}}{HA \models FA \models ID_{FA}}}$$

$$V2: \frac{\frac{HA \triangleleft k_{HM}}{\frac{HA \triangleleft (ID_{MU}, IM, x_{MU})}{\frac{HA \triangleleft IM, HA \Rightarrow s}{HA \models ID_{MU}, HA \triangleleft SID}}}{HA \models SID}}$$

According to the assumption A7 and A8, HA believes the message m_1 and m_2 are fresh, and he has never received them before, applying the seeing rule:

$$\frac{A \triangleleft (x, y)}{A \triangleleft x},$$

HA holds $h(ID_{FA} \parallel V_1 \parallel x_{FA} \parallel t_{FA} \parallel k_{HF})$, with the assumption A10, applying the message-mean rule:

$$\frac{P \models P \xleftarrow{k} Q, P \triangleleft \{x\}_k}{P \models Q \sim x},$$

HA believes that FA has said ID_{FA} , applying fresh rule:

$$\frac{A \models \#(x, y)}{A \models \#x},$$

HA believes $h(ID_{FA} \parallel V_1 \parallel x_{FA} \parallel t_{FA} \parallel k_{HF})$ is fresh, applying nonce-verification rule:

$$\frac{P \models \#(x), P \models Q \sim x}{P \models Q \models x},$$

HA believes that FA believes ID_{FA} , so HA believes ID_{FA} .

After verifying the correction of MAC , HA believes the session key k_{HF} , so he believes the identity of FA , and also believes the message m_1 has not been tampered. Then HA computes K_{HM} to decrypt V_1 in message m_1 , applying the seen rule:

$$\frac{A \triangleleft (x, y)}{A \triangleleft x},$$

HA can get ID_{MU} , IM and x_{MU} . According to the assumption A9, HA believes the real identify of MU , then HA

can verify the anonymous identity of MU with the received value SID in message m_1 .

With the proof above, it can be found that HA believes the anonymous identity (SID) of MU and the identity (ID_{FA}) of FA .

Theorem 2. FA believes that HA has verified MU 's anonymous identity SID .

Proof:

$$V3: \frac{\frac{FA \triangleleft m_3}{FA \triangleleft h_1, FA \models FA \xleftarrow{k_{HF}} HA}}{FA \models HA \models SID}$$

After FA receives the message m_3 from HA , applying the seen rules:

$$\frac{A \triangleleft (x, y)}{A \triangleleft x},$$

FA receives the value h_1 in message m_3 . According to the assumption A14, we know that HA has verify the anonymous identity SID , so FA believes that HA believes the anonymous identity SID of MU after he verify the message h_1 in the received message m_3 .

Above all, FA believes that HA has verified MU 's anonymous identity SID .

Theorem 3. MU believes that HA believes FA is a legitimate agent.

Proof:

$$V4: \frac{\frac{\frac{MU \triangleleft m_4}{MU \triangleleft h_2, MU \models MU \xleftarrow{k_{HM}} HA}}{MU \models h_2, MU \models r_{MU}}}{MU \models HA \models ID_{FA}}$$

After MU receives the message m_4 from FA , applying the seeing rules:

$$\frac{A \triangleleft (x, y)}{A \triangleleft x},$$

MU receives the value h_2 in message m_4 . According to the assumption A5, MU believes the value h_2 , and also believes MU believes that HA believes FA 's identity after he or she verifies h_2 under the assumption A4. So MU believes that HA believes FA is a legitimate agent.

Theorem 4. MU believes the session key between himself and FA , that is MU has already generated the session key with FA .

Proof:

$$V5: \frac{\frac{\frac{MU \triangleleft m_4}{MU \triangleleft h_3, MU \models (SID, h_2)}}{MU \models h_3, MU \triangleleft K_{MF}}}{MU \models MU \xleftarrow{K_{MF}} FA}$$

After MU receives the message m_4 from FA , applying the seeing rules:

$$\frac{A \triangleleft (x, y)}{A \triangleleft x},$$

MU receives the value h_3 in message m_4 . According to the assumption A2 and the theorem 3, MU believes h_3 .

After MU verifies h_2 , he or she computes the session key K_{MF} between MU and FA , so MU holds K_{MF} , according to the proof above that MU believes h_3 , MU can verify the key K_{MF} is right with h_3 , that is MU believes the session key between himself and FA .

Theorem 5. FA believes the session key between himself and MU , that is FA has already generated the session key with MU .

Proof:

$$V6: \frac{\frac{\frac{FA \models r_{FA}}{FA \models T_{FM}, FA \triangleleft m_5}}{FA \models FA \xleftarrow{K_{FM}} MU}}$$

After FA receives the message m_5 from MU , according to the assumption A15, applying the belief rules:

$$\frac{A \models x}{A \models (x, y)},$$

FA believes T_{FM} , as FA holds the message m_5 , so FA believes the session key K_{FM} between FA and MU .

B. Performance Analysis

The performance evaluation of the existing protocols [9-13] and PPRAS will be discussed in this section. The overall results are listed in TABLE III.

TABLE III. COMPARISON ON PERFORMANCE

	Client	Server
Farash et al.'s[13]	$6T_H$	$6T_H + 2T_E + 2T_D$
Mao et al.'s[12]	$8T_H + 2T_E$	$4T_H + 2T_E + 4T_D$
Xue et al.'s[11]	$4T_H + T_E + T_D$	$3T_H + 2T_E + 3T_D$
Shin et al.'s[9]	$4T_H$	$7T_H + 2T_E + 2T_D$
Wen et al.'s[10]	$4T_H + T_M$	$10T_H + 3T_M$
PPRAS	$4T_H + T_E + 2T_C$	$9T_H + T_D + 2T_C$

Since the authentication is a series of synchronized processes, the total computational cost of the client and server during the authentication and key agreement should be investigated. As the cost of XOR operation and module addition are rather cheap, these two operations are not included in the comparison, and only symmetric encryption/decryption operation, chaotic map operation, hash operation and modular exponential operation are evaluated. As shown in TABLE III, the computational cost of client in [9,10,13] is much cheap than PPRAS, however, as discussed previously, the scheme in

[9,12] cannot resist to the man-in-the-middle attacks, and the scheme in [10] cannot preserve the user's privacy. However, the efficiency of [11] is not desirable. The scheme in [13] is vulnerable to the replay attacks. Furthermore, the schemes in [9-13] will inevitably incur huge key management for the symmetric and public key encryption. Although no explicit advantage of performance for PPRAS cannot be found in TABLE III, the underlying featured chaotic map based encryption for handshake message would save much more computation and storage cost.

C. Security Analysis

In this section, the security analysis and performance comparison are illustrated.

1) *User Anonymity*: The user who wants to authenticate others should provide its real identifies to the trusted three party in the 3PAKE [26] protocol. If the user transfers authentication messages including his identity in plaintext via an insecure channel, an attacker can identify the user by intercepting and analyzing the message, this is not a desirable scheme for authentication. In PPRAS, the real identity of mobile user is encrypted with the session key computed using Chebyshev polynomial. Even if the adversary got the ciphertext, he or she still faces the difficulty of solving DL hard problem if he or she want to compute the decryption key. Since the temporary identification of MU is generated with the XOR operation on the random number and real identity, it is infeasible in polynomial time to guess the right identity since the space of identity is big enough. Therefore, FA can get nothing about the user's real identity and the privacy of the user is preserved well.

2) *Resistance to The Man-in-The-Middle Attack*: Suppose there exists an active attacker over the communication channel, who attempts to intercept and tamper the messages transferred via this channel to carry out the man-in-the-middle attack. If the attacker wants to tamper m_1 , he or she needs to tamper V_1 in message m_1 produced by symmetric encryption with the session key, which is computed with the Chebyshev polynomials. However the attacker will face the difficulty of solving the DL problem. As for the messages m_2, m_3, m_4, m_5 generated with the secure one-way hash functions, if the attacker wants to tamper them, he or she will face the difficulty of breaking the secure one-way hash functions according to the definition of the protocol. Above all, PPRAS is secure enough to counter the man-in-the-middle attack.

3) *Forward Secrecy*: In PPRAS, the forward secrecy means that even if an adversary has obtained the current session key and the password of MU, he or she cannot deduce the previous used session key. The agreement of the session key K_{MF} (or K_{FM}) between MU and FA is based on the random number x_{MU} and x_{FA} , and even MU does not know x_{MU} which is chosen dynamically by the smart card, so the adversary can get nothing about K_{MF} (or K_{FM}), that is, the proposed scheme achieves forward secrecy.

4) *Backward Secrecy*: The backward secrecy of PPRAS refers to the adversary cannot successfully fulfil authentication

and session key agreement with the password of MU and all previous used session key together with the current session key. However, all the messages are produced by the smart card and transferred in anonymous way, thus he or she cannot generate a valid message without possessing this smart card according to the protocol, even if he or she is given PW_{MU} . So PPRAS achieves the backward secrecy.

5) *Resistance to Password Guessing Attack*: This attack means that an attacker attempts to deduce the password of the user with interception and analysis over the transferred messages. In PPRAS, however, there does not exist user's password in all these messages, and the attacker can get nothing about user's password. Thus, the proposed scheme can resist to password guessing attack.

6) *Resistance to The Replay Attack*: According to the construction of the presented protocol, all the transferred messages among MU, FA and HA combine the timestamp t_{FA}, t_{MU} to provide freshness. What's more, the parameters (x_{MU}, r_{MU}) and (x_{FA}, r_{FA}) are chosen randomly to ensure freshness at the beginning of every authentication session. So the adversary can not replay those messages.

Finally, the overall security comparison of PPRAS and the existing similar schemes are listed in TABLE IV. As shown in the table, only PPRAS can achieve all the security features.

TABLE IV. COMPARISON ON SECURITY

Security Features	Farash et.al.'s[13]	Mao et al.'s[12]	Li et al.'s[8]	Shin et al.'s.[9]	Wen et al.'s[10]	PPRAS
Forward Secrecy	✓	✓	✓	✓	✓	✓
Backward Secrecy	✓	✓	✓	✓	✓	✓
Anti-replay attack	✓	✓	✓	✓	✓	✓
Anti-MIM attack	✓	✓	✓	✓	✓	✓
User Anonymity	✓	✓	✓	✓	✓	✓
Anti-guessing attack	✓	✓	✓	✓	✓	✓

V. CONCLUSIONS

Roaming authentication is essential to the ubiquitous networks, and a lot of efforts have been done to better the security and performance in authentication. However, the existing authentication protocols cannot avoid the huge burden of key agreement and management for authentication which comes from the encryption and poses a barrier to apply it to the multi-user situations. Thus a novel roaming authentication scheme based on Chebyshev chaotic map with user anonymity is proposed in this paper. With the advantage of semi-group property of Chebyshev polynomial, the entities involved in the authentication can agree the session key at low cost, and no additional key management is needed. Meanwhile, the foreign agent can authenticate the user without knowing his real identity, which achieves privacy preserving for the user.

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Segmentation and Recognition of Handwritten Kannada Text Using Relevance Feedback and Histogram of Oriented Gradients – A Novel Approach

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Abstract—India is a multilingual country with 22 official languages and more than 1600 languages in existence. Kannada is one of the official languages and widely used in the state of Karnataka whose population is over 65 million. Kannada is one of the south Indian languages and it stands in the 33rd position among the list of widely spoken languages across the world. However, the survey reveals that much more effort is required to develop a complete Optical Character Recognition (OCR) system. In this direction the present research work throws light on the development of suitable methodology to achieve the goal of developing an OCR. It is noted that the overall accuracy of the OCR system largely depends on the accuracy of the segmentation phase. So it is desirable to have a robust and efficient segmentation method. In this paper, a method has been proposed for proper segmentation of the text to improve the performance of OCR at the later stages. In the proposed method, the segmentation has been done using horizontal projection profile and windowing. The result obtained is passed to the recognition module. The Histogram of Oriented Gradient (HoG) is used for the recognition in combination with the support vector machine (SVM). The result is taken as the feedback and fed to the segmentation module to improve the accuracy. The experimentation is delivered promising results.

Keywords—Optical character recognition; Histogram of oriented gradients; relevance feedback; segmentation; Support Vector Machine; handwritten Kannada documents

I. INTRODUCTION

Optical character recognition (OCR) refers to a process of transforming the images of either handwritten or printed document to a machine readable and editable format. In general, all OCR systems have the following stages: image preprocessing, segmentation, extraction of features and finally recognition of characters. The results of each of these stages are greatly affected by the performance of the previous stages. To make the results of the subsequent stages more accurate, segmentation plays an important role. The extraction of region of interest from the given image is termed as segmentation. In the segmentation of document images, first we extract the lines then the words and finally the characters. Segmentation of characters from a document is still a open challenge in the are of developing efficient OCR systems.

Because of the large dataset and structural complexity, the development of OCR for some of the Indian languages like kannada and telugu is considered to be a tedious task [1]. To add to these complexities in some cases the characters may overlap with each other. In spite of several attempt, the development a high accuracy OCR system for all the Indian languages is still a open challenge. The rest of the paper is organized as follows: In section II a brief discussion about the previous work is reported, the proposed method details can be found in section III. Section IV discusses the experiments and results followed by the conclusion in section V.

II. LITERATURE

In the recent past, due to the existence of digital library of India, the amount of document images for various Indian languages has grown tremendously. The library has taken care to collect documents from different sources and also retaining the original structure, size, font etc. Developing a robust OCR to handle all these issues is still a open challenge. In [1] the authors have highlighted the complexities involved in the segmentation of handwritten documents for some of the south Indian languages like tamil, telugu and malyalam. The existence of the curved characters poses special challenges in the segmentation process. Different strategies like Graph based, Hough transform based, and projection based techniques are proposed for the segmentation of the documents [2]. Arivazhagan et al. [3] proposed the projection-based algorithm in which first obtains candidate lines from the piece-wise projection profile of the document .The lines traverse around any obstructing handwritten connected component by associating it to the line above or below. The author claims that the proposed method is invariant to the skew present in the documents. A level set based new approach for the text line segmentation was proposed by Li et al [4].

In [5] a grouping approach for segmentation was suggested in which a block of connected components are grouped together to identify the characters in a text document. But this approach cannot be used on degraded documents as claimed by the authors. A combination of iterative hypothesis validation through hough transformation and connected components was proposed in [6]. This method is found to be effective in skewed

documents. A peak fringe number (PFN) based approach for segmentation is proposed in [7]. Here the author compute the fringe map for the text document and from that they calculate the PFN. This is used to perform the line segmentation. A method based on separation of header line, base line and contour is presented in [8] for the handwritten Hindi text documents. The authors have claimed that this method is invariant to non-uniform skew in the document. In [9&10] segmentation of English characters where proposed based on the skeletonization methods. A combination of horizontal and vertical projection profiles method is presented in [11] for the Gurumukhi handwritten characters.

It is observed from the literature survey that a lot of work is done for languages like Chinese and English where as a very few work is reported for some of the south Indian languages like kannada, tamil and telugu. This served as motivation for us to develop an efficient segmentation and recognition method for the handwritten kannada documents. A sample of kannada vowels and consonants are shown in figure 1 and 2 respectively.

ಅ ಆ ಇ ಈ ಉ ಊ ಋ ಎ ಏ ಐ ಒ ಓ ಔ

Fig. 1. Kannada vowel samples

ಕ ಖ ಗ ಘ ಙ
ಚ ಛ ಜ ಝ ಞ
ಟ ಠ ಡ ಢ ಣ
ತ ಥ ದ ಧ ನ
ಪ ಫ ಬ ಭ ಮ
ಯ ರ ಲ ವ ಶ ಷ ಸ ಹ ಳ

Fig. 2. Kannada consonants samples

III. PROPOSED METHOD

In this paper, we propose a relevance feedback based approach for the segmentation of handwritten kannada documents. Traditionally in all the optical character recognition system, the output of the segmentation process is fed as input for the recognition phase. If, the sample is wrongly segmented then the recognition system fails to recognize such samples. However, this information is not communicated back to the segmentation phase. In our proposed method we have attempted to fix this gap between the segmentation and recognition phases.

A. Segmentation module

Firstly, we extract the lines from the input document using horizontal projection profile method proposed in [12]. Here we compute the number of ON pixels along the row in the image. A very minimum number of ON pixels represent the rows with no contents. This will help us to identify starting and end of the lines and hence we can extract the lines from the documents. Once we extract the lines in the next step we try to extract the characters from each line. We have employed an adaptive window based technique to extract the characters. At the beginning, the width of the window is initialized to a

predefined quantity. From empirical data we have identified this predefined width of characters. Using this window, we extract the character and this is passed to the recognition module for the feedback. If the recognition module can correctly classify the sample then, we consider the character is correctly segmented. If the recognition module is not able to identify the character, then this information is communicated back to the character segmentation phase and the window width will be increase by 'x' quantity and again the character segmentation will be done. This process will be continued till either the character is correctly classified or till the width of the window reaches the double the initial size. These steps are summarized in the following algorithm:

Input: Handwritten kannada document image of size $m \times n$

Output: Set of segmented characters

Step 1: If the image is in RGB format then convert it to monochrome image using otsu's thresholding method

Step 2: Calculate the number of On pixels in every row. Any valvue close to zero represent the discontinuity. This will help to identify the start and end of height of a line. Using this information perform line segmentation.

Step 3: For every line do

Step 3.1: Initialize the window width to a predefined value

Step 3.2: Extract a character using this window

Step 3.3: Pass the segmented image to the recognize module

Step 3.4: If the root mean square error between the training and segmented sample is less than a predefined threshold then accept the segmented image else return negative acknowledgment to the segmentation module

Step 3.5: If negative acknowledgement is received then increase the window width by 'x' quantity. If window width is less than 2 times the original window width then proceed to step 3.2 else proceed to step 3.1.

B. The Recognition Module

The relevance feedback is provided by this character recognition module. The performance of the segmentation is dependent on the performance of this recognition module. The accuracy of the recognition system depends on the features that we extract from the image. To achieve high accuracy, we preferred to use histogram of oriented gradients, which is believed to be free from illumination changes and shading [13].

Navneet dallal et. al proposed histogram of oriented gradient descriptors which are widely used in various applications of image processing and computer vision. The basic concept behind the HoG is that the shape of the object within an image can be easily captured by the distribution of edge directions. To implement this, we divide the images into smaller connected regions called as cells. We calculate the gradient directions for each pixel in each cell. To enhance the performance and to make it invariant to illumination changes, we contrast normalize the local histograms by calculate the measure of intensity across a larger part of image known as blocks. The procedure to extract the HoG descriptor is

described in the next algorithm. We have used the support vector machines (SVM) for the classification of the samples.

Input: segmented sample

Output: HoG descriptor

Step1: Gradient computation: We can calculate the gradient magnitude and direction for a given image I, as

$$\text{Gradient, } G = \sqrt{I_x^2 + I_y^2} \quad (1)$$

$$\text{And orientation of gradient, theta} = \arctan \frac{I_x}{I_y} \quad (2)$$

Where I_x and I_y are obtained by convolving the given image I with the masks $D_x = [-1 \ 0 \ 1]$ and

$$D_y = \begin{bmatrix} -1 \\ 0 \\ 1 \end{bmatrix} \text{ respectively.}$$

Step2: Orientation binning: In this step, we calculate the cell histogram. Each pixel that belongs to the cell, cast its weighted vote for the orientation based histogram. We have used unsigned gradients and a total of 9 bins for the histogram channels. The weight of the vote depends on the gradient magnitude.

Step 3: Obtaining the HoG descriptor:

In order to nullify the effect of illumination and shading the cell histograms obtained in step 2 need to be normalized. The normalization is done based on the overlapping blocks. The normalized cell histogram values are represented in the form of a vector and this is called as HoG descriptor.

IV. EXPERIMENTS AND RESULTS

The method was tested on the standard dataset, Kannada Handwritten Text Document (KHTD) Dataset which was proposed in [14]. The authors have considered four different category of kannada text. They are related to sports, medical documents, movies and general news. The data is collected from 51 individuals who belongs to different age groups and have different educational qualifications. The data was captured in unruled A4 size papers and the authors were free to choose the type of pens. On an average there were 21 lines per every document. The collected documents are then scanned using a flat bed scanner at a resolution of 300 dpi. We have considered 200 such documents for the experimentation. An average segmentation accuracy of 94% is achieved by the

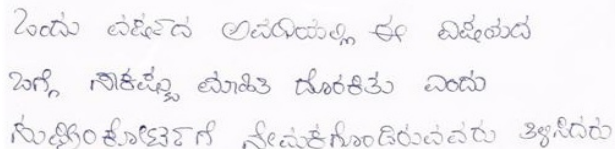


Fig. 3. Original handwritten kannada document image

proposed approach. The comparison of the proposed method with few existing methods is shown in the table 1. An example of the original image, line segmented image and character segmented image is shown in figure 3,4 and 5 respectively.

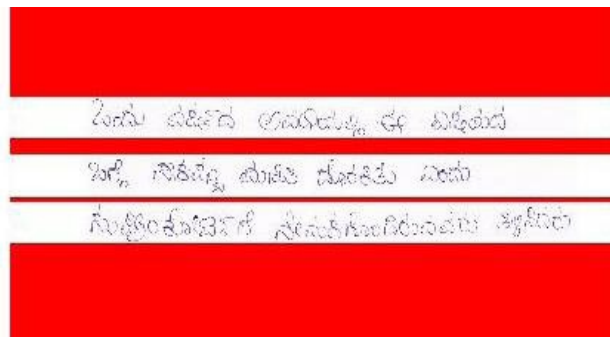


Fig. 4. Line segmented image



Fig. 5. Character segmentation

TABLE I. COMPARISON OF THE METHOD WITH THE EXISTING METHODS

Author	Method	Dataset size	Accuracy
Aradhya et al.,[15]	Component extension technique	250	Not specified
Alaei et al.,[14]	Potential Piece-wise Separation Line technique	204	94.98%
Mamatha et al [16]	Hough Transform	100	91%
Proposed Method	Adaptive window sizing and histogram of oriented gradient	200	94%

The performance of the histogram of oriented gradient descriptor was also evaluated on a test dataset consisting of 18800 samples. The recognition accuracy of vowels and consonants are shown in table 2 and 3 respectively. Table 4 indicates the comparison of the proposed method with the existing methods.

TABLE II. PERFORMANCE OF THE METHOD ON VOWELS

Vowels in sequence as per Fig.1	Recognition Accuracy (%)	Vowels in sequence as per Fig.1	Recognition Accuracy (%)
V1	93.75	V8	93.75
V2	95	V9	96.25
V3	96.25	V10	97.5
V4	96.25	V11	96.25
V5	97.5	V12	96.25
V6	95	V13	97.5
V7	97.5		

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TABLE III. PERFORMANCE OF THE METHOD ON CONSONANTS

Consonants in sequence as per Fig.2	Recognition Accuracy (%)	Consonants in sequence as per Fig.2	Recognition Accuracy (%)
C1	95	C18	97.5
C2	96.25	C19	93.75
C3	96.25	C20	96.25
C4	97.5	C21	97.5
C5	95	C22	96.25
C6	97.5	C23	96.25
C7	93.75	C24	97.5
C8	96.25	C25	95
C9	97.5	C26	96.25
C10	96.25	C27	96.25
C11	96.25	C28	97.5
C12	97.5	C29	95
C13	95	C30	97.5
C14	96.25	C31	97.5
C15	96.25	C32	96.25
C16	97.5	C33	97.5
C17	95	C34	97.5

TABLE IV. RESULT COMPARISON

Authors	No. of samples in data set	Feature extraction method	Accuracy (%)
Rajashekhar aradhya S V et.al[17]	1000	Vertical projection distance with zoning	93
R Sanjeev Kunte et al[18]	2500	Wavelet	92.3
V N Manjunath Aaradhya et al[19]	2000	Radon features	91.2
Dinesh Aacharya U et. al[20]	500	Structural features	90.5
Proposed	18,800	HOG	95.02

V. CONCLUSION

A new relevance feedback based approach for character segmentation and recognition in the handwritten kannada documents is proposed in this paper. The method was tested against a standard dataset called KHTD. We have achieved a segmentation accuracy of 94% and a recognition accuracy of 95.02% .The results are compared with the existing methods and found to be promising. However, the method does not address the segmentation of the touching characters in the document.

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Intelligent Accreditation System: A Survey of the Issues, Challenges, and Solution

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Abstract—International educational institutes are aiming to be accredited by local and international accreditation agencies such as the Association to Advance Collegiate Schools of Business (AACSB), Accreditation Board for Engineering and Technology (ABET) and so forth to be recognized by stakeholders. The institutes are striving to meet the stakeholders' expectations by integrating quality in all standards of educational practices and guarantee a continuous improvement. This study has acknowledged the principal barriers that need to be addressed and resolved such as collection & population of data, time constrains, compensation and lack of guidance and expertise. A web-based survey was conducted to identify the obstacles and the respondents' expectations in the optimization process for accreditation. This research proposes an Intelligent Web-Based Accreditation System (IWBAS) that addresses the above issues and streamline the accreditation process.

Keywords—Challenges of Accreditation Process; Intelligent Accreditation System

I. INTRODUCTION

The main responsibilities of accreditation agencies are then to establish the standards and procedures required for academic assessment and quality assurance as well as to provide training for faculty and staff members involved in the accreditation process. In this respect, each agency has created its own set of standards for the major areas of operation in education. The main areas of the standards in higher education include mission, goals, and objectives; program administration; management of the program; quality assurance; learning and teaching; student administration and support services; educational resources, facilities, and equipment; financial planning and administration; employment process; research; and relationships with the community.

During the accreditation process, all the standards are required to be addressed at several levels of detail. In specific, each standard is broken down into a number of sub-standards dealing with the requirements within each of its major areas. For example, the learning and teaching standard could be divided into a number of sub-standards that address the learning outcomes; students' learning and assessment; and quality of teaching. To be accredited, institutions have to provide a list of the good practices that are carried out within each of the sub-standards supported with sufficient evidences on the implementation of these practices.

In this regard, accreditation is asserted an effective way to assess and improve the quality of education [1]. Academic program accreditation, for example, is emphasized as an effective quality assurance mechanism that could serve a broad range of constituencies from the perspective of the profession and society. The growing interest in a sustainable improvement in the academic programs, and not just in meeting the minimal educational requirements, is a current positive trend that will help to guarantee successful educational systems. Accreditation should ultimately contribute to assure that graduates will have the knowledge and skills required to practice the profession successfully and to be of service to their societies [2]. Researchers [3] believe that the growing emphasis on accountability, assurance of learning, and the rising concern about the quality of education shed light on the significance of the accreditation process at both program and institutional levels. They stress that professional workforce is becoming internationally mobile and higher education institutions are expected to compete for participating through their graduates and research in the global economy.

Despite the positive expectations and inevitability of the quality affirming process, it still includes some barriers that need to be addressed and resolved should we require capturing all its potentialities. The nucleus of the whole process is to collect a vast amount of data and commentary necessary for fulfilling the standards required for the accreditation. Institutes' quality units are composed mainly of faculty members who are most likely not experienced in the accreditation process. In this respect, Al-Yafi [4] discussed such a lack of guidance and expertise as the principal obstacle to many institutions not to complete their evaluation reports. Faculty members are also in doubt about the accreditation ultimate outcome and confused about its relative priority within their typical list of duties including teaching and research. In sum, without adequate expertise or obvious motive, faculty members are constantly under pressure to collect an enormous amount of data, analyze it, identify areas of improvements, provide evidence, and finally write reports, all besides their normal duties.

This study discusses the common barriers of the accreditation process, focusing on the key problems of collecting and presenting data. This discussion is based on the viewpoint of faculty and staff members who have been involved in the accreditation process long enough to judge its complications. The study analyses the factors that cause the

complications and confirms the necessity of an Intelligent Web-Based Accreditation System (IWBAS) to overcome the challenges.

This paper is divided into six sections. After this introduction, Section 2 includes a brief literature review. Section 3 introduces the methodology adopted in this research. Section 4 presents the main challenges that are embedded in most of the accreditation projects and how a business process might address these challenges. In Section 5, a prototype of the IWBAS is presented. Finally, Section 6 discusses and concludes the research.

II. LITERATURE REVIEW

Brodie et al, [5] emphasizes the requirements of the academic institutes to provide a robust evaluation of the quality of their degree programs and to benchmark that quality internationally. The study addresses multiple methods that may be used to evaluate courses and programs including, surveys, grades analysis, and progression data. It also compares typical examples of some approaches to study the robustness of the link between the data collected and the evaluative judgments. The main concern seems to be in collecting and analyzing the data required for the evaluation.

The work has been [6] presented a comparison of two programs based on different accreditation criteria. The comparison is based on senior students' exit surveys and feedback obtained from faculty who taught the students of both programs. The research claims that graduates of the program following the outcome-based criteria have acquired better skills as compared to the other program following input-based guidelines. Once again, the study is based on collecting and analyzing data required for the comparison.

Gonge and Ghatol [14] assert the necessity of the classification of the educational accreditation into three groups: primary educational accreditation, secondary educational accreditation, and higher educational accreditation. They also emphasize accreditation as a continuous process that should meet specific standards of quality required for each level of education. In other words, it is a continuous process of collecting and analyzing data for evaluation purposes. However, their research focuses on the art of teaching in the learning process as an essential part of the educational accreditation, quality, and assessment. This can be contrasted with paper-based accreditation processes that highlight the importance of collecting and analyzing data.

Abou-Zeid and Taha [8] address the requirements of different international accreditation agencies such as the Saudi National Commission for Academic Accreditation and Assessment (NCAAA) and the Accreditation Board for Engineering and Technology (ABET). They discuss the challenges that arise during the preparation of the accreditation documents, including the inadequacy of knowledge of the accreditation needs and requirements. In their results, the challenges faced during the accreditation process include the inability to properly prepare required forms and documents, lack of faculty commitment to the accreditation process, high faculty turnover, and lack of proper support from higher administration. They notice that the differences between the

programs and institutes accreditation are not the problem; the readiness of the program and the institution itself is proving to be the catalyst for the accreditation process. This primary outcome by Abou-Zeid et al, [8] is further discussed in this paper to highlight the risk of the added workload on staff members that may lower the priority of teaching and research for filling the accreditation forms and conducting the surveys required. There are similarities in Abou-Zeid et al, findings and the work presented in this paper, especially the lack of belief in the outcome of the accreditation of the institution, which may hinder the accreditation process immensely.

III. METHODOLOGY

This research utilizes a descriptive survey to investigate the hidden obstacles in respect of collecting vast data and convert it into the needed format for the different accreditation projects. A quantitative data was collected through a Web-based survey and fed into a Statistical Package for Social Sciences (SPSS V.13) to analyze the relationships among the different responses. The survey's objectives were to attempt to answer the following questions:

- How have you participated in the accreditation process?
- How has your experience been during the completion of the accreditation requirements?
- What are the obstacles that arise during the process?
- What is your opinion about the solutions including the streamlining of the whole process through automation?

On February 9, 2015, a questionnaire was posted online along with a note that provided general information on the nature and importance of the study and the significance of the contributions. Participants were assured of the confidentiality, of their responses and promised a summary of the study results if they so desired. By February 28, 2015, the researchers had received 121 responses which were analyzed using SPSS statistical software.

Surveys are used in many areas of human activity, such as analyzing the attitudes and perceptions of people towards any phenomena [7, 9]. A survey provides a quantitative description of a sample of a population could enable the researcher to generalize the findings from this sample.

The Web-based surveys are used extensively for both scientific and non-scientific purposes, because of their ability to describe a large population in terms of a broad range of characteristics, attitudes, and behaviors. Many researchers [10] have recently used Web-based surveys as an appropriate method to collect data from Internet users due to its cost effectiveness, rapid turnaround, high response volume, and ability to cover a large geographical area. Many researchers utilize Web-based surveys for several other reasons including the ability to skip questions that are irrelevant based on previous responses, more time for questions that need extra time to respond, the elimination of interaction bias, and the manner in which they facilitate the collection of a large amount of data in a relatively short period of time. Due to the electronic submission of the responses, surveying cost is relatively small with a much fewer data entry requirement.

Surveys may also be distributed to the respondents by hand or in a group to reach respondents who do not have frequent access to the Web and could offer necessary explanations concerning the questions, if required.

In contrast, the Web-based surveys still have their own drawbacks. As the respondents will be on their own, Web-based survey's questions must be carefully chosen and pilot-tested. Self-selection and unequal opportunity are significant limitations of this sampling procedure [10]. Finally, in a Web-based survey, only people who have access to the Web could participate, but this was entirely appropriate in this study.

In this research, the questionnaire is developed in-house and includes ten main questions, seven of which give the respondents the chance to select more than one answer and even allow them to add their own comments as necessary. In addition, a separate question is dedicated to allow the respondents to add their own final comments and suggestions (see Appendix A). The questions, though limited in number, could have enabled collecting valuable data about the whole accreditation experience, however, the analysis below will focus on the problems and the proposed IWBAS as a solution. The results of the other aspects of the questionnaire will be discussed in a further research.

IV. ACCREDITATION: CHALLENGES AND SOLUTION

This study addresses some of the complications that are encountered throughout the accreditation projects, including NCAAA, AACSB, and ABET, from the viewpoint of staff members who have been deeply involved in implementing their procedures long enough to be able to judge their pitfalls. Some of these complications, as described below, are unnecessary and absolutely not inevitable by automating the process through implementing and using an intelligent web-based accreditation system.

In their responses to the overall experience of the different accreditation projects, 63% of the respondents agree that the accreditation process is an unsatisfactory or bad experience with limited expected influence on their teaching and research activities. This result confirms the initial claim that many staff members might not believe in the accreditation process as a quality assurance mechanism and they are in doubt of its outcomes on the overall educational systems. There is a high risk that accreditation procedures are then implemented haphazardly just to comply with the regulations, without a clear understanding of their pedagogical functions. It could be argued that for some staff members, implementing the regulations has turned to be a goal in itself rather than just a way of achieving quality.

Regarding the obstacles to complete the accreditation requirements, most of the respondents have agreed that the lack of expertise and the lack of time are the main problems. It has to be stated that many of the accreditation mechanisms and procedures cannot be understood through the simple guidance attached to the reports to fill. Even if the guidance is clear, there will always be a huge set of unanswered questions. For example, how to select acceptable and consistent local and global Key Performance Indicators (KPIs)? How could benchmarking get along with the selected KPIs? What are the

appropriate assessment methods that would enable the Assurance of Learning (AOL)? How are busy staff members supposed to analyze the grades of their large number of students, for each course they teach, with respect to each learning outcome? How could a course report on the same subject with different groups be written?

Surprisingly, the same claim about the lack of expertise was discussed by researchers [13] and after 10 years now it is still unsolved. A comprehensive survey that adopted personal interviews about the NCAAA also presented the paradox of the substantial deficiencies that are often found in the submission of the accreditation documents, though prepared by experienced PhD-holder staff members [Ibrahim et al. 2016]. Surely there is nothing in these deficiencies related to the lack of skills or knowledge; it is all about an assumed expertise that does not exist. The previous research [2, 3] has suggested that the implementation and evaluation of the quality assurance process could be monitored by offering proper training to the active committees. Otherwise, it will be tough for faculties to follow the accreditation process while are not equipped with the required specialized experience.

Similarly, about 59% of the respondents agreed on the incomprehensibility of the accreditation process. In addition, about 40% confirmed that the inconsistency and conflicting feedback about the reports from the different stakeholders, reviewers, accreditation commissions is another source of confusion that delay the completion and submission of the reports. It is argued here that many different answers could be obtained for each question about the process from each different stakeholder. For example, with respect to the course report question presented above, three different viewpoints have been received: (1) statistical data of all the groups should be summarized in a single report; (2) statistical data in each group should be presented and compared to the other groups in the same report; and (3) each group has to have a separate report. Researchers of this paper have also witnessed conflicting feedbacks about the accreditation submissions from an external consultant, quality vice-deanship in their college, quality deanship in their university, and the accreditation commission. Such feedback is absolutely confusing for the committees who struggle to satisfy each of those revising authorities through repeating the work again and again. A single, consistent report would definitely be much more beneficial and take much less time to implement should an efficient communication between the quality assurance and accreditation units be established.

Regarding the main data problems, redundancy, unavailability, and the difficulty of the analysis required are the main problems confirmed by 87% of the respondents. Redundancy with respect to all the aspects of the accreditation process, including efforts, data, and communication is surely a serious cause of the submission delay and time waste. For example, considerable efforts are duplicated by all the academic programs in the same college to get and report data that are common, such as, university and college level data. Another form of redundancy is that some data are required to be repeated in different places in the forms. For example, just consider the redundancy of the learning outcomes, objectives, and assessment methods in the program specification, course

specifications, and course reports of the same academic program. Resubmission redundancy is also expected each time the documents and requirements are changed as usually happens with all accreditation commissions. Although a certain considerable bulk of the data would usually be the same in the old and new templates, significant work would be required to refill the new documents.

In the light of the barriers and the outcome of the survey with the concerned subjects, this study recommends the implementation of an IWBAS. First, the time and efforts of gathering, compiling, analyzing, managing, and presenting quantitative data in effective ways, not to mention the training and professional development required to accomplish these tasks, could exceed the time and efforts invested in designing and implementing an IWBAS. Getting accredited is not a final destination; it is indeed just a beginning of an endless continuous journey of quality fostering and maintaining process. IWBAS then is the platform that would enable and facilitate such an ongoing communication between the academic institutes and the accreditation commissions.

Second, the correct information, requirements, suggestions, guidelines, feedback, and recommendations may be available through IWBAS to provide a more robust, fair, and transparent system. The availability of a centralized online information system could streamline the whole accreditation process and speed up handling the requests than at present. The IWBAS may be accessed by the quality units and deanships to review formal and final submitted documents from colleges and departments to make final accreditation decisions. Through IWBAS, commissions can disseminate information and guidance, explain the requirements for successful accreditation, and help institutes prepare for the formal accreditation submission. In this context, IWBAS could be considered as a liaison between the institutions and the accreditation institutes that enables effective communication with faculty and staff to help them integrate and align existing documents into a coherent accreditation submission.

Third, IWBAS could work as an interactive whiteboard system on which all the comments and feedback are communicated. This availability of centralized reviews may help other programs to avoid obvious mistakes on the fly. Concerned committee members may contribute virtually in the process while not present physically. Administrators may also check the progress, assign additional resources, and make comments to speed up the process. In the absence of the interactive reviews, academic programs or concerned personal may compile data or integrate information into the reports that may not be required by the accreditation reviewers. This unnecessary excessive work could be avoided by using IWBAS that may bank all the comments, reviews, feedbacks, explanations, and responses about every submission at one place and communicate them intelligently as required.

Fourth, the serious problems of redundancy are automatically solved by the centralized database of the system. The system could then facilitate the repetitive submissions of the accreditation documents to the same commission in response of the sequential revisions or changing templates. The submissions to different local and global accreditation

institutes are no more that complicated as most of the data required for these different accreditations are quite similar and the system could enable producing the data in any format required.

The IWBAS payoff may surpass the expectations of its originators. For example, internationalization of higher education requires curriculum communication between learning opportunities and/or outcomes achieved for other users like students, prospective employers, and administrative institutions [11]. To achieve such a centralized integration, IWBAS may serve as a central reporting mechanism and a generator of publicly available trend data about universities and their colleges, departments, and programs. It may automatically compile new measures and statistics from databases and disseminate trend data to all of the institutes involved. Each institute could benefit from publicly available trend data and guidance on how to compile, interpret, present, and use the data in many ways including benchmarking. The availability of centralized data at IWBAS could eliminate extra data-gathering practices that do not address strategic concerns or are not required for reviewers.

V. AN INTELLIGENT WEB-BASED ACCREDITATION SYSTEM

As discussed earlier, IWBAS could assist in overcoming some common complications of the accreditation processes, streamlining its detailed procedures, facilitating the submission to different accreditation commissions, and helping to ensure the quality aimed in the first place. This research discusses the conceptual ideas of the system. Detailed analysis or design of the system is out of the scope of this study and may be carried out in a future research. The IWBAS is supposed to be a web-based or cloud application system that is developed and maintained globally by a non-profit or for-profit organization. Although the system has to be based on a comprehensive survey of the requirements of the most common accreditation institutes, it should not be restricted to these requirements. It has to be built as a quality-oriented system that satisfies excellence through accreditation. In other words, the system should be built with a mission that "To facilitate accreditation as a quality assurance mechanism, not as a goal." With respect to this mission, the system is supposed to provide the decision makers and stakeholders of the registered institutions with customizable, interactive dashboards of the historical trends and current status of the main metrics and KPIs required. In addition, this type of interface could facilitate measuring and tracking all the accreditation requirements that are declared, according to the questionnaire in this study, incomprehensible and difficult to compute. For example, a course tracking dashboard should provide information about the students' attendance; grades analysis and progression of each student and the overall students; assurance of learning of each learning outcome; and students feedback on each class or topic; all with a detailed comparison with all the other groups of the same course in the current semester and the previous semesters and automated benchmarking with local and global similar courses (see Fig. 1). Staff members will not have to worry much about the data and analysis of this dummy work anymore; they will have more time to consider teaching and course improvements.

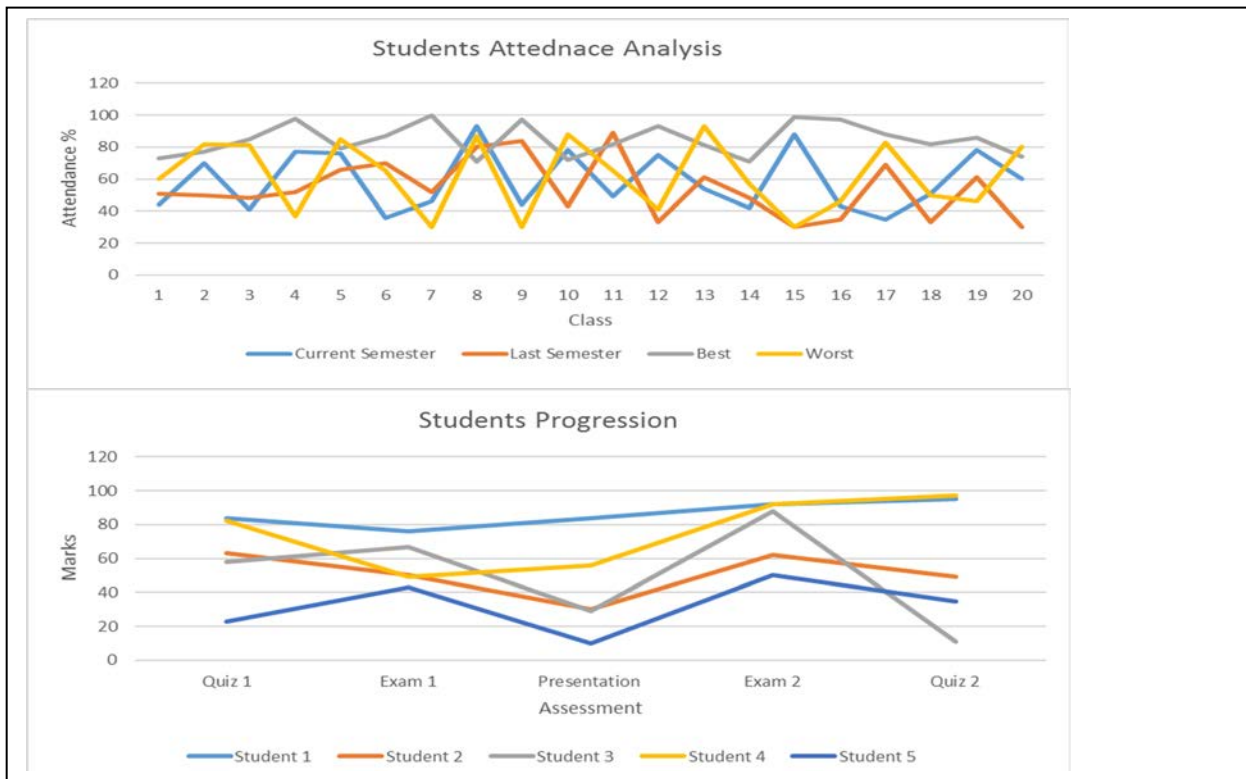


Fig. 1. Sample Course Dashboard

The system has to have a specific predefined workflow for commencing an accreditation process. The workflow starts with a step of filling a registration form including data about the institution and the type of accreditation it applies. The registration form is surely different if a university, college, department, or program is applying. An academic program application form is supposed to be automatically linked to its corresponding department application, which is linked to its corresponding college form, which is finally linked to its university (see Fig. 2). In this way, data could be easily propagated back and forth among these levels of entities within a particular institution with a substantial elimination of redundancy and a significant saving of time.

The application forms should only include the essential information about the applicants. A full access to material, forms, and regulations corresponding to the accreditation process is granted once the registration step is completed. The access would allow the applicant to traverse the material, fill the forms, get online assistance, and receive feedback.

With respect to filling the accreditation forms, that is, compiling and entering the data in the centralized database, a uniquely different approach should be adopted. Accreditation forms should be just seen as reports that are required by the agencies about the candidates. Even the format and data included in these forms are always a subject to revision by the accreditation institutes. For example, three different templates have been introduced by the Saudi NCAAA in the last 5 years. From that perspective, data must be organized and entered according to their relevance rather than the format of the reports required.

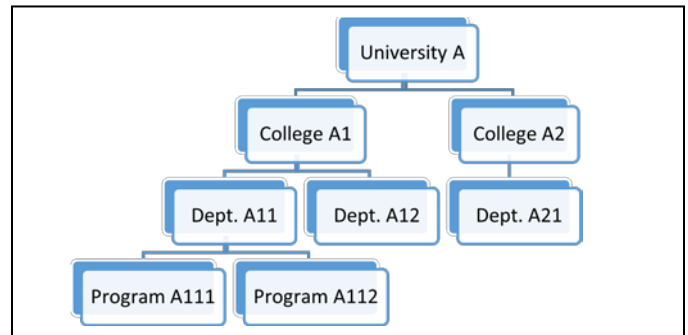


Fig. 2. Institutional Entities Registered for Accreditation

This would actually help further reduce the data redundancy as each piece of information would be entered only once, according to its relevance and can then be automatically propagated in all the reports in which it is required. Researchers of this article themselves have encountered the drafting problem twice. Once a new curriculum for one of our B.Sc. programs has been introduced, a comprehensive revision of the department's accreditation forms has been undertaken, despite most of the data has not been changed. This incident occurred again when the NCAAA commission has introduced new forms in 2013. Transferring the data from the old templates to the new ones was a tremendously tedious work while most of the data was again unchanged. The tireless work of re-entering and updating the data could be absolutely avoided by adopting the IWBAS.

As stressed above, automated data integration, propagation, and manipulation is essential for the ultimate utilization of any

IWBAS. For example, data from academic departments, academic programs, and courses offered in each of these programs could be manipulated with consistent propagation and highest integration. Fig. 3, for example, represents how the department vision, mission, and goals have to be reflected in its all programs' mission, goals, objectives, and learning outcomes, which in turn have to be considered in each of the courses the programs offer. And all of these have to be initially in a full compliance with the department's college and university vision and mission. Things should get more complicated when considering programs with different tracks, colleges with interdisciplinary departments, and universities including similar programs. Ibrahim et al, [10] stated that considerable comments and feedback are usually received with respect to the consistency among the objectives and goals of the departments, programs, and courses. It can be argued here that the absence of a platform that can present and organize these levels of details is an essential reason for the inconsistency problem rather than the staff members imprecision of setting the links required among those subjects.

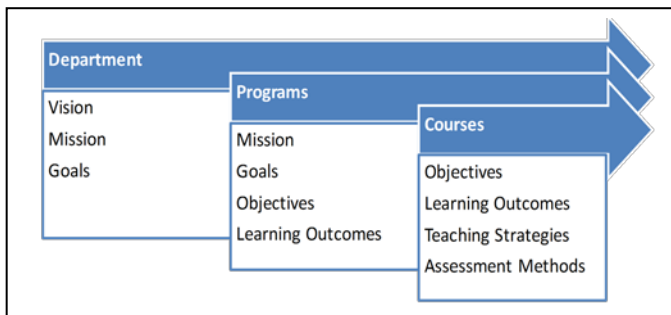


Fig. 3. Data Manipulation and Integration of Departments, Programs, and Courses

From the implementation perspective, in the case of the accreditation of an academic bachelor program of a specific department, for example, an initial web-form for the courses offered in the program must include elementary information about each course, such as, course name, level, and description. Once the courses web-form is submitted, the data must be manipulated in a way of creating, for example, a course specification web-form for each identified course. Data such as the university name, college, department, course code, course name, and so forth must be included automatically in the newly-created course specification forms. Some of this information could have a read-only restriction, while others must be left open for editing. Editing the course code, for example, must be propagated back to the courses form exactly as the data updated in the courses form propagated forth to the other forms, including the course specifications and course reports, requiring this data.

Data requirement and data granularity are critical issues of the system. Both of the obligation and granularity could be simply identified through a comprehensive scan of all the data requirements by the different accreditation systems. Data requirements are identified by the fields of the forms to be filled, while the granularity by the last field that requires this information in the different accreditation systems. For example, while the course name could be considered granular as any piece of it is not necessary in any form, the course

learning outcomes are not. Each learning outcome is needed to be linked to teaching strategies and evaluation methods. From the perspective of database concepts, learning outcomes have a Many-to-Many relationship with both teaching strategies and assessment methods. From the implementation perspective, it is so expected in the course specification web-form to include two fields for each learning outcome. The first field is represented by a Combo-Box from which the appropriate outcome "verb" is selected. The second field will be a Textbox to enable writing down the learning outcome. Two more Multiple-Selection Combo-Boxes have to be added to identify the different teaching strategies and assessment methods that correspond to the learning outcome (see Fig. 4).

The system intelligence is supposed to help validating the fields, recommending the outcome verbs, associating the right verb with the right teaching and assessment methods, and checking the relevance of all of them. Such relevancy is usually confirmed as a serious problem for inexperienced staff members, which hinders writing a precise course specification and reports. In that respect, the system can be used as a learning mechanism for the appropriate manipulation of course specification in addition to its primary accreditation and quality purposes. Learning capabilities could be incorporated in all the parts of the system to provide faculty members with the quality expertise currently required for the academic job.

Learning Outcome Verb :	Learning Outcome:	Teaching Strategies:	Assesment Methods:
Describe	<input type="text"/>	Lectures	Exams
List			
Name			
Define			

Fig. 4. Learning Outcomes Web-Form

The system has to include an advanced *Query and Reporting Tool (QRT)*. The QRT must include predefined queries that initiate all types of forms and reports required by the different accreditation agencies from the data entered and saved in the system. Also, the QRT should enable querying the database for any other ad-hoc reports. The QRT retrieves the data required for a particular report/form and organizes the data in any particular predefined format. For example, short, full, or ad-hoc course specification reports can be extracted and developed from the database as long as all the data required about courses are saved without the need of any extra efforts. Changing the accreditation templates is not a problem anymore.

As this study is intended to shed light on the concepts and applicability of IWBAS, many capabilities that could be added to the system are omitted here. Development of unified standard exams of similar courses around the globe, comparisons of similar academic programs, and benchmarking with suggested national, regional, and international institutions are among these additional functionalities. Online automated assistance and guidance should be a part of the intelligence system. Sophisticated dashboards for each level of administration in the educational institutes, online data analysis, automated feedback, students' progress reports, staff members' evaluations, etc. are highly applicable with the data

available. Online communication among similar departments or colleges could be also permitted to enable new applicants to get benefited from the experience, expertise, and material of the institutions ahead.

VI. CONCLUSION

The rationale of going through the rigorous process of accreditation to make sure that high quality in educational processes, continuous improvement efforts, assurance of student learning, emphasis on ethics and code of conduct, faculty and student engagement, and mission-driven management processes are firmly ingrained in academia.

An accreditation is a quality-affirming process that includes the creation of accreditation standards and awarding of accredited status. Contrary, it is a laborious task if not planned appropriately from the start-up. In this respect, online survey was conducted with the people who are involved in the accreditation task. The rationale of the survey is to learn the existing challenges and hardship and expected solution.

The collected data showed that several problems exist in the current practices of data gathering and formulating reports for the accreditation process, such as, redundancy, decentralization of information, lack of expertise, the inherent difficulty in reaching the quality assurance personals, conflicting feedbacks, and insufficiency of time allocated for data gathering and writing reports. These concerns represent real challenges and issues and required an intelligent tool to replace the tedious labor and streamline the whole process.

This result revealed that many respondents do not believe in the accreditation process as a quality assurance mechanism and are in doubt of its outcomes on the overall educational systems. They have stated that lack of expertise and lack of time are the main problems. On the same note, they mentioned that accreditation mechanisms and procedures cannot be understood and followed through the inaccurate guidance provided to populate the reports and forms. It has been noted from the responses that many agreed on the complexity, redundancy, unavailability, inconsistency and conflicting feedback on the accreditation process.

This study suggested the that Intelligent Web-Based Accreditation Systems (IWBAS) to be implemented to standardize the whole accreditation process, eliminate redundancy and wastage of time, avoid repetition of general information, and provide the availability of 24/7 communication channels. IWBAS then is the platform that would enable and facilitate such an ongoing communication between the academic institutes and the accreditation commissions. The proposed system could generate accurate information, suggestions, guidelines, feedback, and recommendations to facilitate the robust, fair, and transparent process. The proposed intelligent system could work as an interactive whiteboard that host comments and feedback from all stakeholders. The availability of centralized comments and feedback may help in avoiding common errors. The whiteboard could be accessed by all concerns personnel through Internet to observe latest, check the progress, available resources and comments to speed up the process. Contrarily the concerned person may compile unnecessary data or integrate information

into the reports that may not be helpful or required by the accreditation process. The system would eliminate the redundancy through the centralized database of the system. This study has presented and discussed the conceptual framework of the proposed tool while its implementation will be presented in a future study.

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APPENDIX A

Online Survey Questions

As presented above, data are collected through an online survey, implemented on Google Forms, to be found at [12]:

<https://docs.google.com/forms/d/15-2Nhil0UfT9J1Xpo677BE73M7jgsNeMW15sYRyzZgE/viewform>

In addition to the introductory questions, there are three main groups of questions that address the overall experience, challenges and problems, and solutions.

There is a final question that allows the respondents to add their own comments. Table 1 includes the questionnaire.

No.	Question
Introductory Questions	
	Institute: College/Department (if applicable): Position: Position in the accreditation committee: Accreditation project(s) you have been working on: Period of time you have been involved in the accreditation projects (in months):
I. Overall Experience	
1.	How did you learn about the accreditation procedures? Internet - Colleagues - Place of work – Accreditation agencies – Trial and error – Others (Checkboxes)
2.	How was your overall accreditation process experience like? Excellent - Satisfactory – Neutral – Unsatisfactory – So bad
3.	The accreditation process has a positive significant influence on my teaching, research, and other duties. Strongly Agree – Agree – Neutral – Disagree – Strongly Disagree
II. Challenges and Problems	
4.	What are the main obstacles to complete the accreditation requirements? Lack of expertise – Lack of time –Lack of administrative support – Lack of support from accreditation agencies – Others (Checkboxes)
5.	What are the main challenges/problems in the accreditation process? High teaching load – Incomprehensible accreditation process – Staff resistance – Inconsistent or conflicting feedback about the reports from different stakeholders – Others (Checkboxes)
6.	What are the main problems in the data required for completing the accreditation files? Unavailability – Redundancy– Incomprehensibility – Size of data – Difficulty of the analysis required – Others (Checkboxes)
III. Solutions	
7.	How do you think the administrative obstacles could be solved? Administrative support – Administrative accountability – Ownership of the process – Hiring accreditation qualified staff – Others (Checkboxes)
8.	How do you think the other challenges and problems could be solved? Low teaching load – Special dedicated accreditation committees – Training – Financial compensation – Others (Checkboxes)
9.	If you are the head of the accreditation commissions, how do you think the accreditation process could be handled better? Provide online guidance – Adopting a more realistic approach – Avoid redundancy – Automation – Others (Checkboxes)
10.	An Intelligent Web-Based Accreditation System could facilitate the accreditation process and solve its data problem? Strongly Agree – Agree – Neutral – Disagree – Strongly Disagree
Suggestions and Recommendations	
	Any final comments or suggestions:

Online Survey Questions

Multi-Objective Optimization Algorithm to the Analyses of Diabetes Disease Diagnosis

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Abstract—There is huge amount of data available in health industry which is found difficult in handling, hence mining of data is necessary to innovate the hidden patterns and their relevant features. Recently, many researchers have devoted to the study of using data mining on disease diagnosis. Mining biomedical data is one of the predominant research area where evolutionary algorithms and clustering techniques are emphasized in diabetes disease diagnosis. Therefore, this research focuses on application of evolution clustering multi-objective optimization algorithm (ECMO) to analyze the data of patients suffering from diabetes disease. The main objective of this work is to maximize the prediction accuracy of cluster and computation efficiency along with minimum cost for data clustering. The experimental results prove that this application has attained maximum accuracy for dataset of Pima Indians Diabetes from UCI repository. In this way, by analyzing the three objectives, ECMO could achieve best Pareto fronts.

Keywords—Clustering; Genetic Algorithm; Multi-objective Optimization; ECMO; Diabetes Disease

I. INTRODUCTION

Among numerous diseases, health department of India have identified that Diabetes disease lists top for cause of death domestically. In the recent years, it is inferred that this problem is growing at an alarm rate with massive patient data [1]. In this view, this work adopts Evolutionary Clustering Multi-objective Optimization Algorithm (ECMO) which was the extended work of NL-MOGA for analyzing diabetes disease datasets [2]. There are some global optimization tools such as genetic algorithms uses validity measures for evaluating clustering accuracy [3]. However, as no single validity measures works equally well for different datasets which could simultaneously produce high clustering accuracy. Some recent studies have posed the problem of data clustering as a multi-objective optimization problem in which several cluster validity measures are optimized concurrently to obtain the tradeoff clustering solutions.

Depending on the dataset properties and its inherent clustering structure, different cluster validity measures perform differently [4]. Therefore, it is important to find the best validity measures that could be instantaneously attain good clustering results. In order to evaluate the quality of the clustering, external measures like Jaccard-index, Minkowski-index, Rand-index, and so on can be utilized to optimize the multi-objective problem [5]. This measure are used to identify the intra-cluster similarity or compactness and the inter-cluster separation. In this paper, the cluster compactness and the

separation is evaluated using Rand- index. This index measures both cohesion and separation of clusters using distance measures between the points in the closest cluster to the points in the same cluster [6]. The Rand-index for the point xi is calculated as

$$R(T, C) = \frac{a+d}{a+b+c+d} \quad (1)$$

Whereas, T is the true cluster of the selected dataset for C the clustering result returned by some algorithm. The points a, b, c, and d are the objects belonging to T and C. The value close to +1 indicates a good clustering. Hence, the best cluster accuracy can be calculated using this index. Thus the inaccuracy could be the values nearing -1[7]. However, in clinical diagnosis, the inaccuracy could be in diagnosing false positivity and negativity.

The false positive like inaccurate-positive depicts the percentage of patients diagnosed to have no disease while in reality they have. Inaccurate-negative implies that the ratio of patients being diagnosed with disease but is diagnosed contrarily. In general, false negative results could cause greater impact than false positive results for both the doctors and the patients. At this juncture, the analysis of medical disease needs more concentration on the state of lower cost or false negative than the state of high cost or false positive.

Therefore by applying ECMO algorithm which uses data mining technology along with genetic algorithm that would help in analyzing the disease to produce high accuracy results by optimizing the low cost and high cost values. In this light, the accuracy and cost are the conflicted objectives. Hence, the optimum results could be achieved by setting minimum acceptable accuracy rate. On the premium that all the conditions were attained, the higher accuracy and low cost values could result better. The optimum values could be drawn from Pareto fronts. The rest of the paper is organized as follows. In Section II, a brief review of some past studied are presented. Then in Section III, methodology of ECMO is discussed in detail. Section IV shows the experimental results obtained from the study. Finally, conclusion and possible research issues are presented in Section V.

II. LITERATURE REVIEW

Sriparna et al. [8] proposed a multi-objective clustering technique to partition the data into appropriate clusters. This work aims to find total compactness of the partitioned clusters, symmetry of the clusters and the connectedness of the clusters. The algorithm uses Silhouette-index to measure the validity of

the clusters. Hector et al. [9] presented a technique to identify the main folds in the large datasets. Author summarized the original search space with Map-reduce architecture to identify the voronoi regions. Guang et al. [10] depicted generate-first-choose next method using upper bounds, lower bounds and inequality constraint engineering problem based on surrogate models. The algorithm failed to adopt weighted sum approach.

Lei et al. [11] devised clustering-ranking algorithm using a series of reference lines as cluster centroid. The solutions are ranked according. Anibran et al. [12] defined an interactive genetic algorithm based multi-objective approach that could simultaneously found clustering solution by evaluating the validity measures. The algorithm reduces fatigue of the decision maker by generating only important solutions from the current population. A massive on clustering based multi-objective genetic algorithm is presented in [13] and the author extended research by depicting an enhanced K-means Genetic algorithm for optimal clustering. The author overcomes the drawback of local optima with suitable dataset and also the algorithm fails in computational time. It is inferred that the algorithm produced more than the 90% accuracy for real life dataset. The author also adopted a neighborhood learning strategy for optimizing multi objective problems. This algorithm used k means Genetic algorithm to find the compactness of the clusters. It is noted that the algorithm could produce minimum index value for the maximum datasets. However, there is a need for proper feature selection for better, more optimal solution [14, 15]. Ruby et al. [16] suggested two methods for ranking of MOPs. This ranking methods were used to prune large data-sets of solution to small subset of good solution. Edward et al. [17] presented an approach by extracting the knowledge of conflicting interests like traceability and transparency to obtain the group of consensus data. Min Han et al. [18] considered mutual information based feature selection to enhance the searching capability of the data. Partha Pratim et al. [19] proposed high dimensional feature selection technique to preserve sample similarity using shared neighbor distance technique to reduce the outliers with a minimum computational complexity.

III. METHODOLOGY

This section address the issues specified in Section II by applying evolutionary clustering algorithm (ECMO) for MOPs. Primarily, ECMO generates uniform set of objects as the population. Then, the population is treated with three main procedures until the termination condition is satisfied. The three major operations are criterion learning algorithm (CLA), knowledge acquisition algorithm (KAA) and optimal cluster-ranking algorithm (RA). The ultimate goal of CLA is to perform global search based on the discovered criteria and then the knowledge is acquired through constant learning to dominance. While RA refine the process by grouping most relevant data with the help of ranking strategy.

A. Evolutionary Clustering Algorithm for Multi-objective Optimization

This research inherits ECMO which handles data by adopting criterion learning algorithm. The criterion for the particular objective was designed based on cluster location. The neighborhood data such as closest neighbor, farthest

neighbor and indirect neighbor were identified using knowledge acquisition algorithm. Hence, based on the dominance of individuals the data can be grouped and ranked using best knowledge ranking algorithm. The optimal Pareto fronts was achieved using balancing Pareto front algorithm that was capable of finding the best features the particular data set. Therefore, the fitness function for diabetes disease diagnosis using ECMO could be maximizing the cluster accuracy with minimum number of false negatives and false positives. It can be represented as follows:

$$f_1(x) = \text{Max} \sum_{i=1}^n |C(\text{origi}x_i) + C(\text{pred}x_i)| \quad (2)$$

$$f_2(x) = \text{Min} \sum_{i=1}^n (|\text{Inaccu}_{\text{negative}}|) \quad (3)$$

$$f_3(x) = \text{Min} \sum_{i=1}^n (|\text{Inaccu}_{\text{positive}}|) \quad (4)$$

Hence, by adopting the rules of knowledge acquisition algorithm true negatives and true positives objects can be identified. Maximum cluster accuracy could be achieved through best knowledge ranking algorithm of ECMO.

IV. EXPERIMENTAL STUDIES

To evaluate the performance and efficacy of the proposed algorithm ECMO, an unsupervised genetic algorithm is discussed in this section.

A. Data Set and Experimental Setting

The algorithm is tested Pima Indian Diabetes microarray datasets which are taken from UCI repository [20]. There are 768 records, out of which 268 cases are with diabetes disease and 500 cases are without diabetes with 376 records contain missing values. Pima Indian Diabetes microarray datasets contains 8 attributes with on class attribute. Table I contains the information about the dataset for the analysis. The algorithm were implemented in 7.6 and executed using Pentium with 2.99 GHZ CPU and 2 GB RAM. The operating system Microsoft Windows XP.

TABLE I. INFORMATION ABOUT ATTRIBUTS OF DIABETES DISEASE

No	Attributes	Domain
1	Age	continuous
2	No. of times of pregnancy	0,1,2
3	Diastolic Blood pressure	continuous
4	Plasma glucose concentraion	continuous
5	Triceps skin fold thickness	0,1
6	2-hrs serum insulin	continuous
7	Body mass index	0,1
8	Diabetes pedigree function	continuous
9	Class	Healthy/Sick

B. Testing Datasets and Performance Metrics

The experiment on the dataset was conducted on 90% of training dataset with 10% of test data. Testing has undergone 20 independent runs. The foremost aim of cluster validity indexes is to validate clustering solution. This index is useful in comparing the performance of the cluster. We adopted rand index (RI) to compare the performance of the algorithm with the selected diabetes datasets. The cluster accuracy, inaccurate positive and inaccurate negative for predicting diabetes disease is shown below:

$$\text{Rand}_{\text{clusAccu}} = \frac{\text{sick+healty}}{\text{sick+indirt}_{\text{nei}}+\text{fart}_{\text{nei}}+\text{healthy}} \quad (5)$$

$$Inaccu_{negative} = \frac{sick}{\sum_{i=1}^n C(sick-indirt_{nei}) + \sum_{i=1}^c C(healthy-far_{nei})} \quad (6)$$

$$Inaccu_{positive} = \frac{healthy}{\sum_{i=1}^n C(healthy-fart_{nei}) + \sum_{i=1}^c C(sick-indirt_{nei})} \quad (7)$$

After eliminating the missing values using mutation operator, the testing of data starts with phase training samples.

1) TEST: 90% of testing dataset (353 cases) and 10% training dataset (39 cases) of 392dataset.

In this test phase, 353 cases training sets with 39 cases of testing samples are considered. During each run, ECMO select different features from the original attributes and the clustering accuracy is recorded. The experiment was repeated 20 times and the results are recorded in Table II.

TABLE II. AVERAGE CLUSTER ACCURACY FOR 20 RUNS

Test Run	No. of Attributes Selected	Rand-Index (%)
1	5	97.65
2	7	96.34
3	8	98.21
4	4	98.49
5	6	99.92
6	8	97.01
7	7	97.57
8	7	98.53
9	6	99.01
10	7	98.76
11	4	96.32
12	5	99.67
13	6	99.46
14	8	98.99
15	8	98.52
16	7	99.21
17	4	99.05
18	3	98.47
19	3	99.38
20	6	98.25

It is inferred from the result that the average closer cluster accuracy is determined using rand index metric. The average clustering accuracy is 98.48%. The results of Pareto fronts was presented in Fig.1. shows the best cluster accuracies produced by the selected objectives. Blue color implies the healthy objects whereas pink and yellow color indicated inaccurate negative and inaccurate positive respectively. The evaluation metrics obtained by ECMO algorithm is recorded in Table III. The Fig. 2 Shows the best Pareto fronts obtained by the selected class variables for the single run. The selected from the Pareto fronts were mostly in the knee regions of the Pareto fronts.

It is noted that cluster prediction the algorithm could able to produce accurate cluster classification with low inaccurate positive and negative results. Table IV represents the impact of ECMO on inaccurate negative and positive results.

ECMO takes 20 iterations independently on diabetes dataset for its clustering process. It is praiseworthy that ECMO could form cluster along with good convergence and diversity as shown Fig.1. It is observed from Fig.2. ECMO can produce Pareto optimal solution for the selected objectives.

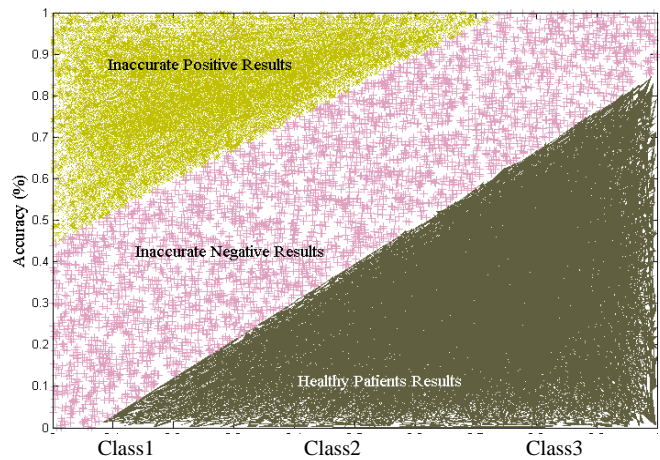


Fig. 1. Cluster Accuracies of the selected objectives

TABLE III. PERFORMANCE MATRICS OBTAINED BY RAND-INDEX

No	Attributes	Domain
1	No. of Attributes	8
2	$Inaccu_{negative}$	83.27%
3	$Inaccu_{positive}$	74.87%
4	$Rand_{ClusAccu}$	98.48%
5	No. of False Negative	5
6	No. of False Positive	10

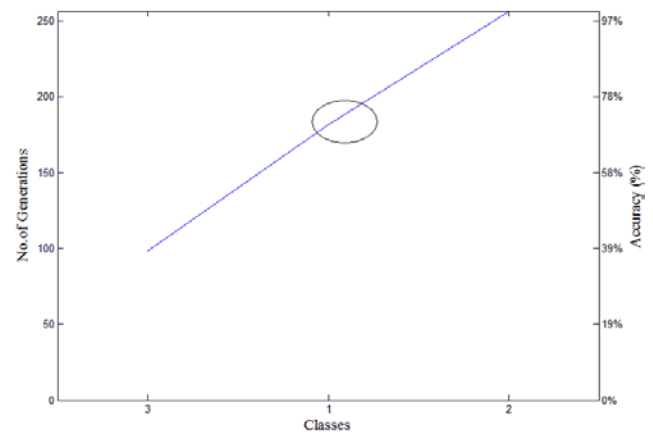


Fig. 2. Cluster Accuracies of the selected objectives

It can be identified from the Table III rand index value of the proposed algorithm is comparatively low than other algorithms except few. When the value of RI is equal to 1, the formation of cluster will be good. Hence, it is certain that ECMO generates better convergence and diversity. Experimental results substantiates that the algorithm ECMO, can identify appropriate features set using criterion and produces better clusters by utilizing the procuring the knowledge from the neighbors. The algorithm adopts neighborhood learning from the previous work and the NLMOGA procedure is extended to figure the closest-neighbor, farthest-neighbor and the indirect neighbor. Based on the outcomes of CLA and KAA, excellent clusters were ranked with more compact and less in diversity. The Table IV reveals that performance of ECMO on healthy, inaccurate positive and inaccurate negative results for diagnosis of diabetes disease.

TABLE IV. COMPARISON OF $Inaccu_{negative}$ AND $Inaccu_{positive}$ USING ECMO

Original \ Prediction		Prediction	
		Healthy	Sick
Healthy	Inaccurate Positive	305	48
	Inaccurate Negative	325	29
Sick	Inaccurate Positive	5	34
	Inaccurate Negative	10	29

Hence, it was inferred that the algorithm selected minimum five attributes and the maximum of eight attributes as its feature to process the objective function. It was also noted that the algorithm could able to produce maximum accuracy of 99.92% at the 5th iteration.

Total number of false negative and false positives was noted to very minimum. Therefore, the ECMO produced high cluster accuracy at minimum computation time. Henceforth, it was recorded that the algorithm ECMO produced maximum cluster accuracy for the healthy dataset of disease diabetes by minimizing the inaccurate positive and inaccurate negative results in minimum CPU running time that could reduce the cost substantially.

V. CONCLUSION

This research application on diagnosing diabetes disease using evolutionary clustering multi-objective algorithm which helps in analyzing the datasets found in Pima Indian Diabetes datasets of UCI repository. In this work, the best feature of the dataset was identified using selecting features (CL) of criterion learning algorithm.

The inaccurate positive and inaccurate negative neighbors were identified using knowledge acquisition algorithm. Hence, the algorithm could able to recognize more suitable healthy and sick objects while it possesses the similar dissimilar properties from the selected feature respectively.

ECMO shifts the objects position according to their relative proximity. Hence, the experimental results recorded the optimal solution with good Pareto fronts and high accuracy in healthy clustering. The algorithm could able to produce better cluster accuracy in identifying the inaccurate positive and negative results. Therefore, the reliability by satisfying the considered objectives. Also, algorithm can predict appropriate number of clusters for all the three objectives respectively. Much further work is needed to investigate the utility of having different and more objectives and to test the approach still more extensively, to investigate the utility of having different and more objectives, to hybrid ECMO with multi-objective Particle Swarm Optimization technique for high effectiveness, efficiency, and consistency and to enhance with heterogeneous data.

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Comparative Analysis of Energy Detection Spectrum Sensing of Cognitive Radio Under Wireless Environment Using SEAMCAT

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Abstract—In the recent years, the Cognitive Radio technology imposed itself as a good solution to enhance the utilization of unused spectrum and globalized the radio environment for different band users that utilize or require different techniques for transmission. In this paper, the energy detection spectrum sensing technique that is used to detect the presence of unknown deterministic signal is studied under the non-time dispersive fading environment using the Hata propagation model for picocell communication systems. The different aspects of non-time dispersive fading regions over energy detection spectrum sensing and impact of changing a detection threshold of the secondary user Cognitive Radio on interference at primary user for non-cooperative spectrum access have been studied in the terms of probability of interference. The entire Comparative Analysis of Spectrum Sensing in Cognitive radio has been carried out with the aid of SEAMCAT software platform.

Keywords—Cognitive Radio; Primary User; Secondary User; Detection Threshold; Interference Probability; Energy Detection; Desired/interfering/sensing received signal strength

I. INTRODUCTION

Cognitive radio is a futuristic technology that will delimit the congestion problem in a radio spectrum, and globalize the radio environment for different band users which use different techniques for transmission. In today's wireless communication scenario, the Radio Frequency spectrum is occupied for different purposes like cellular, television, military, emergency and satellite communication. The frequency spectrum used for cellular communication is getting over crowded with increasing number of subscribers and demand for high data rate text or video transmission, but the frequency spectrum in other wireless broadcast and communication is not utilised efficiently, example in Television broadcast band, some of frequency spectrum is vacant at some instant or for particular time. By FCC in 2002 reported that 70% of the spectrum at certain time and location is ideal, which shows that there is no scarcity of spectrum but underutilization of spectrum. The vacant spectrum could be used in cellular communication, means the spectrum is borrowed from Television Broadcast band to be used for cellular communication. A use of spectrum of licensed frequency bands when primary user is absent, will improve the efficiency of spectrum utilization [1].

By the mean of Cognitive Radio technology, the high data rate could be achieved for multimedia applications. Cognitive Radio covers the services such as speaker recognition, language identification then translation into another, text-to-speech conversion, speech-to-text conversion, noise suppression, and noise management. For military applications, Cognitive Radio will allow finding free channel automatically to communicate instead of manually setting to a particular radio channel. Another benefit, CR can resist the jamming in a particular area where radio signal jammers are used and some radio spectrum is restricted. In disaster and emergency conditions, due to over-crowding of the radio spectrum, collapses the communication system. In such circumstances, when radio spectrum is completely occupied and no direct link is available to the access point but through other CR terminals, access point is reachable by forming spontaneous network [2-3].

The remainder of this paper is organized as follows: in Section II, the most popular spectrum sensing technique Energy Detection is discussed along with the flow chart explaining the method used for the detection of signal. Section III describes the Non-Time Dispersive Propagation Model: Hata Propagation Model. In section IV the system model specifications, for simulation on precisely built Cognitive Radio Environment for Global Systems for Mobile Communication using Spectrum Engineering Advanced Monte Carlo Analysis Tool (SEAMCAT), has been discussed. Section V includes the Simulation Results from different Non-Time Dispersive Fading regions as rural, sub-urban and urban area. Whereas Section VI, describes the impact of variations in number of secondary users on interference probability. In Section VII we draw our conclusions. Section VIII describes the Impact of present study on Cognitive radio Performance while Section IX shows the Future Scope for continuation of the research work on efficient Cognitive Radio Spectrum Sensing Techniques.

II. ENERGY DETECTION SPECTRUM SENSING TECHNIQUE

The secondary users are required to sense and monitor the radio spectrum within their operating range to detect the frequency bands that are not occupied by primary users. In this section, we discuss the most popular spectrum sensing

scheme, the energy detection. The energy detector employs a non-coherent detection technique, which does not require prior knowledge of pilot data. Energy detector spectrum sensing based approach, also known as radiometry or periodogram, is the most common way of spectrum sensing because of its low computational and implementation complexities. In addition, it is more generic as receivers do not need any knowledge on the primary user's signal. The energy of the received signal at secondary user is measured over an observation time window and compared with some threshold value to detect the presence of primary user over that particular channel. Mathematically, the energy detection method is a DFT based method to estimate power spectral density (PSD) and the energy of the input signal and compares it with a threshold energy value. The signal is said to be present at a particular frequency if the energy of the signal exceeds the threshold Energy level [3]. Energy Detection in IEEE 802.22 WRAN system is possible using the two methods namely (i) Received Signal Strength Measurement (RSSI) (ii) Multi resolution Spectrum Sensing (MRSS). In RSSI approach, one can select the unoccupied channels using received signal strength by converting the energy in an interested band to the input signal strength whereas in MRSS approach, it senses the interested band in the analog domain using a wavelet transform using Fourier Transform. Since, it is performed in analog domain, so spectrum sensing can be reduced. Above cited Energy Detection technique has been followed as shown in Figure 1 below.

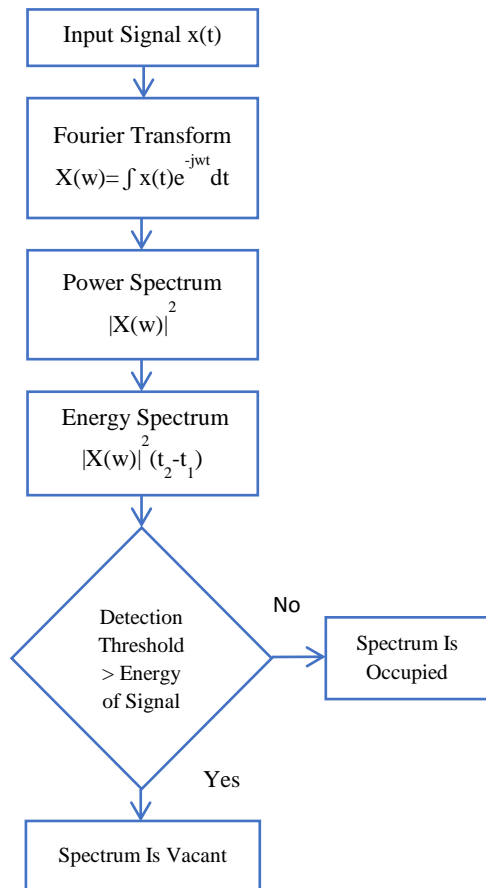


Fig. 1. Flowchart for Energy Detection Spectrum Sensing Technique

III. NON-TIME DISPERSIVE PROPAGATION MODEL

Wireless channel models are critical to both the analysis of interference to primary receivers in CR environment and the design of real-time methods to control such interference. An empirical model is based on data used to predict, not explain a system and are based on observations and measurements alone. It can be split into two subcategories, time dispersive and non-time dispersive. The time dispersive model provides the information about time dispersive characteristics of the channel like delay spread of the channel during multipath.

The Stanford University Interim (SUI) model is the perfect example of this type. The non-time dispersive provides the information about loss of signal strength in different propagation environments. Hata and ITU-R model are examples of non-time dispersive empirical model [4]. Radio propagation is also just as hard to predict as weather. Too many parameters have to be measured real-time, and too many details of terrain and obstacles have to be modelled to get accurate results. The extensive measurements of urban, suburban and rural radio environment losses, Okumura et al. published many empirical curves useful for cellular systems planning. These empirical curves reduced to a convenient set of formulas known as the Hata model. The basic formula for the median propagation loss given by Hata is: -

$$f_{propagation}(f, h_1, h_2, d, env) = L + T(G(\sigma)) \quad (1)$$

- Where f: frequency (MHz)
 h_1 : transmitter antenna height, m, above ground
 h_2 : receiver antenna height, m, above ground
d: distance b/w transmitter and receiver(km)
env: general environment.
Symbols: L = median propagation loss (dB)

IV. SYSTEM SPECIFICATIONS

The virtual simulator has been chosen to test the impact of environmental fading over the working of Cognitive Radio's or White Space Devices using energy detection spectrum sensing technique, is Spectrum Engineering Advanced Monte Carlo Analysis Tool [5-6].

The SEAMCAT tool models a primary mobile station transmitter (P-MS_{Tx}) connected to a primary base station receiver (P-BS_{Rx}) which describes a primary user link and operating amongst a secondary mobile station transmitter (S-MS_{Tx}) acting as a interferer to the primary user spectrum or channel as shown in the Figure2.

A secondary user belongs to either same system as the primary or different. The interferers are randomly distributed around the primary base station receiver in a manner decided by the user. It is common practice to use a uniform random distribution.

A. Calculation of Interference Probability

If

$$\frac{C}{I_{trail}} > \frac{C}{I_{target}} \quad (2)$$

Then there is no interference and event is considered good.

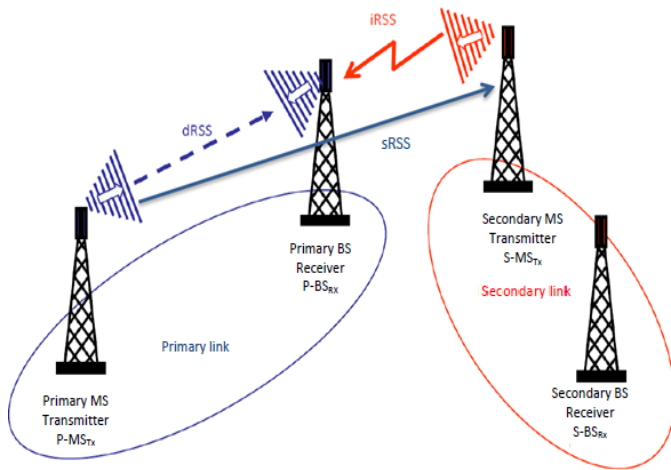


Fig. 2. Primary and secondary user modelling in SEAMCAT Where: dRSS - desired Received Signal Strength; iRSS - interfering Received Signal Strength; sRSS - sensing Received Signal Strength
If

$$\frac{C}{I_{trail}} < \frac{C}{I_{target}} \quad (3)$$

Then interference exists.

The probability of interference is given by: -

$$P_{interference} = 1 - \left(\frac{N_{good}}{N_{all}} \right) \quad (4)$$

Where: N_{all} = total random events for particular simulation

N_{good} = good events i.e. no interference.

The sRSS(sensing Received Signal Strength) at the channel 'm' is calculated as follows.

$$sRSS(f_m) = P_{P-MSTx}(f_m) + G_{P-MSTx \rightarrow S-MSTx} + G_{S-MSTx \rightarrow P-MSTx} + L \quad (5)$$

- Where
- P_{P-MSTx} : is the transmit power in dBm
 - f_m : is the frequency of the WSD.
 - $G_{P-MSTx \rightarrow S-MSTx}$: antenna gain in dBi of the P-MS_{Tx}, in the P-MS_{Tx} to S-MS_{Tx} direction.
 - $G_{S-MSTx \rightarrow P-MSTx}$: antenna gain in dBi of the S-MS_{Tx} in the S-MS_{Tx} to P-MS_{Tx} direction.
 - L: is the path loss in dB between the S-MS_{Tx} and the P-MS_{Tx}.

The non-cooperative spectrum access technique is used with the energy detection spectrum sensing [7]. The study has been conducted for the Global System for Mobile Communication, considering two channels with channel spacing 200 kHz to study the effect of detection threshold on interference at primary user receiver or victim receiver by single secondary user and to study the impact of increasing the number of secondary users or WSD on a primary user. Figure 3 shows the flowchart for the process of calculation of interference. The frequencies of the Primary Mobile Station Transmitter (P-MS_{Tx}) and Secondary Mobile Station Transmitter (S-MS_{Tx}) are same i.e. 890.1 MHz and 890.3 MHz because the Secondary Mobile Station Transmitter (S-MS_{Tx}) is a Cognitive Radio Device that will find the vacant channel in accordance with the detection threshold specified for it. The distance between primary mobile station transmitter

(P-MS_{Tx}) and primary base station receiver (P-BS_{Rx}) is fixed to 100 metres and the simulation radius of secondary mobile station transmitter (S-MS_{Tx}) for random distribution around P-BS_{Rx} is approximately 40 metres. The distance between secondary mobile station transmitter (S-MS_{Tx}) and secondary base station receiver (S-BS_{Rx}) is also fixed to 100 metres.

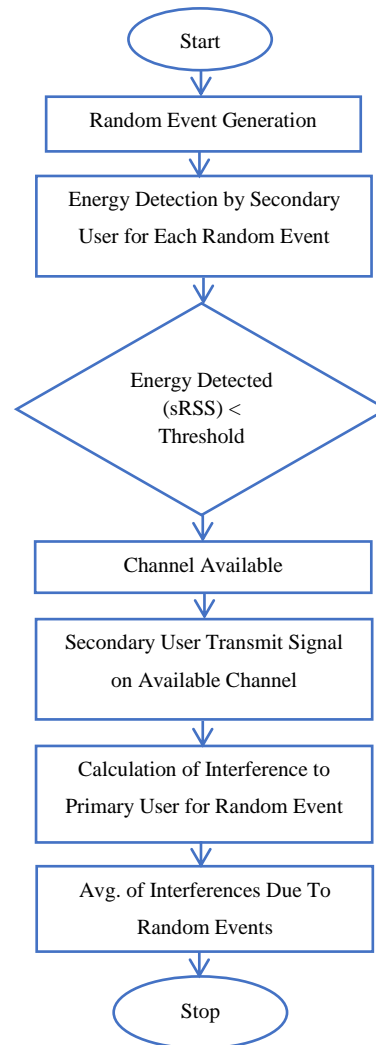


Fig. 3. Flow process of interference calculation

The antenna gain of mobile station transmitters and base station transmitters are 0 dB and 9 dB respectively. The antenna's pattern is omni-directional. The carrier to interference ratio (C/I) is considered 19 dB and sensitivity of receiver is -104 dBm. From this, the noise floor is -123 dBm and the blocking mask of the P-BS_{Rx} receiver is constant that is -23 dBm. Propagation model has been used for study, is Extended Hata model with variables in three different environments that are rural, sub-urban and urban areas. In model, the location of transmitter is also changed either indoor or outdoor and loss of walls is taken into consideration for study which is 10 dB for signal transmission from indoor to outdoor. The base station receiver is fixed outdoor. The propagation model specifications are kept same for each propagation link as the primary mobile station transmitter (P-MS_{Tx}) to the primary base station receiver (P-BS_{Rx}), secondary

mobile station transmitter (S-MS_{Tx}) to the primary base station receiver (P-BS_{Rx}) and the primary mobile station transmitter (P-MS_{Tx}) to the primary base station receiver (P-BS_{Rx}) [8-9]. The power transmitted by both P-MS_{Tx} and S-MS_{Tx} is 33 dBm and the emission mask of transmitters is considered as per the recommendation of the European Telecommunications Standards Institute (ETSI). The detection threshold for energy detection is considered with different constant values in dBm for different studies. The probability of failure or false alarm is considered 10 % for different propagation environments from prospective of comparative analysis (It could be different for different areas and may be much lower than 10%) and the reception bandwidth of secondary mobile station is 200 kHz.

V. SIMULATION RESULTS AND DISCUSSION

In this study, impact of changing a detection threshold of the White Space Devices on interference at primary user is studied for two picocell communication systems to share two primary channels. The interference has been compared with the detection threshold of energy detection spectrum sensing in terms of probability of unwanted interference occurrence. The comparison has been done for inter wireless radio environments for both indoor and outdoor mobile station transmitters as primary and secondary users as shown in Figure 4 and Figure 5. Then the comparison has been done for intra wireless radio environment such as rural area with indoor and outdoor mobile station transmitters as shown in figure 6, 7 & 8. The probability of interference is only accounted for the unwanted signal because the interference due to blocking is very low and negligible. The probability of interference has been increasing and attaining maximum value at different detection threshold values for different wireless radio environments for outdoor primary and secondary user transmitters. For rural area, interference arises at higher detection threshold value than sub-urban area and for sub-urban; it arises at higher detection threshold value than urban area. In rural area, interference probability arises as detection threshold value increases above -90 dBm and becomes constant and highest above -10 dBm; in sub-urban area, interference probability arises as detection threshold value increases above -110 dBm and becomes constant and highest above -30 dBm and in urban area, interference probability arises as detection threshold value increases steeply above -110 dBm and becomes constant and highest above -50 dBm as shown in figure 4.

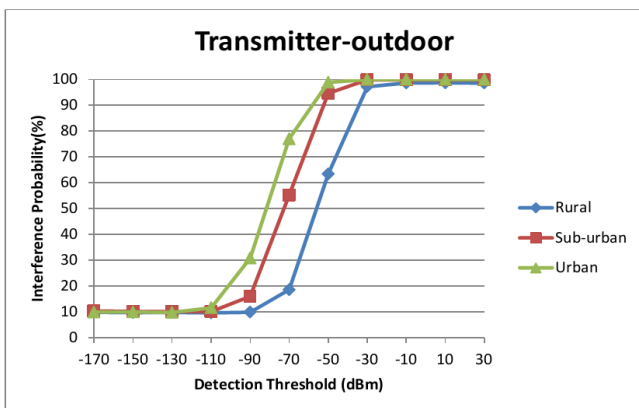


Fig. 4. Inter-Wireless Radio Environment with Outdoor Transmitters (DT)

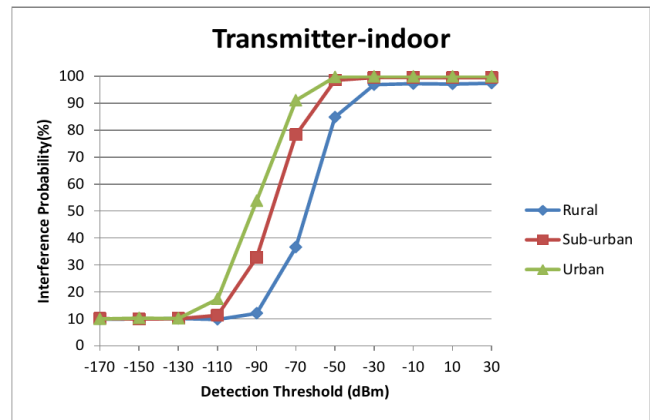


Fig. 5. Inter-Wireless Radio Environment with Outdoor Transmitters (DT)

In rural area, interference probability arises as detection threshold value increases above -100 dBm and becomes constant and highest above -30 dBm; in sub-urban area, interference probability arises as detection threshold value increases steeply above -110 dBm and becomes constant and highest above -50 dBm and in urban area, interference probability arises as detection threshold value increases above -130 dBm and becomes constant and highest above -60 dBm as presented in figure 5. The probability of interference increases when the mobile station primary and secondary users are moved from outdoor to indoor environment at particular detection threshold. For intra-wireless radio environment: rural area, the interference probability arises at -100 dBm when the transmitters are indoor and arises at -90 dBm when the transmitters are outdoor as shown in fig 6.

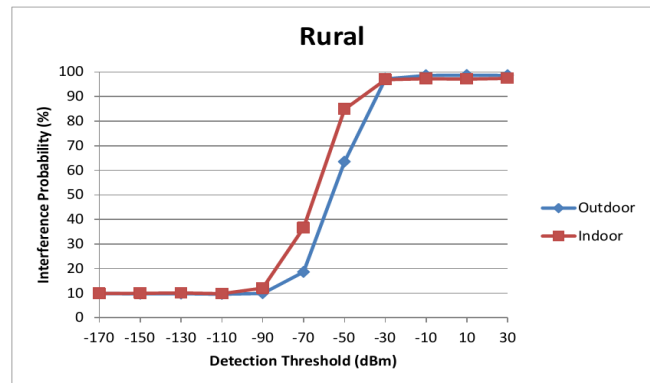


Fig. 6. Intra-Wireless Radio Environment – Rural Area (DT)

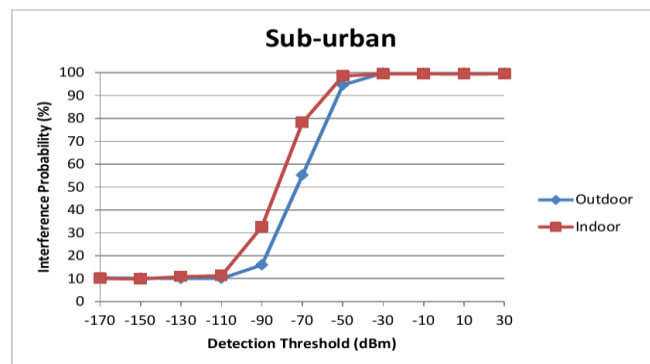


Fig. 7. Intra-Wireless Radio Environment – Sub-urban Area (DT)

In intra –wireless radio environment for both sub-urban and urban areas, the detection threshold for indoor transmitters is required less than the detection threshold for outdoor transmitters as predicted in Figure 7 and Figure 8 respectively.

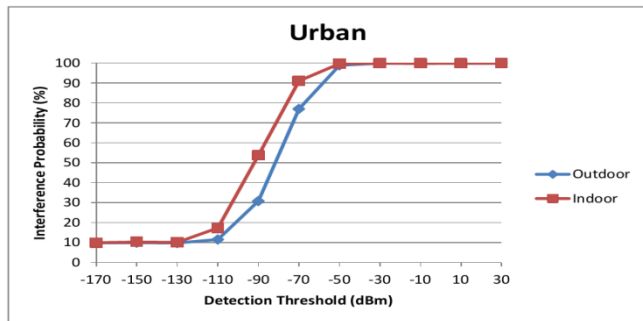


Fig. 8. Intra-Wireless Radio Environment – Urban Area (DT)

The interference probability arises as mobile station users moved from outdoor to indoor environment at particular detection threshold value due to the increase in the path loss and decrease in Sensing Received Signal Strength; which makes more randomly generated samples to fall under lower value of detection threshold.

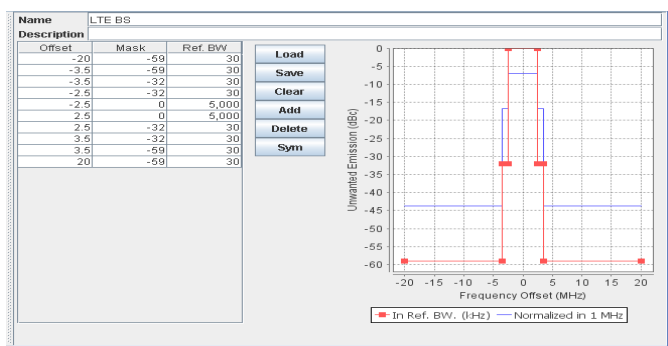


Fig. 9. shows the Spectral Emission Mask for LTE-BS

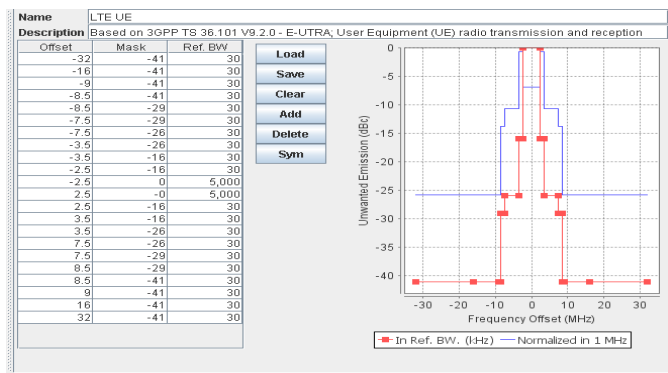


Fig. 10. shows the Spectral Emission Mask for LTE-UE

Figures 9 and 10 show the Effective Isotropic Radiated Power [EIRP] in block mask using SEAMCAT which inputs the In-block CR emission limit as a triplet offset, Mask, Ref.BW] where Offset in Mhz is equivalent to the channel spacing, Mask in dBm is the In-block CR EIRP max limit and Ref BW is the bandwidth of the primary base station receiver.

TABLE I. SHOWS SRSS VALUES WITH OUTDOOR TRANSMITTERS

Areas	Mean	Min	Max
Rural	-53.75	-101.88	-7.80
Sub Urban	-70.38	-125.06	-23.25
Urban	-79.50	-129.41	-29.64

TABLE II. SHOWS SRSS VALUES WITH INDOOR TRANSMITTERS

Areas	Mean	Min	Max
Rural	-63.55	-117.03	-12.33
Sub Urban	-80.49	-138.99	-30.02
Urban	-89.37	-150.12	-32.62

The minimum and maximum sRSS values or cumulative probabilities of sRSS values could differ little due to the generation of random samples for every simulation but a mean value is approximately same.

VI. VARIATIONS IN INTERFERENCE PROBABILITY DUE TO CHANGE IN NUMBER OF SECONDARY USERS

In this study, the impact of increasing the number of mobile station secondary users over the primary link has been studied in terms of interference probability within different radio environment[10-11].The detection threshold is assumed -100 dBm that is average value of all detection threshold values for inter wireless radio environment. The interference probability for each number of secondary users for indoor transmitter’s in sub urban area results in same interference probability as for outdoor transmitters in urban area due to the probability of random generation events of sensing Received Signal Strength under detection threshold being same for both. Wireless channel models are critical to both the analysis of interference to primary receivers in CR environment and the design of real time methods to control such interference[12-13].

VII. CONCLUSION

The difference in the sensing Received Signal Strength, the interference experienced by the primary link base station receiver is also different with changing the values of detection threshold in different wireless radio environment. At highest detection threshold, the interference experienced by the base station receiver has achieved its highest value for each radio environment and the interference occurs by secondary mobile station transmitter using both adjacent and co-channel in equal proportion of randomly generated events. As the detection threshold decreases, then the presence of primary user is detected more precisely and the interference is reduced as the secondary user is not allowed to transmit over the frequency channels used by the primary user. The cognitive radio should be programmed for lesser detection threshold value for the urban area than other areas to achieve minimum interference probability at primary users. Cognitive radio is a futuristic technology that will utilize the spectrum efficiently and globalize the radio environment for different band users which use different techniques for transmission.

VIII. IMPACT OF STUDY

The study will be useful in radio environment which is opportunistic, adaptive and intelligent under different strategic

conditions. This study will provide base for maximum capacity or spectrum utilization under different wireless fading environment which will be helpful in the design of Intelligent Wireless Communication System under the constraints of Interference. The study has its impact on design and development of cognitive radio system under ubiquitous pervasive environment [14-18].

IX. FUTURE SCOPE

It is well-known that energy detector's performance is susceptible to uncertainty in noise power under such cases alternate detection schemes such as cyclic feature detection, Wavelet based detection and Filter bank multicarrier approach or power control scheme may be employed. The Performance analysis of spectrum sensing in this scenario is the subject of our current research.

Further study will be carried out by accounting other wireless radio environment parameters such as refractivity, troposphere scatter, etc using propagation models like ITU-R P.452-14 over the performance of Cognitive radio using energy detection spectrum sensing. The effect of small scale fading can be accounted for different wireless radio environment. The impact of variable distance can be studied using cooperative approach in future by modeling different wireless radio propagation environments for Cognitive radio as the number of secondary users is more than one. The interference effect could be studied using power control feature for secondary users in different wireless radio environment.

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Adaptive Lockable Units to Improve Data Availability in a Distributed Database System

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Abstract—Distributed database systems have become a phenomenon and have been considered a crucial source of information for numerous users. Users with different jobs are using such systems locally or via the Internet to meet their professional requirements. Distributed database systems consist of a number of sites connected over a computer network. Each site deals with its own database and interacts with other sites as needed. Data replication in these systems is considered a key factor in improving data availability. However, it may affect system performance when most of the transactions that access the data contain write or a mix of read and write operations because of exclusive locks and update propagation. This research proposes a new adaptive approach for increasing the availability of data contained in a distributed database system. The proposed approach suggests a new lockable unit by increasing the database hierarchy tree by one level to include attributes as lockable units instead of the entire row. This technique may allow several transactions to access the database row simultaneously by utilizing some attributes and keeping others available for other transactions. Data in a distributed database system can be accessed locally or remotely by a distributed transaction, with each distributed transaction decomposed into several sub-transactions called participants or agents. These agents access the data at multiple sites and must guarantee that any changes to the data must be committed in order to complete the main transaction. The experimental results show that using attribute-level locking will increase data availability, reliability, and throughput, as well as enhance overall system performance. Moreover, it will increase the overhead of managing such a large number of locks, which will be managed according to the qualification of the query.

Keywords—Granularity hierarchy tree; Lockable unit; Locks; Attribute level; Concurrency control; Data availability; Replication

I. INTRODUCTION

Distributed database system (DDBS) may be defined as a collection of multiple, logically interrelated databases distributed over a computer network [12]. This system stores a huge amount of data that have been accessed by a large and increasingly growing number of users. Distributed database system is a crucial source of information for numerous users who access the database locally or via the Internet for different tasks. To meet the professional requirements of users, data must be available at all times, because data availability plays a major role in the success of information systems.

Data can be accessed by a local transaction when it does not require other sites, or by a distributed transaction in which two or more database sites are involved [12]. Each site has a local transaction manager responsible for coordinating

transactions across one or more database resources [1, 2]. During transaction execution, the lock manager locks the required database items by sending a message to the central site (in case a central lock manager location is used). If the requested lock is granted, then the lock manager sends a message to the requested site; otherwise, it waits. In case of a write operation, the lock manager must lock all copies of the requested database item in all sites where it exists, but in a read operation, the transaction is executed at a local copy that exists or at any copy at any available site [4,13,16].

In the study of locking techniques, the size of the lockable units clearly has a major effect on the concurrency control and the availability of data, because while the database unit is locked, it will be unavailable for a time. Thus, if the locked unit is a table, then no other transaction can access that table in a conflict mode until the lock is released. According to this problem, the study on the means to reduce database units, which in turn increases the database resources, became necessary.

The present research proposes a new approach for increasing the data availability by suggesting the attribute as a new lockable database unit. This technique may be implemented by increasing the database hierarchy tree by one more level down to include the attributes as lockable units instead of the entire row. The proposed approach may allow several transactions to access the same database row simultaneously, which may increase the degree of concurrency and the availability of data. This research uses three-phase locking instead of two-phase locking protocol. Three-phase locking protocol has a pre-commit phase to prevent the blocking state. To simplify the implementation, a central locking approach is considered, which means there is one site that has a lock manager and must coordinate with other sites in the system. Locking can be granted on some attributes of a row, including the key of that row if no conflicts among transactions could occur as the compatibility matrix adheres [4,12,13], as shown in Table I.

TABLE I. COMPATIBILITY MATRIX

	IS	IX	S	SIX	X
IS	T	T	T	T	F
IX	T	T	F	F	F
S	T	F	T	F	F
SIX	T	F	F	F	F
X	F	F	F	F	F

TABLE II. SYSTEM PARAMETERS

Parameter	Description	Values
Num-of-sites	Number of sites in the system	M
Num-of-DB	Number of databases in each site	1
Rep-degree	Degree of replication	0.2–0.8
Num-of-tables	Number of tables in a database	50
Num-of-trans	Number of transactions in the system	5000
Min-trans-size	Minimum number of operations	1
Max-trans-size	Maximum number of operations	20
OP-Mode	Operation mode	R, RW, W
Queue-Length	Maximum queue length	30
CheckT	Mean time to check a lock	1 ms
SetT	Mean time to set a lock	1 ms
RelT	Mean time to release a lock	1 ms
Ex-Time	Mean time to process a data object	20–150 ms

The remainder of this paper is organized as follows. Section 2 presents the background and literature review. Section 3 presents the proposed approach. Section 4 discusses the experiments and the produced results. Section 5 contains the conclusion.

II. BACKGROUND

The problem of data availability and the degree of concurrent transactions have been discussed by several researchers [2, 3, 8, 10] who concentrated on a strategy of dividing the database into variable size units. The size of such units is dynamically managed by the lock manager based on user needs and competition. This competition increases more in a distributed database system than in a centralized one because of the higher number of users.

A proposed simulator for a distributed object-oriented database to evaluate the concurrency control and performance of the system is presented by Norvag et al. [7]. Their simulation results show the comparison of performance and response times for two concurrency control algorithms, namely, timestamp ordering and two-phase locking. Their results show that two-phase locking outperforms timestamp ordering, specially in long transaction workload, because of the very high abortion rate in timestamp ordering. Defining new lock types and their compatibility matrix in DDBS is presented by Zhangbing et al. [17]. These types are produced to overcome the disadvantages of the traditional locking mechanism in a DDBS. Their experimental results show the enhancement of control and flexibility of locks with improved 2PL protocol and multi-granularity locking. Their improved protocol effectively ensures the serializability of scheduling transactions and decreases the communication costs while locking. It also obtains better transaction concurrency than the traditional mechanism.

Sorapak et al. [15] studied the evaluation of distributed database system performance by using MySQL cluster. Eight nodes are used to test their system. The results of their research showed the relations between query processing time and number of system nodes, which indicates that the processing time is improved when the number of system nodes is increased.

A Distributed Database Performance Tradeoff among fairness, isolation, and throughput features is studied by Jose and Daniel [6]. Their study showed that only two of the three features can be fulfilled simultaneously. Fairness means the received transactions are processed immediately without delay, isolation means a transaction cannot block or abort another transaction, and throughput means that the system will run a transaction without interference among them because of synchronization independence. Maabreh and Al-Hamami [14] implemented a study on the approach to increase the database hierarchy tree. Their study was based on a two-phase locking protocol with three sites, which represent a limited number of sites in terms of the possibility of producing insufficient results.

III. PROPOSED APPROACH

A. Data Set

To investigate the proposed idea of decreasing the size of a lockable unit, a homogeneous distributed database system consisting of m equivalent sites will be implemented. Each site has its own database, which can be accessed locally or remotely. The number of sites is extended to m instead of three as we studied in [14], and the numbers of database objects and concurrent transactions are increased to obtain more significant results and conclusions. In case of update operations, one copy of each object will be selected as a master copy and will be located at a specific site. The sample tables in this study have to be replicated over the system as 1-D partial replication (some objects to all sites) [10]. These tables are randomly filled with 10,000 rows as a sample of virtual data and distributed across the system. Each table has one master copy placed at one site, while the other copies are considered as replicas. To simplify the analysis of the produced results, an example of 40 sites is chosen for study. Transactions with different operation modes are also randomly generated. Table II shows the details of the system parameters that will be used in the simulation program.

B. Approach Methodology

Fig. 1 shows a sample of a multiple granularity tree, which represents the organization of data within a database. Granularity is a dynamic size of the database item, which may be locked by a transaction. Multiple granularity locking will allow several transactions to be executed at different sizes, ranging from whole database to a specific row, because transactions often do not need to lock at the higher granularity and therefore, free up objects that would otherwise still be locked. This research presents an attribute as a new granule size, because whenever the granule is being as small as possible, then more transactions could be executed simultaneously, which increases concurrency in the system.

Before building the simulation program, three essential components were implemented: the transaction manager, the lock manager, and a tree to represent the database objects. The transaction manager is responsible for keeping track of the executed transactions from the start state to the terminated state, whether its commit or abort. The lock manager is responsible for acquiring the locks needed by the transaction and ensuring that no conflict may occur among transactions by

adhering to the compatibility matrix (Table I). The lock manager uses the three-phase locking protocol by assuming that one site could fail in the system, because multiple site failures could cause a problem as proved in [11]. Three-phase locking protocol is a nonblocking protocol because it includes a pre-commit phase. This phase is reached if all transaction participants have voted to commit; in this case, the transaction is committed unless a site fails as a timeout period may indicate. Otherwise, and if this state is not reached, the participant will be aborted and the blocked resources will be released. Finally, a hierarchy tree representing the multiple granularity of the database objects (Fig. 1) is implemented by using NetBeans IDE 8. Deadlock can be detected by using a predefined timeout. Data used in this approach are virtual, so the total execution time for a database object can be guessed in advance according to the system parameters shown in Table II and by using the following formula:

Total-processing-time= CheckT + SetT + RelT + Max (Ex-Time). Given that the timeout is considered a time limit on how long a transaction may be active, when a transaction exceeds this time, the transaction has a deadlock.

As an example, the maximum total execution time for a specific database object could be 153 ms (i.e., 1+1+1+150) by assuming that the object has a maximum processing time. Thus, if the waiting time for the database object that has been locked by a transaction exceeds 153 ms, then the system has a deadlock and must be solved. A drawback of this deadlock detection approach is the long time taken to detect a deadlock.

C. Approach Description

This research aims to include the attributes that would be the new lockable units for allowing several transactions to access the same database row concurrently. This approach may increase the database resources, which would increase the concurrency and throughput in the system and decrease deadlock occurrences. In contrast, the overhead may be increased, but it will be managed as will be explained in a later part of the research. Fig. 1 shows the suggested attributes as new lockable units; the attributes may be locked individually when a transaction requires only some attributes of the database row. This “locking” can be performed as explained in [9] and as follows.

- 1) The database row is locked in an intent exclusive mode (IX).
- 2) The key of that row is locked in a shared mode (S).
- 3) The required attributes can be locked by the requested transaction in read or write.
- 4) Consistency constraints between attributes are considered. For example: if $A=B+C$ is a constraint among the attributes A, B, and C, then when any transaction needs any of those attributes, the lock manager locks all of them to satisfy that constraint.

The following example may illustrate the importance of the idea.

Let R ($A_1, A_2, A_3, A_4, \dots, A_n$) be a database relation (table). A_1, A_2, \dots, A_n are the attributes of R, A_1 is the key, and let $t=\langle v_1, v_2, v_3, v_4, \dots, v_n \rangle$ be any tuple in R. Furthermore,

consider the following three transactions $T_1, T_2,$ and $T_3,$ which need to access the same database row according to the following scenarios:

- T_1 : Update R Set $A_2= A_2+N,$ where $A_1=v_1;$
- T_2 : Update R Set $A_4= A_4+M,$ where $A_1=v_1;$ and
- T_3 : Select A_n from R, where $A_1=v_1;$

where v_1 is the value of the attribute A_1 in the same row, and N and M are some values. In the current situation of a database, these transactions cannot be executed simultaneously because the database row is considered as one block and can be locked by only one transaction; the other transactions will be waiting until the locked row is released. In this example, T_2 has to wait for T_1 to finish (because the data requested by T_2 are not yet available), and T_3 has to wait for both T_1 and T_2 to finish for the same reason of T_2 . This scenario is an example of increasing the average waiting time and reducing the data availability.

When the proposed idea is implemented, all the three transactions in this example will be executed at the same row simultaneously. $T_1, T_2,$ and T_3 will lock the database row in an intent exclusive mode (IX), and thus the key of that row will be locked as a shared (S), T_1 will lock A_2 as an exclusive (X), T_2 will lock A_4 as an exclusive (X), and T_3 will lock A_n as shared (S). This technique may reduce the waiting time and increase the average response time and data availability.

To implement the suggested approach, an event-driven simulation program was built by using Java Netbean IDE version 8.1. The simulator contains the necessary components for a distributed database system, as shown in Fig. 2. Transactions are treated as threads to be executed concurrently and managed by the transaction manager. The lock manager is responsible for locking and releasing the database items as the transaction need arises, as well as communicates with the transaction manager and with the database through a network. For simplicity of analysis, the network model is assumed to be LAN of 5 ms inter-processing time based on Gray and Reuter measurement [5]. The database is implemented as a hierarchy tree representing the granules. Transactions are then requested by the granule size as needed.

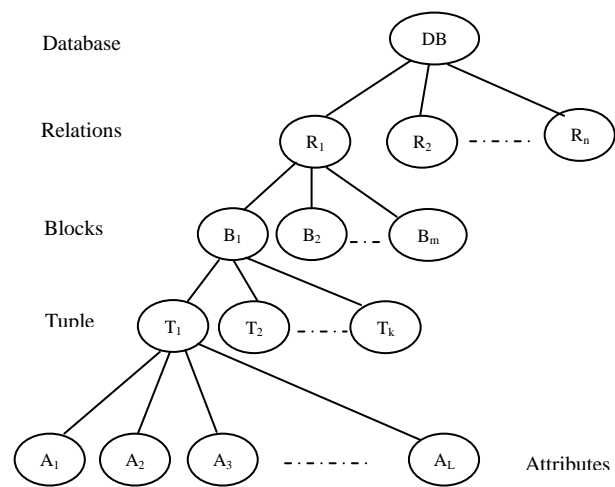


Fig. 1. Sample of Multiple Granularity

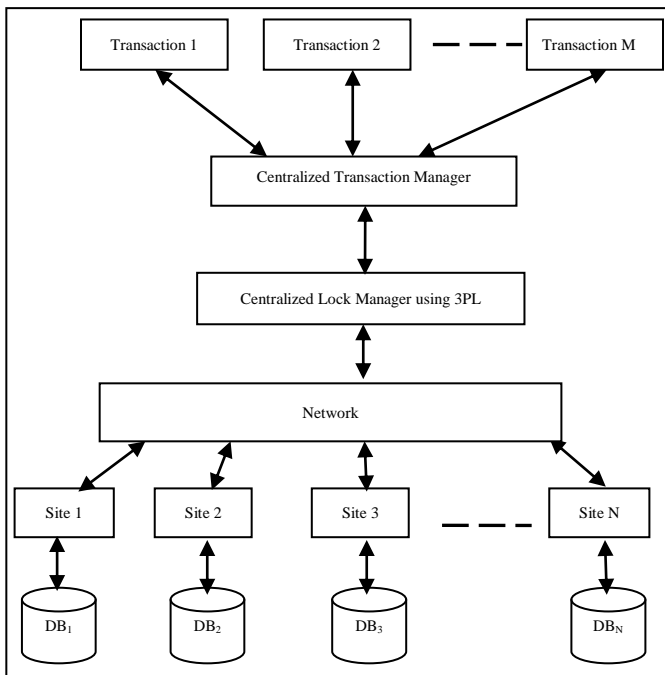


Fig. 2. Simulation Program Architecture

IV. DISCUSSION

This section demonstrates the results of the series of experiments conducted to evaluate the effectiveness of the proposed approach. Five hundred transactions are generated randomly as a sample run with different operation modes. Experiments are repeated several times with different random numbers of transactions under different degrees of replication to observe and evaluate the system behavior. The simulation program is running in both cases: one when the database row has the lowest granule, and the other when allowing the transaction to access specific attribute(s) of the row. Then, the results concerned with average execution time, average waiting time, and the overhead are collected and analyzed.

Fig. 3 shows the average execution time of the system. The average execution time of the proposed approach is less than the average execution time of the current lock mechanism, which means that the transaction almost does not need much time to acquire its requested operation because of the attribute

sharing among transactions. The locks are incrementally and dynamically acquired, so the transaction may request an attribute of the row while the other attributes may be available or acquired by another transaction. This finding means that the transactions access the needed attributes without delay, which reduces the average execution time.

The average waiting time of the proposed approach is also less than that of the current lock mechanism (Fig. 4) for the same reason. The overhead that occurs because of attribute-level locking is managed by locking the database row when the transaction requires numerous attributes of the same row. Fig. 5 shows the increasing overhead and the management of locks at the row level instead of the locking of many attributes. An investigation of Figures 3 to 5 reveals that the system performance of 270, 280 and 290 transactions as a workload seems to be the same in terms of average execution time, average waiting time, and overhead (i.e., The two lock mechanisms operate the same, which is the lock manager's decision). This outcome is attributed to the numerous attributes of the same row a transaction requires, causing the lock manager to return one level up on the tree and attempt to lock the row. This finding is the same result as the one that occurred when a transaction needs multiple rows of the same table; in this case, the lock manager locks the whole table in order to reduce the overhead.

Reducing the average execution time and average waiting time increased data availability because the times to lock and access the data are decreased. Transactions in this case may obtain the data immediately or by less waiting time as much as possible. For example, by using the company schema, Fig. 9.2 in Elmasri and Navathe (2015) [4], the following two transactions will be processed together at the same database row.

T_1 : Update Employee Set Salary=Salary+N, where SSN=M;

T_2 : Update Employee Set Super-ssn=new Super-ssn, where SSN=M; where M is the same employee number. This example shows that the lock manager locks SSN because both transactions are shared, and then the salary is locked exclusively for T_1 and the super-ssn as an exclusive for T_2 . This result means that, both transactions T_1 and T_2 will be executed concurrently because the data are available.

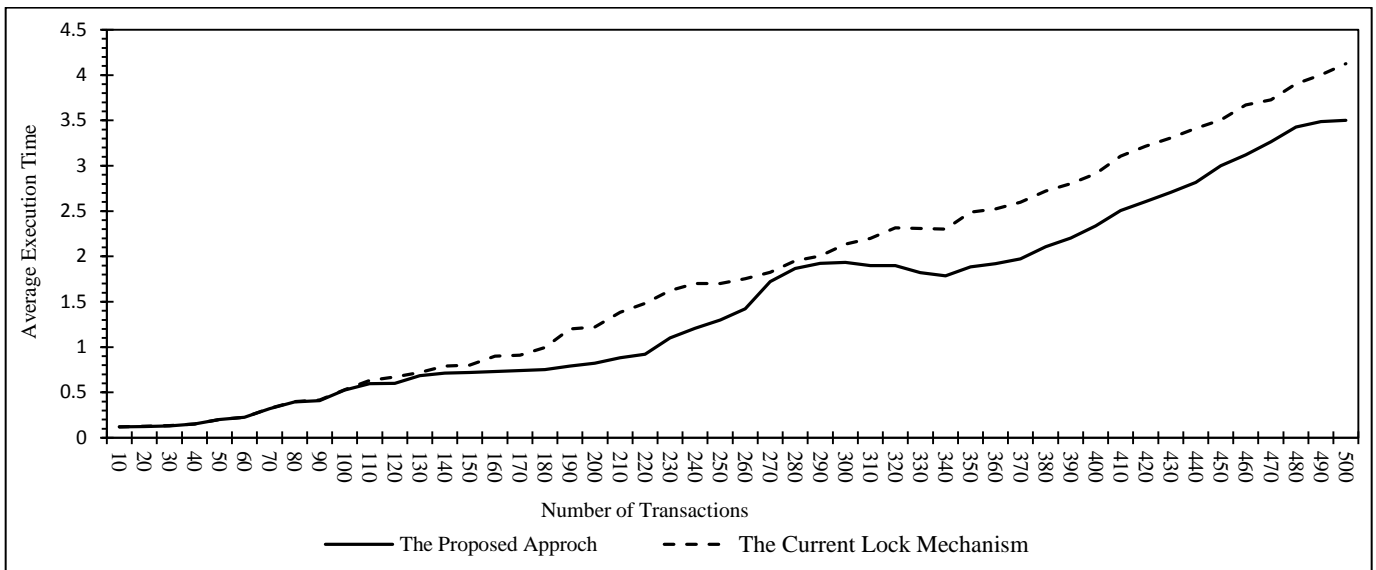


Fig. 3. Average Execution Time

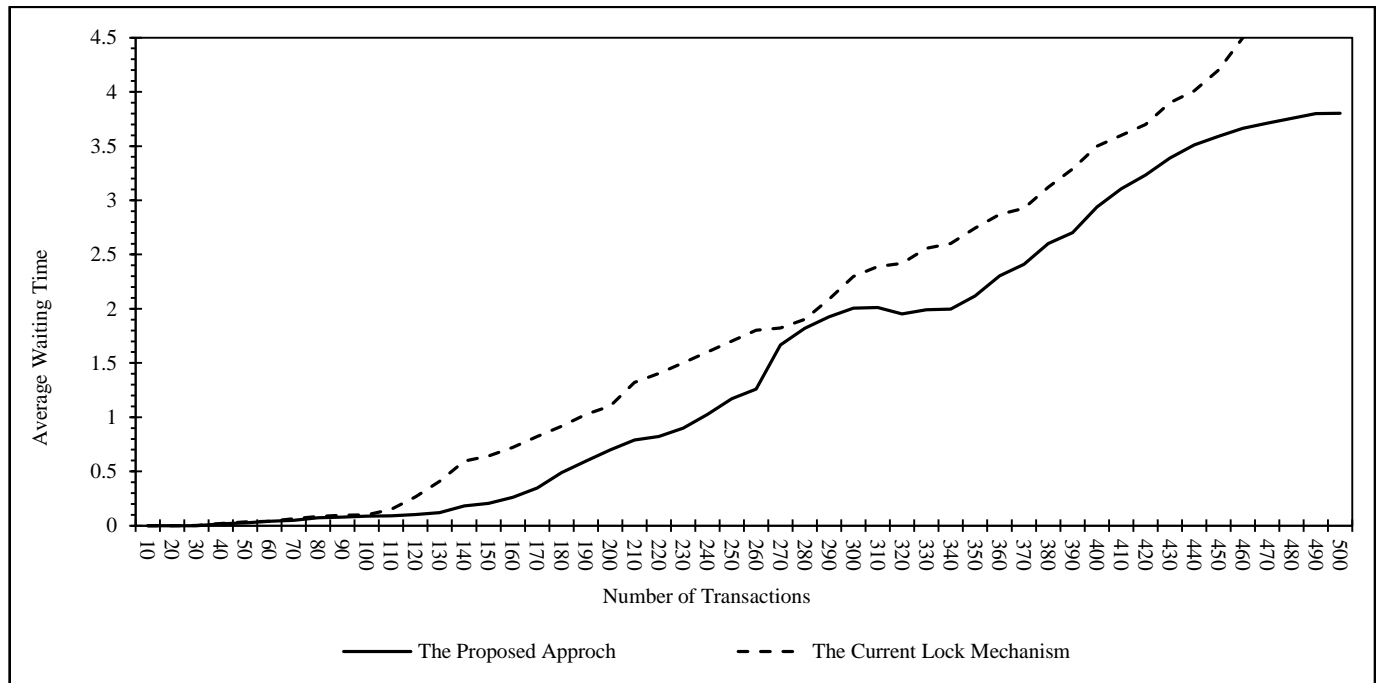


Fig. 4. Average Waiting Time

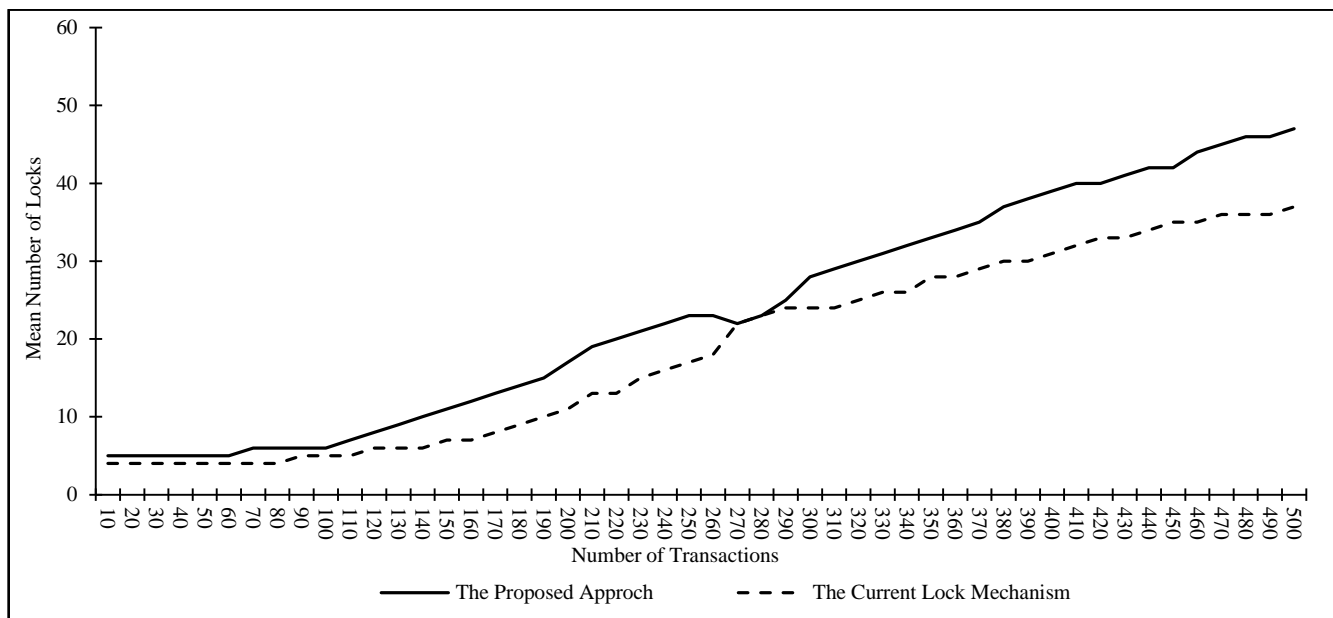


Fig. 5. System Overhead

V. CONCLUSIONS AND FUTURE WORK

Distributed database systems are considered crucial sources of information. Therefore, data contained in such systems must be available at all times as much as possible to satisfy the user's professional needs. To increase the data availability, this paper proposes a new adaptive approach to increase the database items by reducing the size of the lockable units. This reduction can be carried out by locking the attributes instead of the database row, which remains as the other attributes become available for other transactions. The experimental results showed that using attribute-level locking increases the degree of concurrency by increasing the data availability. The overall system performance is also improved because the average waiting time is decreased. The increasing overhead is managed by returning the lock at the row level when a transaction requires many attributes of the same row. The proposed approach is suitable with short transactions of mixed read and writes operations, especially when the degree of replication is less than 50%.

Further work for studying the proposed approach could be implemented by having more sites, larger data set and a higher workload, as well as more practical examples, experiments and comparison with other technologies will also be studied as a future work to improve the quality of the research.

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A Multipath Lifetime-Prolonging Routing Algorithm for Wireless Ad Hoc Networks

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Abstract—Dynamic networks can be tremendously challenging when deploying distributed applications on autonomous machines. Further, implementing services like routing and security for such networks is generally difficult and problematic. Consequently, multi-agent systems are well suited for designing distributed systems where several autonomous agents interact or work together to perform a set of tasks or satisfy a set of goals, hence moving the problem of analyzing from a global level to a local level therefore reducing the design complexity. In our previous paper, we presented a Multi Agent system model that has been adapted to develop a routing protocol for ad hoc networks. Wireless ad hoc networks are infrastructureless networks that comprise wireless mobile nodes which are able to communicate with each other outside the wireless transmission range. Due to frequent network topology changes, the limited energy and underlying bandwidth, routing becomes a challenging task. In this paper, we present a new version of routing algorithm devoted for mobile ad hoc networks. Our new algorithm helps controlling the network congestion and increasing the network lifetime by effectively managing nodes energy and link cost. The performance of our new version is validated through simulation. The Simulation results show the effectiveness and efficiency of our new algorithm compared to state-of-the-art solutions in terms of various performance metrics.

Keywords—Mobile Multi Agent System; Ad hoc Network Lifetime; Ant Routing Protocol; Distributed Algorithm; Network Congestion

I. INTRODUCTION

Dynamic networks can be tremendously challenging when deploying distributed applications on autonomous machines. But, these networks can meet problems when implementing services such as routing and security in general.

We have chosen ad hoc networks as a case study. For this, we focused on multi-agent approach which has a particular interest in the distributed problems in general, and for which it is difficult to prevent all situations. The multi-agent systems [1–3] are well suited for the design of distributed systems. Several autonomous agents interact or work together to perform a set of tasks or satisfy a set of goals, and moving the problem of analyzing from a global level to a local level and reducing the design complexity. In this paper, we present a generic mobile Multi Agent system that has been adapted to develop a new routing protocol version for ad hoc networks (The first version is published in [4]). The first goal is improving the previous version in terms of packet loss and end to end delay by adding new features to it. The second goal is increasing the network

lifetime by effectively managing the nodes energy and controlling the network congestion during route discovery process and data transmission.

In multi-hop wireless ad hoc networks (MANETs) [5], mobile nodes cooperate with each other to form a network without a fixed infrastructure such as access point or base stations. The network nodes perform routing discovery and routing maintenance in a self-organized manner. The routing is particularly a challenging task in MANETs because of the frequent changes in the network topology triggered by nodes displacements, the establishment of new nodes connections and nodes disconnections and the unstable routes discovery process. Practically speaking, efficient routes may quickly become inefficient or even unusable ones. To tackle this problem and ensuring a suitable routing through reliable algorithms, one important way is to update routing information more regularly than in wired networks. However, this requires more routing control packets which is specifically an issue in MANETs since the bandwidth of the wireless medium is very limited and the medium is shared.

Beyond the routing overhead problem, our proposed protocol also attempts to solve several problems such as: packet delivery ratio, end-to-end delay, network lifetime and energy consumption. For this purpose, we propose a hybrid method that entails both proactive and reactive processes. The routes are established and periodically maintained with a constant number of mobile agents. The Agent is periodically created by each node, thus the number of agents in the network can continually be controlled. However, when a connection is planned to be established by a node with another one in the absence of a route in its routing table, the latter makes a route request by setting a local variable available for Agents passing through it.

Our model is based on the ant behavior. A number of ant-based routing algorithms exist either in wired or in wireless [6–13] networks (Ad hoc and sensor networks). They are based on the pheromone trail laying-following behavior of real ants and the related framework of ant colony optimization ACO [14]. In all of these approaches, a source node broadcasts an Agent whenever it plans to build a route to a fixed destination. One of Agent roles is to deposit amounts of pheromone in order to mark optimal paths between a couple of nodes namely source and destination nodes. Unlike these methods, we do not exploit a broadcasting technique that exponentially increases the routing overhead, rather we introduce a new idea via an ant-based algorithm that consists in setting a local route request

whenever a node plans to send a data packet. It is, therefore, the Agent's role moving within the network during the proactive phase to disseminate this information and provide routes towards the requested destination. Note that our protocol doesn't necessarily establish the best route, since the agents are not broadcasted. However, the agents attempt to get as close as possible to the best route.

The remainder of this paper is organized as follows. In section 2, we present some definitions on self-organized multi-agent systems and their analogy with dynamic networks. In section 3, we present the principle of ad hoc networks and then describe the routing problem in this class of networks; afterwards, we give an overview of some representative related works. In section 4, we present the general architecture and mode of operation of a multi-agent system adapted to a generic case of dynamic systems. We detail the instantiations of the generic model: the previous version of our routing protocol for ad networks. In section 5, we present a new version of our routing approach. Before concluding in section 7, we discuss in section 6 results and tests of the new version of routing protocol using the NS2 simulator.

II. MULTI-AGENT SYSTEMS AND DYNAMIC NETWORKS: ANALOGIES AND ISSUES

Over the last years, a new topic of research has emerged: dynamic networks (also called autonomous systems). A dynamic and distributed network consists of a set of auto-configurable nodes that are constantly changing (the number of nodes and links change over time). The topology change is also one of the properties of these networks, because the network nodes can join and/or leave the network spontaneously. The main advantage of this type of networks is the fast and inexpensive deployment and installation.

A multi-agent system (MAS) is a set of agents operating in a common environment. This set of agents, not necessarily smart, is a complex system that shows a collective intelligence. This collective intelligence comes from the emergence of a global behavior of all agents. An example of such collective action concerns the behavior of a colony of ants that act like an entities (ants) with no cognitive capacity, but achieving a high degree of organization and adaptation quote.

An agent is a software entity (program) reactive, proactive and with social skills, able to act autonomously in its environment. Responsiveness refers to maintaining a constant link with the environment when a change occurs. The proactivity means that the system allows agents to generate and satisfy its goals. Social skills indicate that the system allows the agent to interact or cooperate with its environment and / or with other agents.

It is, therefore, clear that there is an analogy between dynamic networks and multi-agent systems. In fact, each node of a dynamic network is autonomous because 1) it is not controllable by any other node on the network, 2) reactive because it can act as a server for other nodes, 3) proactive in the case of its client node status, and, lastly, 4) social because it communicates and cooperates with other nodes in the network.

Accordingly, an inherent issue in the management of dynamic networks is: the definition of a data routing protocol in the case of ad hoc networks.

III. ROUTING ISSUES IN AD HOC NETWORK AND RELATED WORK

In multi-hop "wireless" ad hoc networks (MANETs), "mobile nodes" cooperate with each other to form a network "without a fixed infrastructure" such as access point or base stations, the network nodes perform a routing discovery and a routing maintenance in a self-organized way. In other word, Wireless ad hoc networks are infrastructureless networks that comprise wireless mobile nodes that are able to communicate with each other outside the wireless transmission range. Due to frequent network topology changes and the limited underlying bandwidth, routing becomes a challenging task.

Several types of routing protocols have been specifically designed for ad hoc networks and classified into two main categories: reactive and proactive protocols. In reactive routing protocols such as AODV [15] (Ad Hoc On demand Distance Vector) and DSR [16] (Dynamic Source Routing), the routes are only discovered when required in order to save node and network resources, while in proactive routing protocols such as OLSR [17] (Optimized Link State Routing Protocol) and DSDV [18] (Destination Sequenced Distance Vector) the routes are established in advance, avoiding consequently the delays that occur during the discovery of new routes.

The problem raised by proactive protocols consists in the routing overhead, especially when there are frequent topology changes. This is highly inefficient when updating routes that rarely carry traffic. A reactive protocol is, in contrast, much more appropriate for such situations, since it generates lower overhead in terms of used bandwidth.

There is another kind of protocol that combines both reactive and proactive approaches called hybrid routing protocol. In our work, we focus on a particular class of hybrid routing protocols based on an optimization technique known as ant colony optimization (ACO) which are inspired from the foraging general behavior of some ant species. The ant underlying behavior can be summarized as follows: ants deposit pheromone on the ground in order to mark some favorable paths that should be followed by other members of the colony, for instance, ants walking to and from a food source deposit on the ground a substance called pheromone. Other ants perceive the presence of pheromone and tend to follow paths where the pheromone concentration is higher. Through this mechanism, ants are able to transport food to their nest in a significant effective way.

Several properties belonging to ant-based routing algorithms are strongly appropriate to address the problems inherent in MANETs: they are highly adaptive to network changes, robust to agent failures, and provide multipath routing [19]. However, since they mainly rely on repeated path sampling, a significant overhead can be induced within native routing algorithms. Several ant-based routing algorithms for MANET have been proposed in state of the art previous work.

However, in order to limit the overhead caused by the ants, these algorithms considerably lose the inherent proactive sampling and exploratory properties belonging to the ants behavior adopted in the original ant-based algorithms.

The classic ant methods inspired from Ant-Colony-Optimization algorithm like ARA [20] and AntHocNet [21] use during the reactive phase, a broadcasting technique that exponentially increases the routing overhead and energy consumption: When a data session is started at node s with destination d , s checks whether it has up-to-date routing information for d . If not, it reactively broadcasts out ant-like agents, called forward ants (FANTs), to look for paths to d . At their arrival in d , they become backward ants (BANTs) which trace back the path and update routing tables.

Some more recent algorithms attempt to limit the number of broadcasts like SARA for MANETs [22] and ESARA for Wireless Sensor Networks [23]; they adopt a controlled neighbor broadcast mechanism in route discovery to avoid flooding the network with FANTs. Each node broadcasts the FANT to its direct neighbors, and only one of them further rebroadcasts the FANT to its neighbors. However, there always still broadcasts messages.

Our protocol is based on ant general behavior, but differs from the other algorithms. We detail below the route discovery process of the previous version of our protocol in section 4 and the new version is section 5.

IV. A MULTI AGENT SYSTEM FOR GENERIC AND DYNAMIC NETWORKS

A. Abbreviations and Acronyms

The Generic multi-agent system that we propose to manage services in dynamic networks is composed of two types of agent communities (see figure 1). Both agent communities interact through different types of communication which we will present later on.

The first community of agents is SMA_Node, it will manage the functions of mobile nodes in the dynamic network. Each agent will be called node, then represent a network node whose features will be explained below. Each node agent will be able to provide a set of network data packets that are called packet. This set of data packets will be the second community of agents that will call SMA_Packet. Packet agents will travel along the dynamic network in a completely random way according to some metric contained in the node agents and packet agents at the interaction between these two types of agents. Moving a packet agent to a node agent network is defined by a behavior of packet agent which will be defined later.

Each node agent SMA_Node from community will be defined generically by the following behaviours:

- Detect_neighbors () return Liste_node: This function allows the node agent to be able to see all the node agents that are in its "scope" and with which it can directly communicate.

- Connect (v: node): This function allows the node agent to start a connection with its neighbor node V in order to establish a communication with it.

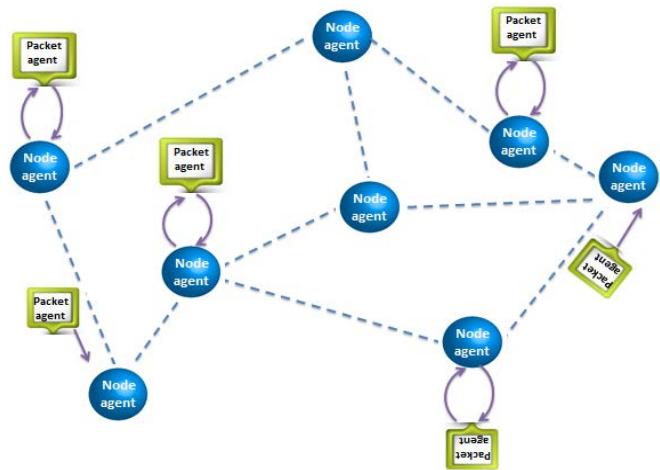


Fig. 1. General scheme of our model

- Disconnect (v: node): This function allows the node agent to delete a connection with a neighbor node. This function also has the effect of removing the node agent v from its list of neighbor nodes.
- Connect (): This function allows a node agent to join a network of node agents.
- Disconnect (): This function disconnects the agent from network.
- Generate () return packet: This function allows the agent to create and distribute a new packet agent in the network.
- Read_info (p: packet, info: information): This feature allows the agent to read and get information broadcasted and carried over the network by the packet agent p while moving from node to another.
- Write_info (p: packet, info: information): This feature allows the agent to write information into the packet agent p in order to be distributed in the network.
- Move (p: packet, n: node): This function allows the node agent to send the packet agent p to another node neighbor agent (node agent n). This the feature that allows the distribution of packet agents in the network.

Each agent from community SMA_packet will be generically defined by the following behavior:

- Create (): This feature allows the packet agent to be created by the creator node agent;
- Delete (): This feature allows the packet agent to be destroyed by its creator node agent;
- Transfert (n: node, info: information): This feature enables the packet agent to transfer the information carried to the current node where it is located. This

feature will be used by the current node agent to perform a read of information in the network;

- Update_packet (n: node, info: information): This function allows the packet agent to update its data with information provided by the current node agent. This feature will be used by the current node agent in order to broadcast information in the network;
- Choice_displacement (n: node) return node: This feature allows us to know randomly according to some metrics contained in the node agent n and in the current packet agent the next node where the current packet agent will move to;
- Move (init: node, final: node): This function allows the packet agent moving from the node agent « init » to the node agent « final ». The move action will be achieved through the move () function of init node agent.

B. Application of the model in the case of routing in ad hoc networks: previous version of AgentRouting protocol

The main idea of the protocol AgentRouting is to build a multi-agent based system where each node provides several kinds of agents (packet agents). Regarding the purposes of the routing task, we design two main types of packet agents. A first mobile agent, called Ant-Agent, is responsible of establishing routes. A second mobile agent, called Rectifier-Ant, is issued by a node whenever a change in the network topology is detected (the features of this agent are detailed in the previous version of the protocol). Our protocol is a complete multi-agent based system, where an agent works independently from the others. This fits spontaneous networks such as wireless ad hoc networks very well, because of the very high mobility and self-organization properties of this type of networks. Our protocol inherits from the advantages of this kind of model: autonomous work, distributed intelligence and robustness. Furthermore, the use of mobile agents allows to easily extend the functionalities of the protocol by simply adding other agents or by assigning other functionalities to the existing ones.

AgentRouting protocol is based on a hybrid algorithm. In the proactive phase, the protocol uses mobile Agent as follows: each node (Origin-Node) periodically creates one Ant-Agent (see function Generate () in section IV.A) that moves across the network from one node to another (see function Move (p: packet, n: node) in section IV.A) and builds paths from the current node to its Origin-Node and paths from the current node to the last visited one when the Ant-Agent returns back to its Origin-Node (the Ant-Agent has two phases: a Go-phase when it is sent by its Origin-Node and a Back-Phase when it returns to its Origin-Node).

When a data session is started between a source node s and a destination node d, s checks whether it has up-to-date routing information about d. If not, the node s makes a local route request. In our case, the route request is not broadcasted to every node as it is the case in a classic ant routing protocol [3,7,8], but it is stored on node s. The broadcast task is assigned to the Agent (packet agents created during the proactive phase and moving within the network) that have the responsibility to "intelligently" disseminate the route request throughout the network.

These Agents gather information about the quality of paths they followed, and at their arrival to node s (which contains the route request to destination d), they return back to their source node by tracing back the path and updating the routing tables (see function Read_info (p: packet, info: information) in section IV.A).

Before detailing the previous version of our routing protocol, let's consider some of these assumptions. Each node must be able to broadcast hello messages to its one hop neighbors. We also consider that the links between the nodes of the networks are always bi-directional. Moreover, as the protocol must operate in an ad hoc environment where nodes are highly mobile, the routing protocol must take into account this constraint and be responsive to these frequent changes of the topology. Therefore, our previous version of routing protocol is consisted of three modules: neighborhood discovery module (see function Detect_neighbors () in section IV.A), path discovery module and managing broken links module. In this paper, we focus our contribution on the path discovery module and data transmission.

1) Route discovery in the previous version of our protocol

a) Proactive phase: AgentRouting Protocol is a routing protocol for wireless ad hoc networks based on mobile Agent. In order to establish routes between nodes, our protocol uses mobile agents which are periodically created by each network node (see function Generate () in section IV.A). An Ant-Agent belongs to a single node called Origin-Node. An Ant-Agent moves across the network from one node to another (see function Move (init: node, final: node) in section IV.A). When it reaches a node, the Agent establishes and builds in its memory (see function Update_packet (n: node, info: information) in section IV.A) and in the routing table of the node, a path between this node and its origin node (Figure 2). Thereafter, the Agent chooses a next hop among its neighbors in a stochastic manner and proportionally to the amount of pheromone deposited by the other Agent during their Back-Phase (see function Choice_displacement (n: node) in section IV.A).

To avoid routing loops, we assign a unique identifier <node_ID, Ant_ID> to agents (see function Write_info (p: packet, info: information) in section IV.A), that is incremented at each creation of a new Ant-Agent. If a node receives several times the same Ant-Agent, it accepts the information given by the first one and ignores the others. In order to monitor its Ant-Agent, a node assigns a configurable Time To Live (TTL) to agents (see function Write_info (p: packet, info: information) in section IV.A) whose value is proportional to the network dimension and decremented on each hop. This means that an Ant-Agent will have two phases during its life cycle: Go-Phase (Figure 2) where the agent builds a path from the current node to its Origin-Node and a Back-Phase (Figure 3) where the agent follows a reverse path from the one followed during its first phase (the Ant-Agent saves in its memory a reverse path during its Go-Phase). At each visited node during the Back-Phase, the Ant-Agent builds and stores in the routing table of this node a path from this current node to the last node visited during the first phase (when TTL=0). This is the first step in the routing discovery process which is proactive.

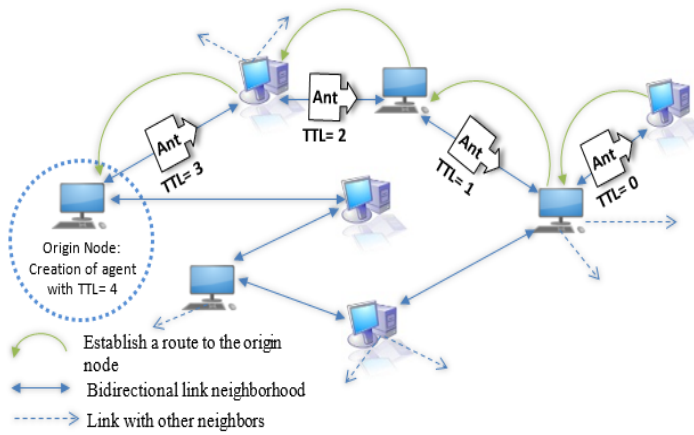


Fig. 2. Go phase

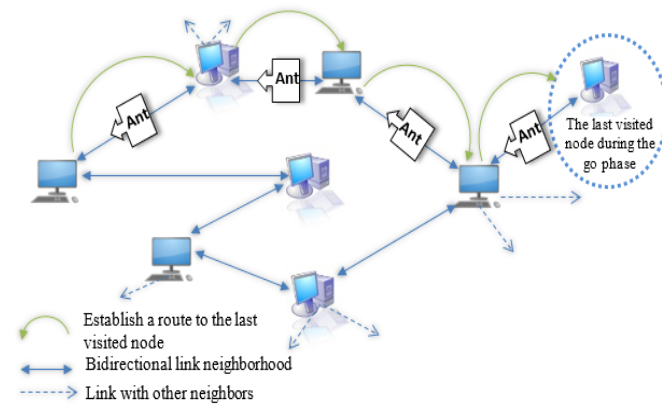


Fig. 3. Back-Phase

b) *Reactive phase:* During the proactive phase, a large number of paths are built. However, when a node d plans to send or forward data packets to an unknown destination node t (see figure 4), it triggers a route request process where the route request is locally saved. When an Ant-Agent, during its Go-Phase, visits a node which has made a route request (a node can have several route requests), the Ant-Agent switches to its Back-phase and deposits an amount of pheromone on each node of the reverse path towards its origin node. This mechanism is used to mark the paths towards the node d and hence informing the other nodes (and Agents) about this route request. The amount of pheromone deposited by the Ant-Agent is defined by the following equation (1):

$$\psi_{it} = \psi_{i(t-1)} + \Delta q \quad (1)$$

Where ψ_{it} is the pheromone level in the node i at time t and Δq is a positive constant.

During its Go-Phase, a Ant-Agent chooses the next hop in a stochastic manner and proportionally to the amount of pheromones; this process increases the likelihood to select a path towards a claimant node d without penalizing the other paths. Choosing the next node randomly and proportionally to the amount of pheromones allows us to increase the number of agents towards the claimant node d. On the one hand, this approach increases the chances of having an Ant-Agent issued from the destination node t (i.e., the Agent's source node is the

destination t). On the other hand, it allows to quickly reaching nodes (like node k in figure 4) that have already established paths towards this destination t.

c) *Stochastic data routing:* In our protocol, the nodes stochastically forward the Agent. When a node has several neighbors concerned by nodes that made a route request, it randomly selects one of them with the probability p (see function Choice_displacement (n: node) in section IV.A).

Each neighbor can have a quantity of pheromone related to nodes which made a route request. Let's consider N (i) the set of i's neighbors and ψ_{jt} the amount of pheromone associated to a neighbor j stored in the routing table of the node i at time t.

The expression that gives the probability p to select a next hop j from node i is defined in equation (2).

$$P = \psi_{jt} / \sum \psi_{kt} \quad (2)$$

In order to consider route requests in an equitable manner leading to a self-organizing system and a better management of frequent changes in the network topology, we propose to set up an evaporation process. The latter allows to no longer take into account the old route requests already satisfied. At each time interval, the amount of pheromone corresponding to each route request is decreased as defined in equation (3):

$$\psi_{it} = (1-\alpha) * \psi_{i(t-1)} \quad (3)$$

Where ψ_{it} is the amount of pheromone related to a claimant node d, stored in the node i at time t and α is a real ($0 < \alpha < 1$) (we choose $\alpha = 0.1$ for our simulations).

V. NEW FEATURES FOR OUR PROTOCOL

The first version of our routing protocol is published in [4]. Additionally considering our routing protocol as a completely mobile multi agent system, we implemented other functionalities in order to improve the network lifetime and the routes discovery.

Our new version strives to avoid low energy routes during route discovery phase and data transmission, so that the network lifetime can be prolonged. Thereby, we added other features in order to improve the previous version in terms of packet loss and end-to-end delay.

A. Adapting pheromone quantity to the link cost and the energy

In the previous version of our protocol, when an Ant-Agent visits a node d (see figure 4) which has made a route request, this agent switches to its Back-phase and deposits an amount of pheromone on each node of the reverse path to its origin node, the goal is to mark the paths towards the node d, and therefore inform the other nodes and Agents about this route request. The amount of pheromone deposited by the Ant-Agent in the previous version of our protocol is constant (Δq in equation 1). In the new version, we adapt the quantity of pheromone deposited by ant agents according to the limited energy of nodes and the link cost.

Each node continuously updates its energy and exchanges 'hello' messages with neighbors to establish and maintain a neighbor list containing updated information such as ID and energy level [24].

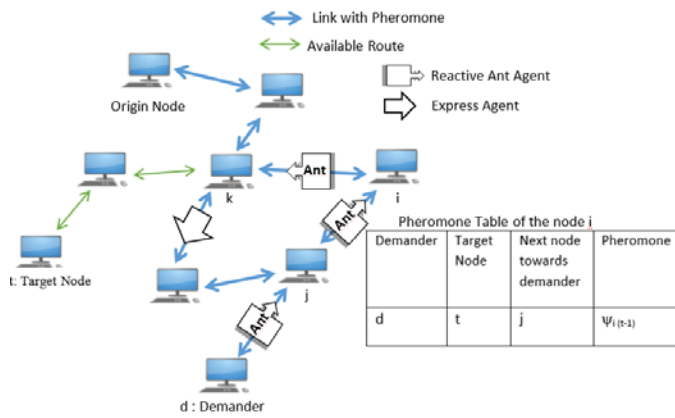


Fig. 4. Reactive phase

In figure 4, Ant-Agent “Ant” is going towards its origin node after visiting a demander node (node d); it deposits an amount of pheromone on each node of the path. When this Ant moves from node j to node i, it deposits a pheromone quantity Δq on the node i. The new value of Δq is:

$$\Delta q = C(i,j,d) * E_j + q, C(i,j,d) = 1 / (1+n) \quad (4)$$

Where $C(i,j,d)$ represents the link cost between node i to its neighbor node j towards destination d (node which has made a route request). This cost is based on the number of times (n) the link was previously used. E_j is the energy of node j. q is a positive constant (we have chosen $q=0.1$ for our simulations).

The Ant-Agent continues in a similar fashion until it reaches the origin node.

We have previously seen that an Ant-Agent during its Go-Phase chooses the next hop in a stochastic manner and proportionally to the amount of pheromones (see equation 2). With the new version of protocol, the probability to choose a neighbor with more energy and lower link cost is higher. The advantages of this new feature are:

Network lifetime[25-29] is one of the most prominent barriers in deploying ad hoc networks for large-scale applications. Our new algorithm helps to enhance the lifetime of the network by taking care of nodes energy. This feature allows to increase network lifetime and to control the network congestion through load balancing.

The link cost parameter helps the Ant-Agents during their Go-Phase to use different paths towards the claimant node; this approach increases the probability of having an Ant-Agent issued from the destination node, or to quickly reaching nodes that have already established paths towards this destination.

B. A new type of agent: Express agents

This simple new feature must use another type of Agents called Express-Agent. When an Ant-Agent switches to its Back-phase after visiting a claimant node, it can pass through a node which has requested the route (like node k in figure 4). This node generates a special Agent called Express-Agent which will be immediately sent to the claimant node. This approach allows improving the end-to-end delay and the packet delivery fraction, because the data packets are sending immediately towards their destinations before the topology

changes. An Express-Agent use the same method detailed in section V.A when it goes towards the claimant node.

The rest of the route discovery procedure is similar to previous version of our protocol

C. A new data routing procedure

Once the routes are discovered, they are selected on the basis of number of hops, energy of the node, and link cost. When multiple routes are discovered using a reactive and proactive phases then the route is selected on the basis of probability procedure. The route selection in our new version of routing protocol differs from the classic ant methods inspired from Ant-Colony-Optimization algorithm. Our protocol introduces Utility of route metric in route selection criteria (Utility (n, n_j, d)).

$$Utility(n, n_j, d) = (1-k) * \frac{1}{1+NH} + k * Cost(n, n_j, d) \quad (5)$$

Where NH is the number of hops between node n and destination d, n_j is the neighbor node of n. The parameter k is used for providing greater importance to the number of hops or the cost of route “Cost”. Cost depends on the cost link C (n, n_j, d) (see equation 4) and the energy e of node n_j. The metric Cost can be calculated as

$$Cost(n, n_j, d) = \frac{C(n, n_j, d) * e}{\sum_{k=0}^{k=M} C(n, n_k, d) * e} \quad (6)$$

M is the number of adjacencies of node n.

The probability to pick a neighbor node n_j for routing data packets towards destination d is given by (7), which is based on Utility of route (energy, link cost, and number of hops):

$$P(n, n_j, d) = \frac{Utility(n, n_j, d)}{\sum_{k=0}^{k=M} Utility(n, n_k, d)} \quad (7)$$

This new algorithm strives to avoid low energy routes and optimizes the routing process through selection of the shortest path (number of hops) with more energy and lower link cost.

VI. SIMULATIONS RESULTS AND DISCUSSION

For performance evaluation of our proposed algorithm with existing protocols: ARA and AntHocNet (a classic ant based routing protocol), SARA (a more recent ant based routing protocol) and the old version of our protocol, we used NS 2 simulator by considering simulation parameters as shown in Table 1. The traffic is automatically generated by a Tcl script which is developed for this purpose. This traffic is randomly generated (the communications are established by randomly choosing pairs of nodes).

The performance metrics used to evaluate the performance of each algorithm are the following:

- The packet loss: this metric measures the number of packets which are not delivered to their destinations;
- The end-to-end delay: this metric represents the average delay between the packet sending time and its reception time;
- The total size of control messages generated by a protocol ;

- Total energy consumed;
- The network lifetime.

TABLE I. SIMULATION PARAMETERS

Parameter	Value
Simulation time	500s
Maximum node speed	30m/s
Sending frequency of Ant-Agent	0.8s
The evaporation frequency	0.5s
The evaporation rate « α »	0.1
The updating value of pheromone « q »	0.1
Packet size	512 bytes
Number of nodes	from 60 to 100 nodes
Number of data packets	from 700 to 1000 packets
MAC Type	IEEE 802.11
Simulation area	600m*600m
Protocol	UDP
Mobility model	Random way point
Receiving range	100m
Propagation model	Two ray ground
Initial energy of node	5J

We have calculated these metrics by varying, during the simulation time:

- The number of nodes : the number of data packets is fixed at 1000 and the maximum speed of the nodes is 5 m/s;
- The number of data packets : the number of nodes is fixed at 100 and the maximum speed of the nodes is 5 m/s;
- The nodes speeds: the number of data packets is fixed at 1000 and the number of nodes is fixed at 100.

In case of the performance metrics, total energy consumed and network lifetime, the number of data packets is fixed at 1000, the number of nodes is fixed at 100 and the maximum speed of the nodes is 5 m/s. The performance metrics are obtained by averaging over 30 simulations run from one source to one randomly selected destination. We assume that a node consumes 0.001J while receiving and 0.003J while transmitting.

A. The packet loss

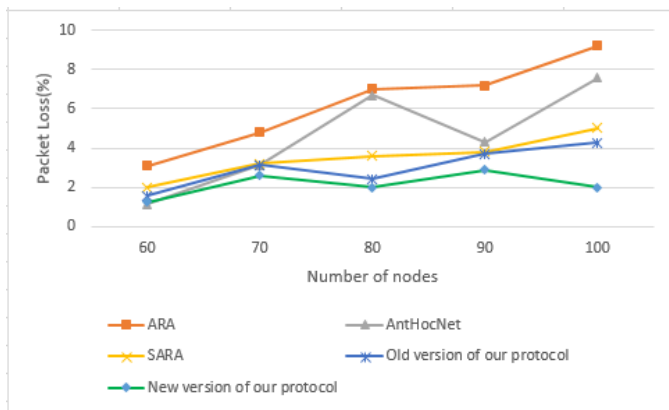


Fig. 5. Packet loss ratio according to the number of nodes

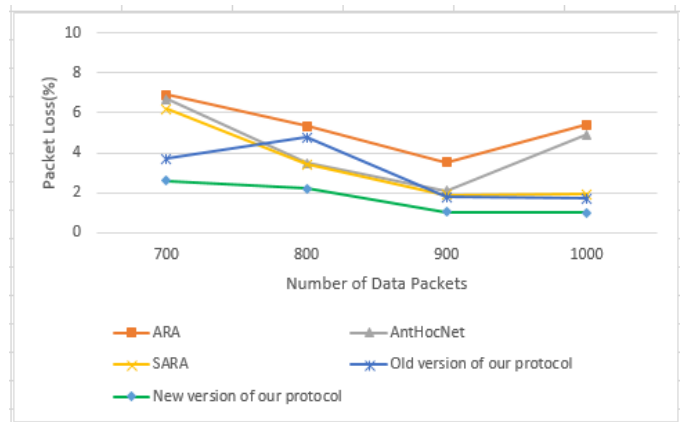


Fig. 6. Packet loss ratio according to the total number of data packets

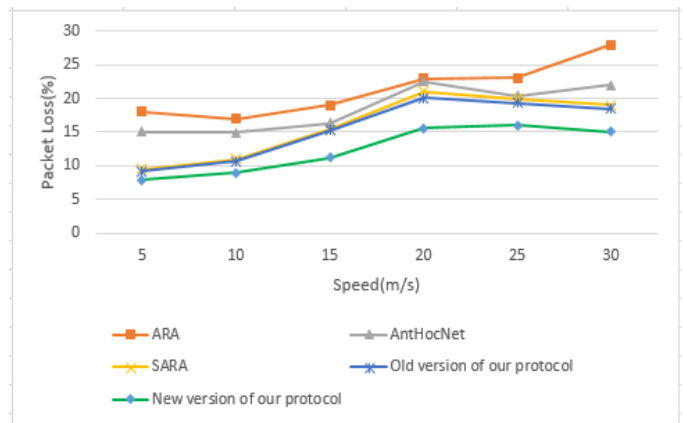


Fig. 7. Packet loss ratio/nodes speed

Figure 5 and 6 show the variation of % packet loss as function of the number of nodes and the total number of data packets. It is clearly shown in these figures that our scheme have less packet loss than the other protocols.

In ARA and AntHocNet, the number of lost packets increases very quickly in networks that have more than 90 nodes and in networks with high traffic (more than 900 data packets). We can explain these results by the fact that the ARA and AntHocNet protocols use a broadcast mechanism that generates a very important overhead and this fact is aggravated by the proliferation of collisions, which overloads the system and generates a very large number of non-accomplished transmissions. In SARA protocol, there is less packets loss because it uses less broadcast control messages than ARA and AntHocNet (see section Routing issues in ad hoc network and related work). In the case of our protocol, these results can be explained by the fact that we use an efficient reactive route discovery procedure instead of using a broadcast mechanism.

Simulation results showed that the new version of our routing protocol performed better than the previous version in terms of packet loss; our new algorithm favors nodes with high energy level and avoids the critical nodes to participating in the route discovery and data packet transmission. This produces fewer broken links and dramatically reduces the packet losses.

The link cost parameter helps to control the network congestion through load balancing, so the number of buffer overflow in the queue of intermediate nodes is reduced (see section: Adapting pheromone quantity to the link cost and energy). As result, the claimant node can have different routes towards the destination node so the claimant can choose the best one in terms of node energy; link cost and number of hops (see equation 7). This concept also helps the claimant node to have alternative routes.

Figure 7, which represents the variation of % packet loss considering the variation of nodes speed. It shows that our protocol have less lost packets than ARA, AntHocNet and SARA protocols when we vary the nodes speed. The use of Express-Agents in the new version allows sending data packets immediately to the destination before topology changes, hence reducing the packet losses.

B. The end-to-end delay

Figures 8, 9 and 10 show the end-to-end delay in each protocol, considering respectively the number of nodes in the network, the total number of packets and the nodes speed. We can see that our protocol and the SARA protocol generate less important delays than those generated by the ARA and AntHocNet protocols.

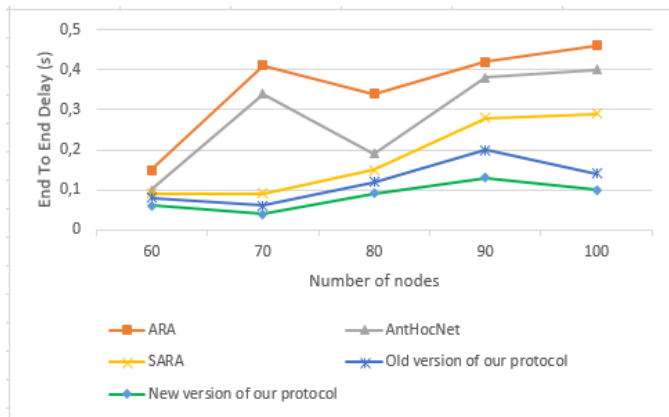


Fig. 8. End-to-end delay (s) according to the number of nodes in the network

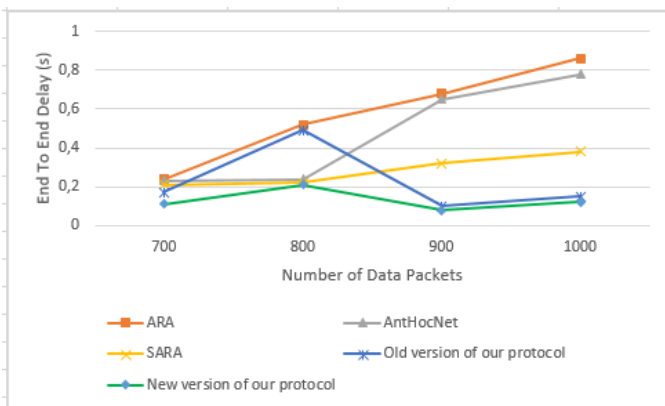


Fig. 9. End-to-end delay (s) according to the total number of data packets

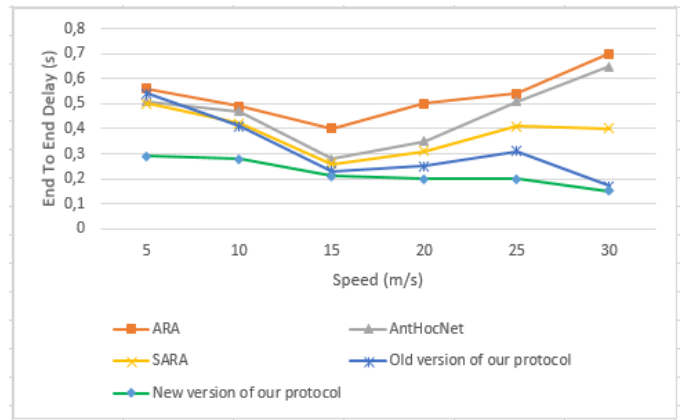


Fig. 10. End-to-end delay (s)/speed (m/s) of nodes

Our protocol and SARA protocol are both more efficient than ARA and AntHocNet protocols, especially in the case of a large number of nodes and in the case of high nodes speeds. Regarding our protocol, this may be due to five main reasons:

- Since our protocol is multipath, it allows each node to have several paths (towards the same destination) whenever it attempts to send a data packet, so it selects the best one (see equation 7 for the new version).
- The hybrid character of our route discovery scheme significantly reduces the transmission times. In fact, thanks to the proactive phase, it is not necessary to make a route request each time; the route may exist in the routing table;
- Unlike ARA, AntHocNet and SARA protocols, our protocol does not require, during the reactive phase, a broadcasting technique; in fact, the latter exponentially increases the routing overhead and thus overloads the network; consequently it delays the paths establishment and the data packets delivery process ;
- Once the size of the network increases, the new version of our protocol produces a better delay. The reason is that our protocol favors nodes having high energy level and avoids the critical nodes to participating in the route discovery and data packet transmission. This produces fewer broken links and greatly reduces the end-to-end delay ;
- The link cost parameter allows to improve the network congestion, so the number of data packets in the queue of intermediate nodes is reduced. This concept reduces the end to end delays.

C. The communication overhead

Control packets are the primary source of energy consumption. Less number of packets transmitted between intermediate nodes for route discovery procedure is an effective way to save energy.

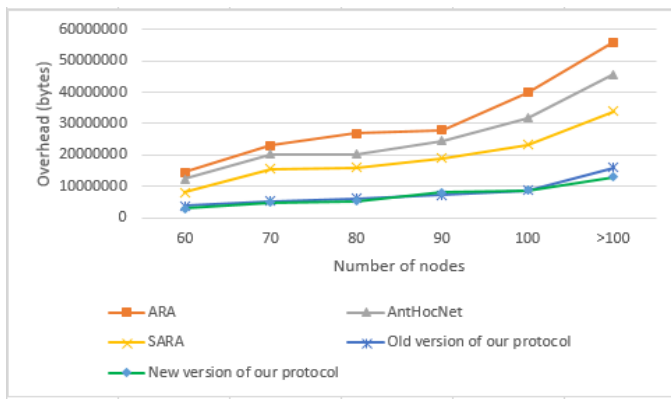


Fig. 11. The communication overhead (total size (bytes) of control packets) according to the number of nodes in the network

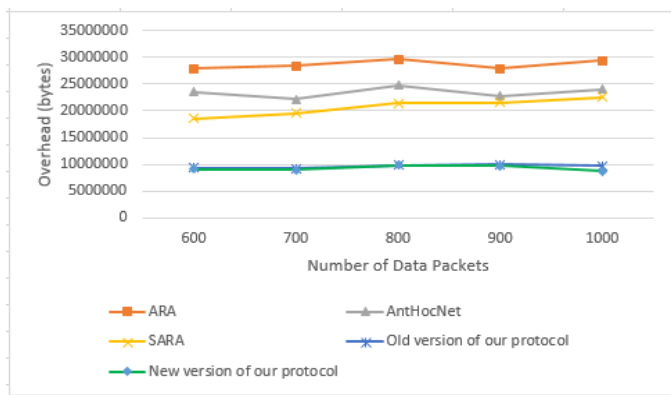


Fig. 12. The communication overhead (total size (bytes) of control packets) according to the total number of data packets

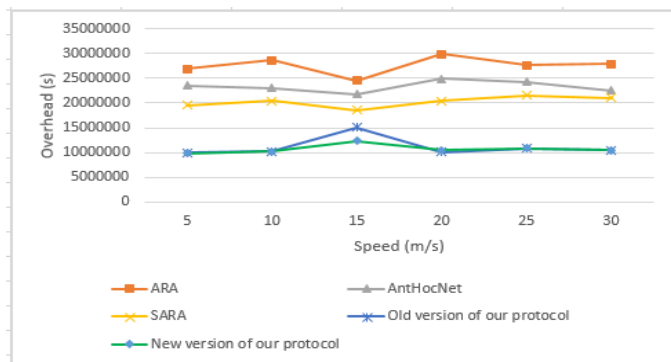


Fig. 13. The communication overhead (total size (bytes) of control packets) /speed (m/s) of nodes

The packets that consume much bandwidth and energy are those issued during the path discovery and maintenance phases. Therefore, we have measured the use of control messages by each studied protocol during these phases. Figures 11, 12 and 13 show the communication overhead in terms of, respectively, control packets size according to the number of nodes, the node activities and the speed of nodes.

We can see an important difference between our protocol and both ARA and AntHocNet protocols. The number of control packets increases slowly and linearly in our protocol, either by increasing the number of nodes or by increasing the network traffic, while it increases rapidly in the case of ARA

protocol. This could be explained by the fact that in the case of our protocol, the number of agents is managed and controlled by each node and this number is still proportional to the number of nodes in the network and to the transmission frequency of agents. Besides, we avoid the broadcast technique that generates a lot of overhead. Instead, we use a more accurate and a more intelligent technique which only makes use of the available agents in the network. In ARA protocol, the number of control packets depends on many factors, including the number of route requests that consume much bandwidth, since they require a significant number of broadcasts. Moreover, we can expect an important number of collisions due to the broadcast technique and network density. The collisions lead to an increasing number of retransmissions, which consequently induces the increasing of the total number of packets in the network.

Figures 11, 12 and 13 show also that our protocol generates less overhead than SARA protocol. This proves the effectiveness of our protocol, since the SARA protocol is known to be efficient in terms of communication overhead.

The total size of control messages in the previous and the new version of our protocol are approximately the same. Each node periodically creates an Agent, thus the number of agents in the network can continually be controlled, when a connection is planned to be established by a node with another one where there is a lack of a routes in its routing table, we only use the available agents in the network to satisfy this route request. We use the same process in the two versions of our protocol.

The adding of Express-agents has no influence on the communication overhead because the size of these agents is very small. Thus, we improved some parameters in the new version like the sending frequency of Ant-Agent (0.8s, see table 1, the previous value was 0.5s).

D. Total energy consumed

The control packets are the primary source of energy consumption. This is demonstrated in Figure 14 in a different way by figuring out the total energy consumption of the overall network. This figure justifies the facts that less number of control packets transmitted between nodes is an effective way to save energy. ARA, AntHocnet and SARA use more control packets, therefore more energy is consumed comparing to our protocol.

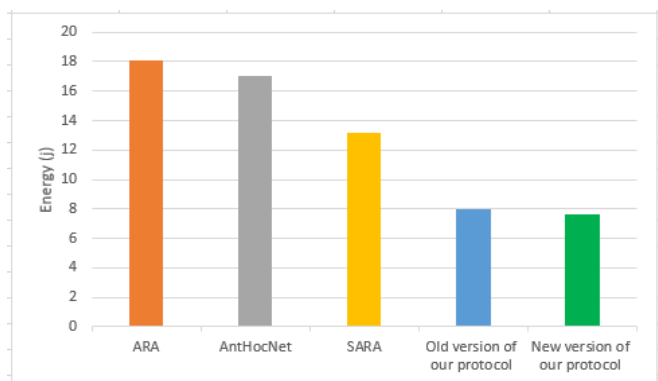


Fig. 14. Total energy consumed

E. The network lifetime

The network lifetime can be defined in three ways [30]: the time taken to exhaust the battery of the first network node, the time taken to exhaust the battery for N network nodes, and the time when the battery of the last network node is exhausted. We choose the second way.

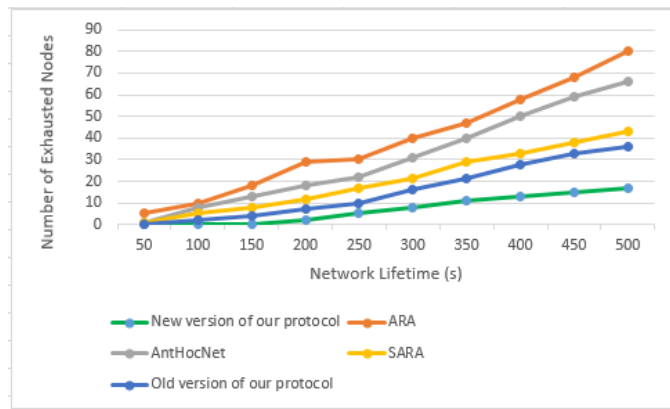


Fig. 15. Network lifetime versus number of exhausted nodes

The network lifetime metric is shown in figure 15 with the number of network nodes equal to 100. The network lifetime of our protocol is longer than those of ARA, AntHocNet and SARA. As result, the new version of our protocol exhausts fewer nodes compared to the previous version, which noticeably increases the network lifetime.

As demonstrated in Figure 15 and seen previously in the new protocol version description, our new algorithm balances the energy amongst all the nodes and expands the individual node lifetime, hence the entire network lifetime. Our protocol also avoids the broadcasting technique which consumes a lot of energy and increases the number of exhausted node.

VII. CONCLUSION AND FUTURE WORK

We have presented in this paper an improved version of our ant based routing protocol for Ad hoc networks, that aims to reduce the packet losses, the transmission delays and to maximise the network lifetime by using a new route discovery technique and data transmission. Considering the simulation results obtained using the new proposed algorithm, we can argue that the new version of our protocol significantly reduces the packet losses and transmission delays. Regarding routing overhead, our algorithm out performs SARA protocol which is assumed to generate fewer messages than ARA and AntHocNet. Our protocol doesn't require any broadcasting mechanism that leads to increase the number of control messages. This underlying concept extends the network lifetime and improves energy consumption when compared with other solutions known in the literature regarding network lifetime and network congestion. Also, our algorithm allows achieving better results than ARA, AntHocNet and SARA. It is capable to avoid nodes with low energy and high cost link in the multipath construction. In other words, paths with higher energy are identified and selected for transmission.

In our future work, we plan to use our generic multi-agent system within another dynamic environment such as Wireless

Sensors Networks (WSN) and P2P networks. In WSN, a new Simple Ant Routing Protocol for Wireless Sensor Networks (SARPWSN) inspired from ant colony optimization will be designed using our multi-agent system. In P2P networks, we will describe the design and experimentation of a new Anonymous and confidential P2P system, the protocol works on the principle of social networks with nodes identified by virtual addresses and uses a set of control messages that work like a Multi Agents Mobile System.

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High Lightweight Encryption Standard (HLES) as an Improvement of 512-Bit AES for Secure Multimedia

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Abstract—In today's scenario, people share information to another people frequently using network. Due to this, more amount of information are so much private but some are less private. Therefore, the attackers or the hackers take the advantage and start attempting to steal the information since 2001. the symmetric encryption algorithm called 512-bit AES provides high level of security, but it's almost be impossible to be used in multimedia transmissions and mobile systems because of the need for more design area that effect in the use of large memory space in each round and the big encryption time that it takes. This paper presents an improvement of 512-bit AES algorithm with efficient utilization of resources such as processor and memory space. The proposed approach resists the linear and differential encrypt analysis and provides high security level using a 512-bit size of key block and data block and ameliorates the performance by minimizing the use of memory space and time encryption to be able to work in specific characteristics of resource-limited systems. The experimental results on several data (text, image, sound, video) show that the used memory space is reduced to quarter, and the encryption time is reduced almost to the half. Therefore, the adopted method is very effective for encryption of multimedia data.

Keywords—Advanced Encryption Standard (AES); Encryption; multimedia data; security; resource-limited systems

I. INTRODUCTION

Cryptology is the mathematical science of secret writing. It is made up of two halves: cryptography and cryptanalysis. Cryptography consists of the techniques for creating of secret writing. it uses mathematics to encrypt and decrypt data. Cryptanalysis encompasses the techniques of breaking that secret writing. it is the study of encrypted information in order to discover secret, Cryptanalysts are also called pirates. Over the past 2,500 years, cryptology has developed numerous types of systems to hide messages and subsequently a rich vocabulary in which to describe them. Cryptography consists of two parts: Symmetric Key Encryption and Public Key Encryption.

In Symmetric Key Encryption, both the sender and the receiver share the same key used for both encryption and decryption of the data. In fact, the two keys may be identical or trivially related.

In Public Key Encryption, two different keys related mathematically are used. Public key encryption encrypts data

using the recipient's public key, and it cannot be decrypted without using a matching private key, and one key cannot be used in the place of the other [1].

The focus of this paper is stationed in the field of Symmetric Key Encryption. The improved AES algorithm that use 512 bit of key and data block provides high level of security, because of the use of key size larger by four times than the original 128-bit AES key [2]. However, it has a deficiency in performance, the encryption process become heavy when it comes to the modern communicating world that depends on the resource-limited systems and real-time operations

The main objective of this work is to design an alternative algorithm that keeps the level of security that the 512-bit AES provides, and minimizes the cost of memory and encryption time that it takes. The rest of the paper are details explain the proposed work as follow: Section 2 speaks about the original AES algorithm with more details and a given algorithm called 512-bit AES that was appeared to provide more security. Section 3 talks in details about the proposed algorithm that is given as an alternative to the one that called "512-bit AES" for improving the performance level, explaining its transformations methods including the general architecture and key expansion. In section 4, we talk about the tests that we have done and the results that prove the amelioration of the performance level. Finally, section 5 concludes the paper with a general thought about HLES algorithm.

II. RELATED WORK

The proliferation of digital communications, multimedia and the transition of social interactions into the cyberspace have raised new concerns in terms of security, trust and performance. The selection of a suitable crypto-algorithm will dynamically affect the lifespan and performance of a device in terms of battery-life, hardware memory, computation latency and communication bandwidth. To select a suitable cryptographic algorithm for an application or an environment, the understandings of both the algorithmic requirements in terms of hardware and the specifications of the development platform intended has to be established.[3].The National Institute of Standards and Technology (NIST) established a powerful encryption algorithm for symmetric encryption procedures called Advanced Encryption Standard. AES is a Federal Information Processing Standard (FIPS) that has three variable

key lengths but block length is fixed to 128 bits. The three key sizes of AES are 128, 192 and 256 bits. [4]. Their number of possible keys is 2127, 2191 and 2255 respectively. [5]. AES was designed to have very strong resistance against the differential cryptanalysis and linear cryptanalysis attacks since it used Wide Trail Strategy in its design. [6]. The United States government organizations has approved for the AES to be used to protect sensitive, unclassified information. It was also widely adopted both commercially and globally. [5]. To enhance the reliability of the algorithm against the brute force attack and provide more security, A new algorithm structure appeared in 2011 that is similar to the original AES algorithm, but having slight difference is that the plaintext size and key size use input of 512 bit instead of 128 bit. It consist of the same four major operations of the original AES performed during each round: byte substitution, shifting rows, mixing columns and finally adding the round key. However, they are designed to be bigger than the original AES operations by four times to be able to occupy the 512 block and key size. this version of AES can be used when higher level of security throughput are required, because more security comes from using larger key size, and more throughput comes from using four times larger block size that the block size used in the original AES. [2]. However, in the modern communicating world with all its ways including the wireless mobile systems and videos that generally process a large amount of data and require real-time operations, there is limited processing power, memory and bandwidth, and is rarely able to provide such security level and handle the heavy encryption-processing load in same time. Real Time Applications (RTA) are application programs that function within a specific timescale. Voice over IP (VoIP) and video conferences are examples of RTA. Transmitting such data via open networks is risky. However, any security must be lightweight and cause no delay. Recently, many algorithms have been created (RC4 (Rivest Cypher 4), RC5 (Rivest Cypher 5), RSA (Rivest Shami Adleman) ...), but very few are viable with RTA. [7]. In addition, it is difficult to use encryption techniques directly in multimedia data because of the large volumes, and require real-time interactions. Therefore, there is a need of efficient and light encryption algorithm as an alternative, which can provide better security and performance. [8]. Based on the state of multimedia encryption, can observe that:

- For complete and provable security of the video data for example, the entire video needs to be encrypted. However, a naive encryption of the complete video stream is computationally slow.
- To solve the problem of speed, there is a need of finding solutions to the naive encryption.
- The traditional naive encryption methods use conventional AES algorithm. There is a need of modification in the algorithm to reduce the time required for encryption and increase the security level.
- The encryption algorithm should not be susceptible to attacks like known plaintext attack and cipher-text-only attack. Computational efficiency should not come at the cost of security.

III. PROPOSED APPROACH

The aim of this paper is to present a new light algorithm called High Light weight Encryption Standard (HLES) that can be used when higher level of security throughput are required, and processing power, memory space and bandwidth are limited (case of multimedia transmissions and mobile systems).

The HLES algorithm uses a key size of 512 bit and data block size of 512 bit. Both of key and data block are divided to 4 blocks of 128 bit for each. Each set of those four 128-bit blocks will be encrypted using an encryption functions that produce a set of four 128-bits encrypted blocks using four 128-bits encryption keys. The encryption process use each one of the four key parts to encrypt the four data block parts respectively but not in same time so that the memory consumption can be minimized. The four encrypted parts are not independents because each part needs the previous one to continue the encryption process (except the first one because it has no previous part).

The proposed algorithm called HLES has four main different transformations. The first transformation is the byte substitution that substitutes the value of 16 bytes and this is achieved via using parallel S-Boxes.

The second transformation is the (SH-Z) function that translate bytes (data block and key block) to bit-stream block and calculate the number of zeros exist in the bit-stream key block, then shift the bit-stream data block to the left side as the calculated number of zeros, after that it retranslate the bit-stream block to bytes data block.

The third transformation is a normal XOR function applied with 128-bit key part and 128-bit data block part. The final transformation has the same role of the previous one (XOR function) but this one is applied with the current part and result of previous part.

A. Architecture

The placement of functions is designed and chosen carefully to guarantee a lower cost of computational resources and good performance. In addition, its architecture provides high level of security because of the 512-bit key length and a special coordination and synchronization between its functions. The encryption process and functions placements are shown as a flowchart in the figure (Fig.1).

- Bytes substitution :The 512 bits input plaintext is divided into four 128-bit parts. Each part is organized in array of 16 bytes that are substituted by values obtained from substitution boxes. The S-Box used in the proposed algorithm is the same one used in the AES algorithms in case there is no need to generate a new one because of its proven effectiveness. This is done to raise the security level according to diffusion-confusion Shannon's principles for cryptographic algorithm design [9]. The S-Box is shown in the figure (Fig.2).

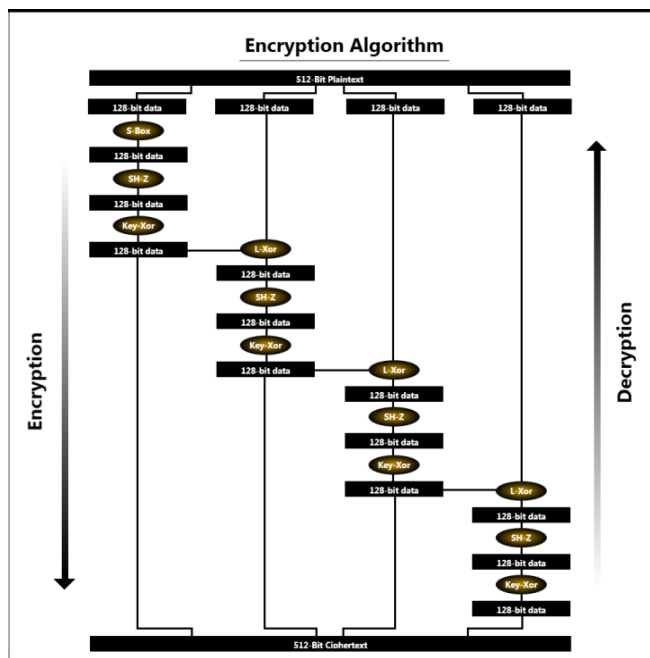


Fig. 1. HLES General Architecture

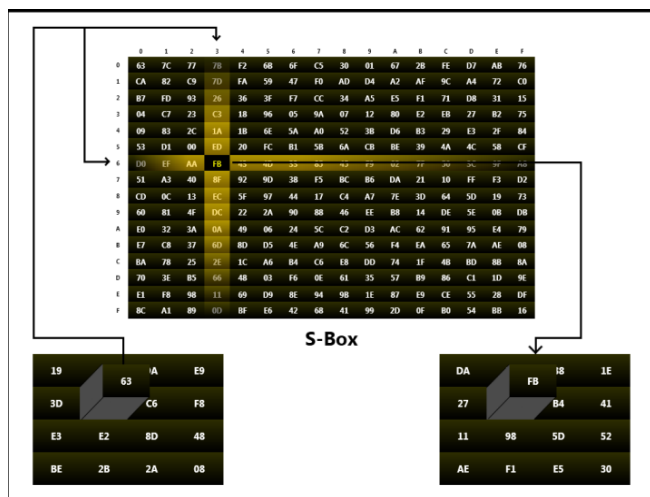


Fig. 2. Byte Substitution

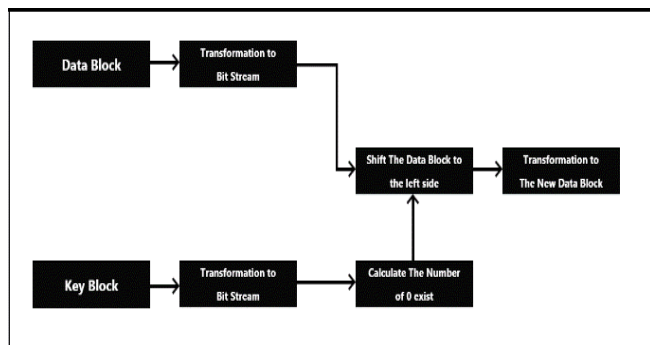


Fig. 3. SH-Z Bit stream shift function

- SH-Z function: This function depends on the number of zeros exist in each 128-bit part of key applied in the block. First, two matrix of bytes (data block part and

key block part) are translated to bit-stream blocks. Second, number of zeros exist in the bit-stream block of key part is calculated. Then, bit-stream block of data is shifted to the left side as the calculated number of zeros. Finally, that bit-stream block of data is retranslated to a result matrix of bytes. The figure (Fig.3) explain the procedure.

- Key-Xor and L-Xor functions: Both functions are EXCLUSIVE-OR functions (called EXCLUSIVE DISJUNCTION functions in some references) the difference between them is the inputs that they use. Key-Xor function depends on the outcome of each process of HS-Z function and each 128-bit key part block to produce a part of cipher text, this function is applied on all the four block parts as shown in the flowchart of the HLES general architecture. L-Xor function depends on the previous outcome of the Key-Xor function and the current block part, it is applied on the last three block parts, because the first one has no previous outcome of HS-Z function.

B. Key expansion and rounds

The original AES uses a key whose length is 128, 192 or 256 bits. The cipher key is expanded to into 10, 12, or 14 round keys, respectively, using the Key Expansion Algorithm, where each round key is of 128 bits. This Key Expansion algorithm depends only on the cipher key, and is independent of the processed data. It can therefore be executed in dependently of the encryption / decryption phase. The round keys are the combination of transformations SubWord (RotWord(tmp)) and SubWord (tmp) and the use of the RCON value. The AES Key Expansion algorithm is described by the pseudo code in the Figure (Fig.4) (the pseudo code is written in terms of double words). [10].

```

parameters
Nb = 4
Nk = number of doublewords in the cipher key (4, 6, 8 for AES-128, AES-192, AES-256, resp.)
Nr = number of rounds in the cipher (Nr = 10, 12, 14 for AES-128, AES-192, AES-256, resp.)

The KeyExpansion routine
KeyExpansion(byte key[4*Nk], word w[Nb*(Nr+1)], Nk)
begin
word tmp
i = 0
while (i < Nk)
w[i] = word(key[4 * i], key[4 * i + 1], key[4 * i + 2], key[4 * i + 3])
i = i + 1
end while
i = Nk
while (i < Nb * (Nr + 1))
tmp = w[i = 1]
if (i mod Nk = 0)
tmp = SubWord(RotWord(tmp)) xor RCON[i / Nk]
else
if (Nk > 6 and i mod Nk = 4)
tmp = SubWord(tmp)
end if
w[i] = w[i - Nk] xor tmp
i = i + 1
end while
    
```

Fig. 4. Key expansion pseudo code

Through research and analysis of 128-bit AES key generation and extension mechanism, only if the attacker gets the wheel of the sub-keys, he can deduce all the sub keys by the AES key generation expansion mechanism. As we can find a new way which can generate sub keys from front to back quickly, but the reverse derivation of the keys causes difficulties. It can both increase the difficulty of brute force AES and can effectively prevent the weaknesses of a variety of AES key expansion attack, and will not affect the speed of its

current run. [11]. In HLES algorithm, the key is divided to four parts of 128-bit that are expanded to occupy 10 rounds. Therefore, the key expansion algorithm that the original 128-bit AES use is enough since it was and still until now one of the most strong key expansion algorithms against the related key attacks. [12].

IV. RESULTS AND EVALUATION

In order to compare between the three encryption algorithms (128-bit AES, 512-bit AES and HLES), there designs were coded in same operating system (Microsoft Windows 7), same programming platform (Visual Studio Platform), same environment (WPF environment) using same programming language, which is C# language. The used operation mode is the ECB mode (Electronic CodeBook), which is the simplest one. We used five data blocks of different memory capacities to be encrypted, (text file, JPG image, PNG image, MP3 sound, MP4 video). First one is text file of 5.8 Kbytes, second one is JPG image of 25 Kbytes, third one is PNG image of 59 Kbytes, fourth one is MP3 file of 105 Kbytes (sound file), and the last one is MP4 file of 219 Kbytes (video file), and we calculated there encryption time. The synthesis results are shown in the table (TABLE I):

TABLE I. COMPARISON OF THE THREE ALGORITHMS (128-BIT AES, 512-BIT AES, HLES)

	128-bit Algorithm	512-bit Algorithm	HLES Algorithm
TXT file (5810)	81	167	64
JPG file (25000)	280	566	228
PNG file (59000)	871	1352	577
MP3 file (105000)	1961	2314	1010
AVI file (219000)	4842	5389	2529

The results show that the proposed algorithm (HLES) encrypts the data spending time less than what the original 128-bit AES spend, and almost the half time that the 512-bit AES spend, and the difference is getting bigger in each time the data gets bigger. The figure (Fig.5) presents the encryption time for the three algorithms, 128-bit AES (in Red), 512-bit AES (in Green), HLES (in blue).

When it comes to the criteria of the memory space, there are two ways to effect an ideal comparison. The first one is to measure the memory space spent by all the functions of the encryption process, in this case, to encrypt one data block of 512-bit size, the 512-bit AES uses four functions that occupy together 2048 bits (512-bit x 4). While the area design of the proposed algorithm (HLES) allows to occupy only 1536-bits (128-bit x 3 x 4), which is less than the other one by 1/4. The second one is to measure the memory space spent by each function of the encryption process. In this case, we have to take in consideration the size of data block that each function consume at the same time for each algorithm. In 512-bit AES, all the functions work with 512-bit data block size. While in HLES algorithm there are functions that work with 128 bit

block size and functions that work with 256 bit block size (functions that need two 128 bit blocks to produce one 128 bit block), but generally HLES algorithm has no function that consume more than 256 bit.

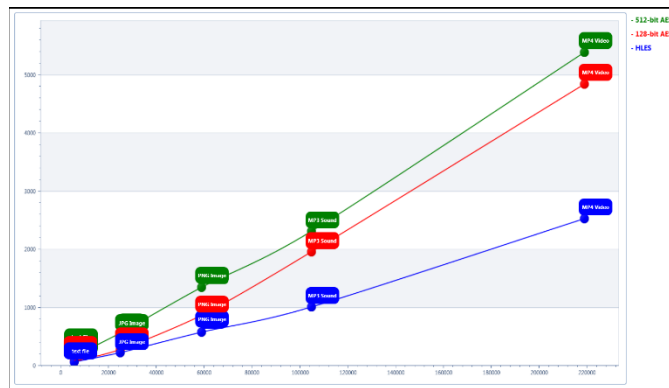


Fig. 5. Comparison of the three algorithms in encryption time using five kinds of files (text file, JPG image, PNG image, MP3 sound, MP4 video)

The proposed algorithm (HLES) uses memory space less than the other one that 512-bit AES uses, which makes it acceptable to be used in resource-limited systems.

V. CONCLUSION

In this work, we proposed a light symmetrical algorithm as an alternative to the 512-bit AES that provides high security level using a 512-bit size of key block and data block, and resisting the brute force attack with efficient utilization of resources such as processor and memory space. HLES ameliorates the performance by minimizing the consumption of memory and time encryption to be able to work in specific characteristics of resource-limited systems. The experiments show that the memory usage is minimized by one quarter of the one that the algorithm called 512 AES uses, and the encryption time is reduced almost to the half, which makes it near in performance to the original 128-bit AES. The future work will be focused on promoting operation speed of key expansion to ensure a good synchronization between key generating and encryption process. Moreover, all possible key attacks will be tested and examined.

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A Distributed Framework for Content Search Using Small World Communities

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Abstract—The continuous growth of multimedia content available all over the web is raising the importance of a distributed framework for searching it. One of the important parameters in a distributed environment is system response time. This parameter specially plays an important role in search and retrieval. A novel two-tier structure is introduced in this paper, which focuses on the community concept to facilitate creation of ontological small worlds that can effectively assist the search task. As a result, user queries are forwarded to nodes that are likely to contain the relevant resources. Evaluation of the framework proves that the small world character of the proposed structure provides queries with better route selection and searching efficiency.

Keywords—Small-world networks; distributed multimedia model; ontology; community; fuzzy similarity

I. INTRODUCTION

Over the last decades, a media availability explosion has emerged with respect to multimedia content. Consumers can simultaneously access widely available digital processing electronic devices. On the other hand, the resulting large amount of generated multimedia content and content descriptive metadata all over the web has greatly increased the required content storage. This growth has encouraged researchers to utilize distributed models for content search and content management. In distributed systems, fault tolerance is advantageous, because failure of one component does not result in a complete system failure. In such systems, if a component fails, in most cases only part of the system will be disabled, and most of it will continue to be active. In fact, this failure will only reduce system performance. As a consequence, a distributed approach is preferable, due to increase of actual data storage and to avoidance of complete loss of system function in the presence of system component failures.

To design a distributed content search framework, this study focuses on some factors which play an important role in order to achieve improved performances. The first one is decentralization, in both system function and data storage. Secondly, an important factor is to achieve admissible performance reduction in cases that the system remains at least partially functional. Another essential factor is the availability of a flexible indexing technique that can be applied to different indexing scenarios. Moreover, an effective technique is needed to insert new elements into the index of the database, as the latter continuously grows. Finally, to retrieve data, a search

method with good generalization ability which can be applied to all types of content such as images, audio, video was used.

So far, a number of distributed frameworks have been proposed for multimedia storage, but all of them use a centralized knowledge part, which is used to determine the location of multimedia data when a query is run or to find a location when the data needs to be stored. This results in a reduction in system efficiency as far as search, insertion and updating time are concerned. Moreover, to retrieve content, most models must send a request to a list of servers that have been selected based on specific criteria and then choose the best answer. So when there is a central infrastructure environment for content retrieval, a large number of messages is needed. Even in cases when the models use summarized requests, it is still needed sending the message to a lot of servers. There are some models that use small world networks to decrease number of messages but due to lack of attention to the community of the small world, they have not significantly reduced this number. Since, increasing number of messages in a distributed system would provide a reduced performance, *decreasing the number of these messages* is very effective in improving the situation.

One of the important parameters in a distributed environment is system response time. Clearly, smaller times are targeted, since they are of great importance in search and retrieval. As search or retrieval time gets smaller, the system performance becomes better. So *a more appropriate framework requests a small retrieval time*.

Lately, some researchers have proposed to use small-world networks for distributed multimedia information. They have imitated social networks. As humans keep track of descriptions of their friends and acquaintances, every media object can store descriptions of the objects that have maximum similarity with itself. Community plays an important role in these small world networks. Whenever, the members of community are many, the query execution time will decrease.

In this paper, appropriate enhancements in schemes which target community creation has been proposed. To achieve efficiency, the present study was focused on designing distributed multimedia frameworks which reduce the complexity of information sharing and time searching on a large-scale network. The proposed system relies on ontologies to describe the structure and semantics of multimedia properties. By gathering nodes with similar fuzzy ontological interests together, search gets more focused and efficient.

The proposed distributed multimedia framework focuses on decreasing centralization in small-world networks and consequently on increasing efficiency. In this context, each society operates as an independent server, permitting insertion or retrieval requests be sent to each community and *increasing decentralization*. In this framework, the members of each community are examined locally. Then, if nothing is found, communities related to the current one are examined, thus *reducing average response time*, as well as *average number of messages*.

In the following, Section 2 makes reference to existing related work. Section 3 presents the components which form the basis of the proposed system. Section 4 describes the proposed system architecture. Section 5 presents an evaluation study in which the proposed system provides improved performances, while section 6 gives conclusions and suggestions for further research.

II. RELATED WORKS

There are a lot of methods which have been proposed for creating distributed frameworks and platforms for multimedia systems [1, 2]. In [3] crisp clustering is addressed, using a clustering model with control constraints that is formulated as a linear optimization problem with Boolean variables. This model allows the control of the quality of clustering through adjustment of the related parameters. Amoretti[4] has provided an ontology-based Grid Service for searching multimedia content. In this work, too many local search services are connected to a global search service. Brut [5] has designed a framework that uses a central server and many remote ones. In this approach, a query is executed only on a set of relevant servers based on user queries, using semantic processing and available knowledge about the distributed servers, the multimedia content and the indexing algorithms. In [6] two components are used to perform distributed query processing: a multimedia application interface, that is a global query processing interface, and a distributed query content-retrieval engine. Laborie [7] has designed a framework for distributed multimedia that also uses a central server and many remote servers. It transfers only a concise version of the distributed metadata to the central server. When a set of servers has been selected by the central server as one that includes data likely to match a particular user query, the query can be processed locally on these servers. In [8] the application of Distributed Information Retrieval (DIR) is described, based on three main phases: resource description, resource selection and merging of results. Resource selection is executed using a centralized sample index of documents and a centralized ranking of sources based on the number and the position of their documents. Then, the user's query is forwarded to the selected sources and the retrieved source-specific results are merged into a single list using score normalization methods. In this framework, gathering information about the sources of a distributed system is the cause for the resulting computational overload.

In all these approaches, a centralized knowledge repository is used to feed the algorithms and perform content filtering. As was above mentioned, decentralization as well as efficiency are important and desired factors. Subsequently, some researchers

have proposed approaches based on the small world network principle [9, 10] due to which a network has a short distance between nodes and a high decentralization ability. In general, small world networks mimic social networks. In these networks, community plays an important role. Androutsos [11] has applied the small network concept to his framework, but he has not paid the necessary attention to the community. Community structure is a common property that can be seen in many networks. Newman has shown that the community has a high density within edges and a lower density elsewhere (M. Newman and Girvan 2004). Therefore, affecting the identity of the members and their degree of cohesiveness can be considered.

One of the key issues for the management and retrieval of relevant information within a community is efficient content indexing. Indexing can be done according to a set of algorithms, which generate diverse and heterogeneous multimedia metadata, and in which resource consuming is high. To design a distributed multimedia systems, there are some choices that must be determined for indexing (Özsu and Valduriez 2011). Examples include using a fixed or variable set of algorithms, using algorithms executed over the entire multimedia collection, or only over a filtered sub-collection, indexing in a distributed manner - executed on the same location as the content, or in centralized mode by transferring the content to an indexation server, the latter being a decision which defines the distributed, or centralized, placement of multimedia metadata.

In the following,, a distributed framework has been proposed for content management and search using small-world networks. It is based on the community concept, while using "fuzzy similarity" to build cliques of data nodes, thus, locally performing most of the tasks. Multimedia data comprises two parts. The first includes multimedia components like text, graphics, animation, sounds, or video. The second part contains multimedia metadata and semantic annotation of multimedia content which can improve the services based on multimedia content. In this work, similarity is measured with regards to the above-mentioned second part of the data.

III. THE BASIC COMPONENTS

A. Small-world networks

In late 1960, Stanley Milgram performed quantitative studies on the structure of social networks [9]. Milgram found that an average of six steps is needed for a letter to get from Nebraska to Boston. This finding has been further extended and called "six degrees of separation" [10].

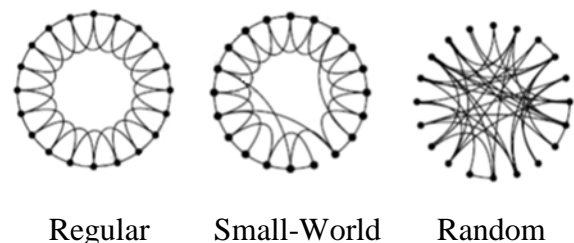


Fig. 1. Three types of networks [12]

In 1998, Watts *et al.* [12] proposed a model for small-worlds ranging between regular network and random distributed network (see Figure 1). It has been proved that some parameters such as size and average distance in a regular network are bigger than in a random network because of their topology [13]. Watts assumed a small probability p for a regular network's edges, and reconnected edges according to the value of p . This was called a rewiring action, with selection of a new node made absolutely at random. In this framework, each edge in a graph can be disconnected from its end point and be connected to another random node. By this action, the graph diameter can be reduced, because the farther nodes are reconnected to shorter ones. In this model, each node in the graph is considered equally likely to be rewired.

Newman and Watts [14] clarified that considering an equal probability for each node in the Watts-Strogatz model has two problems. First, shortcuts do not distribute totally uniformly and all positions are not equally appropriate for rewiring. Secondly, this model poorly defines the average distance between pairs of vertices on the graph because there is a probability of detaching some parts of the graph through the process of rewiring. Consequently, they kept the original links without variation, while adding shortcuts between pairs of vertices that were chosen at random. They also considered more than one link between any vertices.

SWIM [11] is a model resembling Newman and Watts's one, but has some different fundamental assumptions. At first, in contrast to the models in [12] and [14], which exclusively use undirected edges, all connections within the SWIM network are directed, similarly to the structure of the WWW.

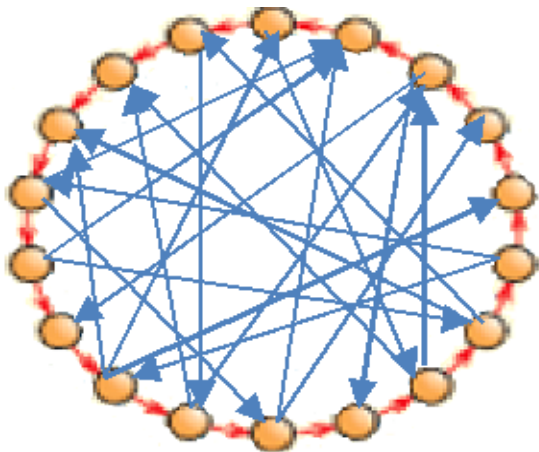


Fig. 2. The SWIM graph made by combining a similar graph and K-lattice

Specifically, in the SWIM system, an underlying directed k -lattice is embedded together with a directed pseudorandomgraph in an N -node network (see Figure 2). It should be noted that the SWIM network does not establish random connections between nodes according to a probability value (as in the Newman-Watts model). Instead, connections are built in accordance to a particular external distance that is computed using data stored locally at each node. There is one descriptor in all nodes that is used to calculate distances. The similarity graph is related to a specific descriptor and distance measure (see similarity section). Therefore, using a different

descriptor for describing the data will lead to a different similarity graph. Similarly, the use of a different distance measure also results in a different graph. It is important to note that the k -lattice (called weak network) in SWIM is not dependent on the distance measure. Instead, it is only dependent on the order in which nodes are introduced into the network.

B. Small world communities

A common property that can be considered crucial for many networks, is the community structure, i.e., the division of network nodes into groups with the following property: network connections are dense within each of these groups, but sparse, between different groups. By analyzing such groups, it is easy to understand and visualize the structure of networks[15]. Small world graphs contain inherent community structure. Community structure is similar to clustering, with clusters being defined through similarity terms; distance is in general proportional to similarity between any two nodes.

A wide variety of similarity measures can be used for community generation. For example, some networks have built in natural similarity metrics. On the opposite, other networks have no natural metric, but suitable ones can be devised using correlation coefficients, path lengths, or matrix methods. In this paper, a number of communities are constructed according to fuzzy ontology-based similarity, measured using both ontological and fuzzy metrics.

C. Ontologies

From a structural point of view, an ontology is composed of disjoint sets of concepts, relations, attributes and data types [16]. Concepts are sets of real world entities with common features. Relations are binary associations between concepts. There exist inter-concept relations, which are common to any domain and domain-dependent associations. Attributes represent quantitative and qualitative features of particular concepts, which take values in a given scale defined by the data type.

An ontology is usually shown by graph $G=(V, E)$ where the nodes of the graph represent concepts. Each edge expresses a kind of semantic relation between two nodes that represent these two concepts. V is a set of nodes and E is a set of edges. E includes two sub-sets, which are a set of hierarchical components and a set of non-hierarchical components respectively [17].

Ontologies have been developed for many purposes [18] targeting reusability and information sharing. To enable content to be discovered and exploited by services, agents and applications, the latter needs to be semantically described. Ontologies provide a formal description of the abstractions that represent content. Generation of ontological representations for a specific semantic area is of high importance and is continuously expanding in most content analysis cases today. Using ontologies is very useful for semantic processing of multimedia data. Multimedia metadata and semantic annotation of multimedia content are very important for retrieval and management of multimedia content.

Several metadata models and metadata standards have been proposed in which the scope and level of detail is different.

Saathoff (Simou, Saathoff, and Dasiopoulou 2006) has suggested the Multimedia Metadata Ontology (M3O) that is a framework in which both semantic and low-level metadata are integrated. This can be used for a variety of tasks, such as (Sjekavica, Obradović, and Gledec 2013) tagging or labeling of multimedia content, ontology-driven semantic analysis and retrieval of multimedia content, recommendation and filtering of multimedia content based on user preferences, personalization and retrieval.

D. Similarity

Similarity plays a fundamental role in the theories of knowledge and behavior. It serves as an organizing principle by which individuals classify object concepts and make generalizations. Similar, or dissimilar data appear in different forms: rating of pairs, sorting of objects, communality between associations, errors of substitution, and correlation between occurrences. Analysis of such data attempts to explain the observed similarity relations and to capture the underlying structure of the objects under study [19].

The assessment of similarity (from a domain independent point of view) has many direct applications, such as word-sense disambiguation [20], information retrieval [21], ontology learning, and biomedical applications [22].

E. Hierarchy and Semantic Similarity

To compute similarity in hierarchy ontologies, it is necessary to determine the way in which ancestral concepts, that are relative closely to a concept, can be identified. Moreover, it is needed how shared information of a pair concept within ontologies is specified. For example, Wu and Palmer [23] proposed a method that computes the number of is-a links (N_1 and N_2) from each term to their Least Common Subsumer (LCS) (*i.e.*, the most concrete taxonomical ancestor who subsumes both terms) and also the number of is-a links from the LCS to the root (N_3) of the ontology.

$$Sim(a, b) = \frac{2 * N_3}{N_1 + N_2 + 2 * N_3} \quad (1)$$

In (1) a and b represent two hierarchy ontologies.

Semantic similarity assessment is generally based on the estimation of semantic evidence observed in one or several knowledge or information sources. On the other hand, semantic similarity is non-hierarchical. In [17] the non-hierarchical components of an ontology are represented by an additional adjacency matrix. Different semantic relations are linked to each other based on their semantic similarity. Assuming there are k different semantic relations R_1, R_2, \dots, R_k . Then, let $\beta_1, \beta_2, \dots, \beta_k$ represent the corresponding similarity factors (weights). A non-hierarchical semantic relation adjacency matrix S is defined as follows:

$$S_{ij} = \begin{cases} 1 & \text{if } i = j \\ \beta_i & \text{if } i \neq j \text{ and } (i, j) \in R_i \\ 0 & \text{otherwise} \end{cases} \quad (2)$$

The semantic similarity measure may be generalized to a fuzzy semantic similarity measure, if the weights of the relation link are replaced by membership degrees indicating the strength of the relationships between parent and child concepts.

F. Ontology-based Fuzzy Similarity with respect to Imprecise Knowledge

Some models use imprecise knowledge to compute similarity [24]. Due to the fact that imprecise knowledge is acquired by experts/users, this knowledge should be rated, or the best part of it is chosen. In a previous work [25], similarity has been calculated by also giving credit to the experts/users. For example, in the area of European Fashion digitization and service provision, different Fashion stakeholders contribute in related knowledge generation, including experts working for fashion content providers, who annotate images of fashion objects or fashion events, or users who generate new fashion content and images, e.g. in social media. In fact, if the knowledge acquired by experts/users is going to be used in the semantic similarity approach, one should consider validation of experts to be a very important factor. One may give degrees of confidence to the actual expert skills, or to the actual confidence of the expert with respect to the specific topic under consideration. In this framework, using the validity factor, taking into account fuzzy properties of the actual validation process has been proposed. In all cases, when imprecise knowledge is present, similarity is computed based on exploitation of fuzzy properties, since imprecision is of rather fuzzy nature. In ontology-based case-based reasoning (CBR), one typically utilizes two sets of case attributes: one set represents the database (DC), whereas the other one forms the query (QC). To compute the similarity, all members of these sets are considered. At the same time, the effect of validation in all imprecision situations was depicted, as was explained before. The validation degree can be considered as a weight for all imprecise values, because it depicts the amount of the acceptability. At first, let us assume that d_i and q_j are literals for DC and QC respectively (such as value of property attribute that is expressed in an imprecise way). In order to compute the overall fuzzy semantic similarity, including validation in the process, the following equation is proposed:

$$Fsim(QC, DC) = Agg(sim(q_i, d_j) * \max(\max(W_{vk} * FDC(d_{jk})), 1 - FIK) \quad (3)$$

where:

Agg is an aggregation function;

$sim(q_i, d_j)$ is the similarity between the i_{th} literal value of DC and the j_{th} literal value of QC;

FDC is the fuzzy membership degree of the j_{th} literal value of DC, for the k_{th} expert;

FIK value may be either 0 or 1, as it denotes whether imprecise knowledge is considered in similarity evaluation or not;

W_{vk} is the degree of validation related to k_{th} expert.

In (3), the *max* function of Zadeh [26] has been adopted, because when a DC is used, two parameters are important: the fuzzy membership degree which shows the strength of the association between DC and QC literals, as well as W_v , which expresses the validation of this degree.

In addition and as already mentioned, it is also needed to take care of imprecision knowledge. Thus, when *FIK* equals 1, its impact within the inner *max* function of Eq. (3) is considered in order to compute the overall similarity. On the contrary, when *FIK* equals 0, the imprecision knowledge should not be utilized in the process. Consequently, in this case $1-FIK$ equals 1 and the output of *max* function becomes 1, thus neutralizing its effect. Finally, to combine both situations, a fuzzy union operator, which is depicted by the outer *max* function in Eq. (3) is utilized.

IV. THE PROPOSED ARCHITECTURE FOR CONTENT SEARCH

In the following, an architecture that fulfills the requirements of improved performance of the search and retrieval process and of the indexing framework is described. This architecture has a two-layered structure. The lower layer provides the data resources, as well as specific services related to the storage system and the metadata repository. The upper layer includes high-level components, such as communities and small network links. Small-worlds are constructed by grouping nodes according to their fuzzy ontological similarity. In particular, the small-world layer organizes multimedia nodes in such communities, allowing nodes to efficiently locate areas of ontology similarity. The communities facilitate creation of the ontological small worlds. Accordingly, queries can be executed on the nodes that are likely to contain the relevant resource; as a consequence, a lower network load and a better search performance is achieved.

To make a community, firstly, the community coefficient (CC) is computed, defined as follows:

$$CC(Head, New Node) =$$

$$\begin{cases} 1 & Sim(Head, New node) \geq th \\ 0 & Otherwise \end{cases} \quad (4)$$

The threshold (**th**) is determined according to the amount of similarity between nodes of the network. The amount of threshold plays an important role in framework structure because a small value of it causes creation of a big community. In the opposite, a big value of it results in a small community. After CC coefficient computing, if the CC value is equal to one, the node is inserted to the current community. Otherwise, it is moved to the next community head.

Figure 3 shows the proposed two-layer structure, including the physical layer and the small-world layer.

A. The Physical Layer

The physical layer comprises multimedia, as well as metadata collections. These collections can be organized in many ways. In the large scale distributed indexing of multimedia objects (LINDO) system [5], there are some remote servers that store and index all acquired multimedia information. Some modules are used to manage all needed tasks such as the *Storage Manager*, the *Access Manager*, the *Feature Extractor Manager*, the *Metadata Engine*. Laborie [7] proposed a scheme in which parts of every server store multimedia information. Moreover, a multimedia collection is stored on a server dedicated to acquire remote site information. A set of extractors is applied to a given piece of multimedia

content returning a set of content metadata. The latter contain information about the media characteristics, while a metadata collection contains all content metadata describing the objects of the multimedia collection.

Chatterjee [6] proposed an architecture that includes a set of data nodes. Each data node has a multimedia database management system embedded in it. It also has a GridFTP server that takes care of the physical transfer of multimedia objects from one node to another. The data is basically stored in a data server. The multimedia database framework is divided into four major components: a multimedia interface, a core DBMS engine, a content-retrieval engine and a high-level relationship manager. These four components interact with each another to achieve the major functionalities, including queries, and updates.

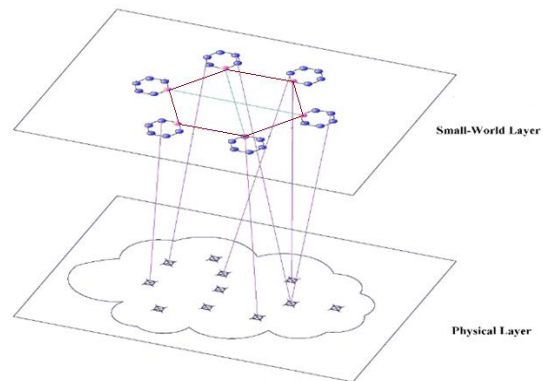


Fig. 3. The two-tier structure

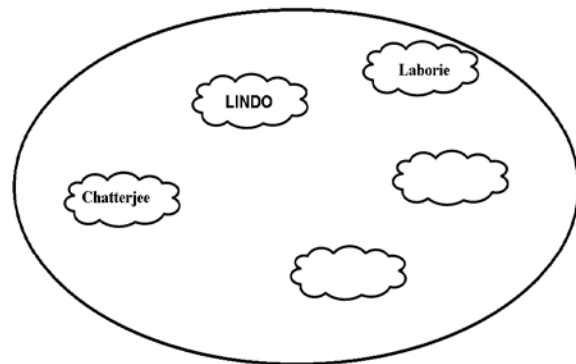


Fig. 4. Physical Layer

In the framework, the physical layer can include different storing policies for each node, as shown in Figure 4.

It is, however, necessary to generate a mapping which will help to meet the requirements of a small-network layer.

B. The Small Network Layer

With the help of this layer, nodes are connected according to their ontological similarity, forming small-world network communities. The topology of each community overlay can be flexible: if the community is large, nodes will be further broken up into small subcommunities to distribute multimedia nodes efficiently into the community, according to their fuzzy ontology similarity. Each community has a node, as

community index (shown, with red circle, in Figure 3), that is used to insert and retrieve nodes.

The layer's distributed nature stems from the fact that it attempts to exploit the small world phenomenon; i.e. all members of a social network retain information about respective small groups of peers and apply it to perform media indexing. As can be seen in Figure 3, the small-world network properties of the layer has been used. The nodes are located in this layer according to the amount of similarity they have with the head of each community. Every community has a head that is used to enter and search content in the community. The properties of small worlds are set by the heads. The latter locally store descriptions of themselves and their peers. The peers can be selected using a distance measure function, based on multiple criteria of similarity. As a consequence, clustering based on this formulation can result in improving system performance.

In addition, this layer attempts to make each community behave as an independent entity in the network in which it participates. Since a small-world network allows only few step searching between nodes (6 according to Milgram), search time is significantly reduced. To understand better the structure, the 'enter' and 'search' methods have been explained in the following.

C. Small World Layer Construction

The first node is considered as head of each community. When adding a new node, it is compared to the head. The following situations may be happen:

Similarity between head and new node is greater than a threshold:

It will be added to the community. Based on the similarity degree of nodes within the community, nodes can be sorted in descending order.

Similarity between head and new node is less than a threshold:

Relevant heads are compared to the new node. The following situations may occur:

There are communities, in which similarity degrees are greater than a threshold. In this case, the new node is added to the community with the highest similarity.

There is no community, in which a similarity degree is greater than a threshold. In this case, a new community, with this node as head, is created. Then, all other communities are sorted based on their degree of similarity. According to the number of peers, the unique number associated with the new community can compute its peers. These steps are shown in detail in Figure 5.

To evaluate the proposed framework, the image database of Lotus Hill Institute (LHI) has been used. The database provides large scale annotated ground truth data including extraction of edges, contours, contour attributes, segmentation, grouping, occluded contour completion, text, and object category recognition, 3D frames, UAV images, Google Earth images, video and cartoons. Part of the ground truth data are packed for various vision tasks and released one-by-one in xml format,

together with respective Matlab code for reading and visualizing the annotation.

V. EVALUATION STUDY

A. Experiment 1

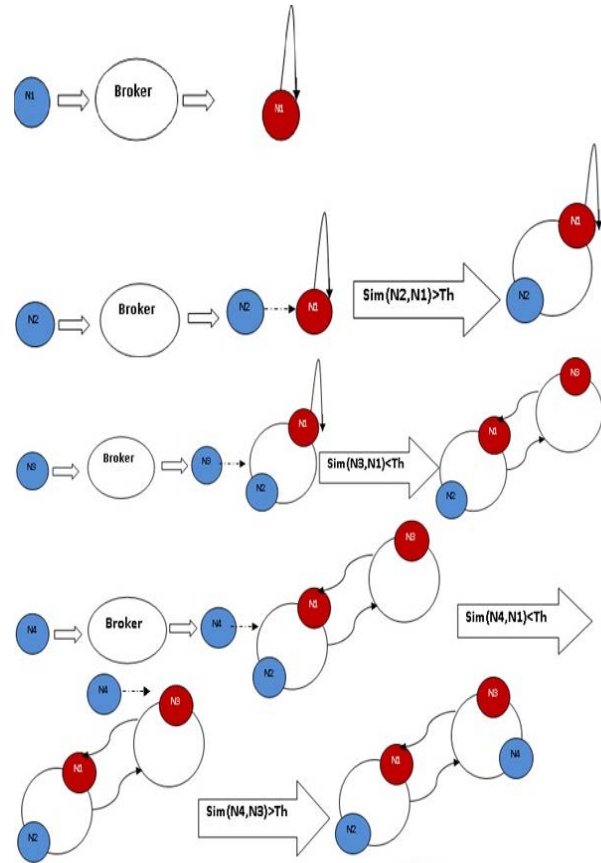


Fig. 5. The Small World Layer Construction

Then, the images of the database is used to test the proposed method. To build the community, the fuzzy XML similarity algorithm has been used as it is presented in [27].

One of the most important factors in evaluating the performance of the distributed system is the number of hops to locate an object; a hop corresponds to a move from one community or cluster to another. Reducing the number of hops is equivalent to improving the system's performance.

As was mentioned before, each node in the framework has some peers that are selected based on specific criteria. First, the number of hops is compared when the peers of maximum, or, of minimum similarity is selected.

First, the effect of an increased number of peers to the required number of hops has been measured. The result is shown in Figures 6 and 7. It can be seen that performance is improved when peers are added.

Finally, the required number of hops have been computed when using a normal clustering framework compared to the proposed framework. It can be seen that the proposed method achieves a better performance; performance gets higher, as the number of images increases (see Figure 8).

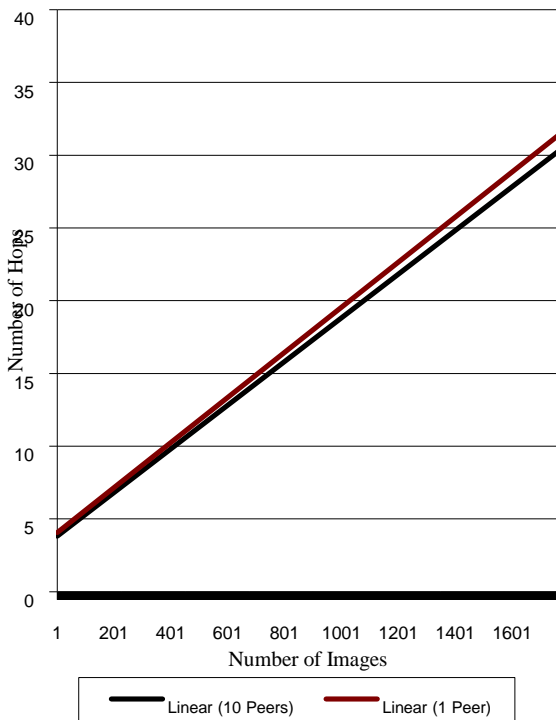


Fig. 6. The effect of increasing number of peers on the required number of hops

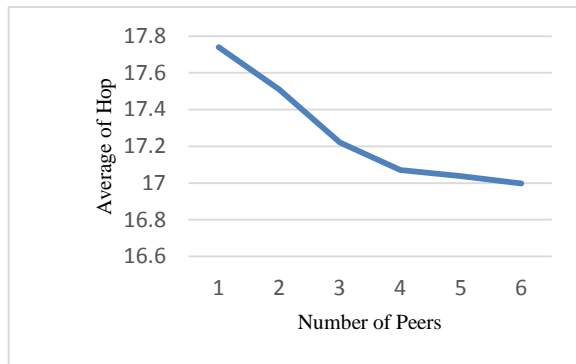


Fig. 7. The average number of hops when increasing the number of peers

B. Experiment 2

The dataset of experiment 2 was a set of 10000 homogeneous XML documents taken from the dblp DTD. The DBLP computer science bibliography contains the metadata of over 1.8 million publications, written by over 1 million authors in several thousands of journals or conference proceedings series. For computer science researchers the DBLP web site is a popular tool to trace the work of colleagues and to retrieve bibliographic details when composing the lists of references for new papers. Ranking and profiling of persons, institutions, journals, or conferences are the other controversial usage of DBLP. The bibliographic records are contained in a huge XML file. Many researchers simply need non-toy files to test and evaluate their algorithms. It is easy to derive several graphs like the bipartite person publication graph, the person-journal or person-conference graphs, or the coauthor graph, which are examples of a social network. Methods for analysis and

visualization of these medium sized graphs are reported in many papers. To evaluate the proposed approach, the protocol develop by [3] was used.

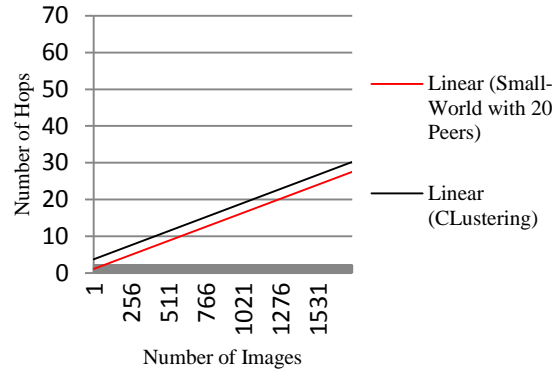


Fig. 8. The required number of hops when using a normal clustering framework and the proposed framework

The DBLP data set is available in <http://dblp.uni-trier.de/xml/>. The file dblp.xml contains all bibliographic records which make DBLP. It is accompanied by the data type definition file dblp.dtd. dblp.xml has a simple layout:

```
<?xml version="1.0" encoding="ISO-8859-1"?>
<!DOCTYPE dblp SYSTEM "dblp.dtd">
<dblp>
record 1
...
record n
</dblp>
```

The header line specifies ISO-8859-1 (Latin-1) as the encoding, but in fact the file only contains characters <128, i.e. pure ASCII. All non-ASCII characters are represented by symbolic or numeric entities. The symbolic entities like ´ for the character 'e' are declared in the DTD. Numeric entities like é should be understood by any XML parser without declaration. In practice there are some obstacles in parsing the large XML file which cost us a lot of time: The SAX parser contained in the Java standard distribution has a limit for handling symbolic entities. When starting the Java virtual machine the option 'EntityExpansionLimit' has to be set to a large number. The Xerces-J from the Apache XML project reads dblp.xml without any problem. The DBLP FAQ3 reports more details. The XML root element <dblp> contains a long sequence of bibliographic records. The DTD lists several elements to be used as a bibliographic record:

```
<!ELEMENT dblp (article|inproceedings|
proceedings|book|incollection|
phdthesis|mastersthesis|jwww)*>
<article key="journals/cacm/Szalay08"
```

```
mdate="2008-11-03">  
<author>Alexander S. Szalay</author>  
<title>Jim Gray, astronomer.</title>  
<pages>58-65</pages>  
<year>2008</year>  
<volume>51</volume>  
<journal>Commun. ACM</journal>  
<number>11</number>  
<ee>http://doi.acm.org/10.1145/  
1400214.1400231</ee>  
<url>db/journals/cacm/  
cacm51.html#Szalay08</url>  
</article>
```

The experiment aims at providing an efficient way to search for xml files in the dblp dataset. The performance evaluation is done on a computer with Intel 3Ghz CPU and 4GB RAM.

Clustering of the files has been achieved using the concept of fuzzy similarity which was above described. In the presented results, shortcuts were built after clustering execution. Three parameters have been considered to evaluate the proposed system: the number of hops (similarly to Experiment 1), the number of shortcuts, and the number of files. In this experiment, at first the data size was changed from 500 to 10,000 xml files. All files were introduced in the clusters and then there was a search for them. Figure 9 shows the evolution of the number of total hops with respect to the data set size at different shortcuts (peers).

It is clear from Figure 9 that an increase of the data set size leads to an increase in the number of hops. This finding is in agreement with previous researches [3, 6, 8]. Moreover, increasing the number of peers from 5 to 10 decreases the respective number of hops when dealing with a small dataset (pink region in Figure 9); if, however, the dataset size increases, this effect is reduced.

On the other hand, the effect when changing the number of peers from 10 to 50 is not significant when the dataset size is small. On the contrary, it gets significant in larger datasets, e.g., when 10000 files are considered (blue region in Figure 9). A regression equation that describes the chart can be computed as follows:

$$Z = 0.0043 + 2.152X - 197.122Y \quad R^2 = .7 \quad (5)$$

Where X, Y, and Z are number of files, number of peers, and sum of hops respectively.

Finally, the required number of hops when using a normal clustering framework is compared to the proposed framework. It can be seen that the proposed method shows a much better performance, which requires less hops as the number of xml files increases (see Figure 10).

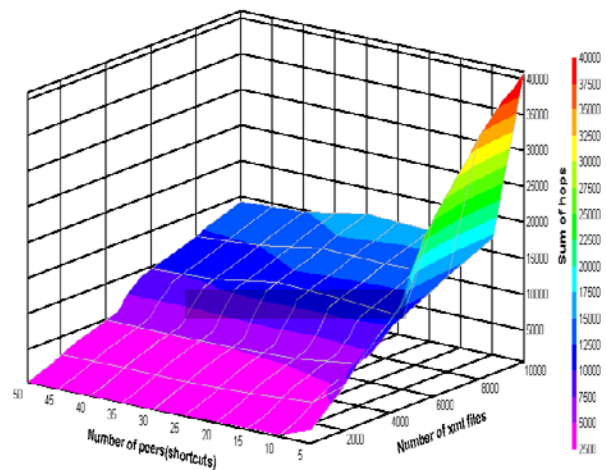


Fig. 9. Dependence of the hops on number of peers and dataset size

VI. CONCLUSIONS

This study has presented a distributed content search framework based on a two-tier architecture, consisting of a physical and a small-world layer. It was shown that this scheme provides efficient content search. In the presented experiments, the performance of the proposed small-world framework has been compared to a clustering system. The results showed that the small-world network requires a much smaller number of hops and subsequently results in lower content search time.

The proposed method can be of great importance for a variety of image and video retrieval and indexing. A great diversity of methodologies can gain from adopting the presented frameworks. In particular, the focus of [28] has been on reducing the data space in the considered clusters. However, in this approach, no relation between the clusters is taken into account, while the advantage of the framework is the small-world network which is built between the clusters. Indexing approaches for image or video files have been presented in [29, 30]. However, in this approach one step goes forward, using an ontology-based fuzzy similarity, based on both semantic and structural issues. The centralization used in these techniques [28-31] for dealing with a distributed framework results in a reduction of the search efficiency. Obviously, gathering information within the distributed system causes a computational overload. This does not occur in the approach, since, in each node of the distributed framework, resource selection can be calculated without gathering information from other nodes – it was built with regard to the shortcut list locally.

Other works (e.g., [32]) adopt (probably semantic based) user modeling techniques to capture users' evolving information needs. Such techniques can be considered as

particular cases of the approach, if it is focused on semantic similarity based on the interests of users.

One of the features of the proposed system is its ability to take into account semantic similarity between resources, possibly expressed in ontological form. There is a great number of publications related to knowledge technologies and multimedia content service provision [33]. Recent advances include creation of the "Billion Triple Challenge" [34] aiming at investigating the scalability of applications as well as the capability to deal with the specifics of data that has been crawled from the public web. Extending the proposed approach in this framework constitutes a topic of future work.

Application of the proposed method in a real cultural heritage environment [35] is currently under development.

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Non Correlation DWT Based Watermarking Behavior in Different Color Spaces

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Abstract—There are two digital watermarking techniques. Digital watermarking techniques based on correlation and digital watermarking techniques that are not based on correlation. In previous work, we proposed a DWT2 based CDMA image watermarking scheme to study the effects of using eight color spaces RGB, YCbCr, JPEG-YCbCr, YIQ, YUV, HSI, HSV and CIELab, on watermarking algorithms based on correlation techniques. This paper proposes a non correlation based image watermarking scheme in wavelet transform domain and tests it in the same color spaces, to develop studying, reach a comprehensive analysis and focus on satisfying the requirements of based non coloration watermarking algorithms. To achieve more security, imperceptibility and robustness of the proposed scheme, first, the binary watermark image encodes by applying ATM, CCM and exclusive OR. Then, the scrambled watermark embeds into intended quantized approximation coefficients of wavelet transform by LSB insertion technique.

Keywords—ATM; CCM; DWT2; color spaces; non correlation watermarking technique

I. INTRODUCTION

In this modern era, computers and the internet are major communication media that bring the different parts of the world into one global virtual world. As a result, people can easily exchange information and distance is no longer a barrier to communication. However, the safety and security of long-distance communication remain an issue. This is particularly important in the case of confidential data. The need to solve this problem has led to the development of watermarking schemes [1, 2]. A digital watermark is an identification code that carries information about the copyright owner, the creator of the work, the authorized consumer, etc. It is embedded in the multimedia data, digital, serial number, text, image, logo and other types of information, and plays a role of copyright protection, signs product, secret communications, confirming data belonging, identifying data authenticity and so on [3].

Typical watermarking schemes embed the watermark by altering coefficients related to the original source in some specific domain, including the spatial domain methods, transform domain techniques using discrete cosine transform (DCT), discrete wavelet transform (DWT) and discrete Fourier transform (DFT), or singular value decomposition (SVD) and vector quantization (VQ) domain schemes [4, 5].

Among frequency domains, the discrete wavelet transform can strengthen the resistance under attacks [6, 7]. Although there are no universal requirements to be satisfied by all

watermarking applications, some main directions are generally considered by the research studies. In order to be effective, the watermark should be perceptually invisible for a human observer (i.e. the imperceptibility property) and its detection should be successful even when the watermarked content is attacked (i.e. the security and robustness property) [8]. There are two digital image watermarking techniques. Digital watermarking techniques based on correlation and digital watermarking techniques that are not based on correlation [9]. In watermarking techniques based on correlation, to extract the watermark, the correlation coefficients between the watermarked content will calculate with a definite threshold and if the correlation is more than the defined threshold, it means that the watermark detector determines watermark image from the content of the image [10, 11]. The simplest example of a watermarking technique that is based on non correlation is the method of least significant bits insertion (LSB). In this method, the least significant bits of many components are watermarked by information [12]. Because the least significant bits do not include visual important information, therefore we can easily replace many of watermarked bits with this level [9, 13].

On the other hand, color spaces abound, but not all of them are appropriate for the entire spectrum of image processing tasks. A color space is a mean of specifying colors, and they can be classified into three basic parts: HVS (human visual system) based color spaces (e.g. RGB, HVS, HSI and etc.), application specific (e.g. YCbCr, YUV, YIQ and etc.) and CIE color spaces (e.g. CIELab and etc.) [14].

The motivation for this study is that there is not a common criterion about which color space is the best one to satisfy image watermarking requirements. There are a lot of studies which describe and use a lot of color spaces to improve each watermarking property, but each study usually focuses on a specific color space, showing the results obtained with that color format. Although, In [14], a DWT2 based CDMA image watermarking scheme proposed and tested in eight color spaces RGB, YCbCr, JPEG-YCbCr, YIQ, YUV, HSI, HSV and CIELab, to explore how the choice of color space, influences the results of correlation based image watermarking algorithms with respect to changes in watermarking anticipating properties such as imperceptibility and robustness against different attacks, but a similar study on based non correlation image watermarking has not done yet. Therefore, in this paper, using wavelet transform, a non correlation based image watermarking scheme proposes and testes in the same color

spaces, to develop the results of previous work in [14] and achieve a comprehensive investigation on the effects of aforementioned color spaces on both based correlation and non correlation watermarking techniques. In the proposed approach, to achieve more security, imperceptibility and robustness of binary watermark image, first, it encodes by applying Arnold's transform map, cross chaos sequence and exclusive OR. Then, the scrambled watermark embeds into intended quantized approximation coefficients of wavelet transform by LSB insertion technique.

II. BASIC THEORIES

A. Two Dimensional Discrete Wavelet Transform

Single level 2D-DWT decomposes an image into 4 different frequency sub-bands LL, LH, HL and HH which are named according to the filter (high-pass or low-pass) applied to the original image in horizontal and vertical directions. For example, HL sub-band is obtained by applying a high pass filter in horizontal direction and low pass filter in vertical direction. The size of new sub-bands is reduced to 1/4 of original size. DWT has many advantages over other transforms due to its ability to represent an image in both spatial and frequency domain simultaneously and to separate the different frequency components of an image. Frequency separation property of DWT is utilized in digital image watermarking to insert watermarks in different frequency sub-bands [15, 16].

B. Cross Chaotic Sequence (CCM)

Chaos is a kind of random-like process which occurred in nonlinear dynamic systems. It is neither periodic nor convergent, but significantly sensitive to its initial conditions. Cross chaotic map is defined as follows [17]:

$$\begin{cases} x_{i+1} = 1 - \mu * y_i^2 ; & x, y \in [-1, 1] \\ y_{i+1} = \text{Cos}(k \text{Cos}^{-1} x_i); \end{cases} \quad (1)$$

where μ and k are control parameters of the system. The system will show better chaotic behavior when $\mu=2$ and $k=6$. Two chaotic sequences $X= x_0, x_1...x_m$ and $Y= y_0, y_1...y_n$, using initial values x_0, y_0 and control parameters, are generated. X and Y are reconstructed as row and column matrix respectively. Then, they are multiplied with each other, to get a new matrix k' . Finally, this matrix is converted to a binary matrix by using the equation (2).

$$f(x) = \begin{cases} 0 ; & 0 < k'(i, j) \leq 0.5 \\ 1 ; & 0.5 < k'(i, j) \leq 1 \end{cases} \quad (2)$$

The advantage of cross chaotic map is that, it has larger key space than 1D logistic chaos sequence, which is used in previous discussed papers. This is due to the six unknown parameters. Because of its larger key space, it gives higher security to image. Also, cross chaotic map resist most of the known attacks such as statistical attack, differential attack and exhaustive attack [17].

C. Arnold's Transform Map (ATM)

The Arnold transform map can shift the positions of pixels instead of changing their values. Recently, it is often used for image encryption and watermarking. It is expressed as [18]:

$$\begin{bmatrix} x' \\ y' \end{bmatrix} = \begin{pmatrix} 1 & 1 \\ 1 & 2 \end{pmatrix} \begin{pmatrix} x \\ y \end{pmatrix} \text{ mod } N \quad (3)$$

where $\begin{pmatrix} x \\ y \end{pmatrix}$ and $\begin{pmatrix} x' \\ y' \end{pmatrix}$ represent the position vector of image pixel shifted before and after, respectively, and mod denotes the modulus after division. The parameter N is the size of the target image, which is used to determine the period of Arnold transform. The number of times Arnold transform is performed is fixed at different values along with different N for enhancing the security of image encryption. The period rules can recover the information of the original image. But we can also find that, when N becomes larger, the period is also larger. That is to say, we should use more transform times to recover the original information. So an inverse Arnold transform is demanded. The inverse Arnold transform can be written as:

$$\begin{bmatrix} x \\ y \end{bmatrix} = \begin{pmatrix} 2 & -1 \\ -1 & 1 \end{pmatrix} \begin{pmatrix} x' \\ y' \end{pmatrix} \text{ mod } N \quad (4)$$

$$A^{-1} = \begin{pmatrix} 2 & -1 \\ -1 & 1 \end{pmatrix} \quad A = \begin{pmatrix} 1 & 1 \\ 1 & 2 \end{pmatrix}$$

where is used instead of

D. Color Spaces

As it was mentioned above, a color space is a mathematical representation of a set of colors. Three fundamental color models are: color spaces based on HVS human visual system (e.g. RGB, HVS, HSI and etc.); application specific (e.g. YCbCr, JPEG-YCbCr, YUV, YIQ and etc.) and CIE color spaces (e.g. CIELab and etc.) [14]. In RGB model, colors are specified in terms of the three primary colors red (R), green (G) and blue (B). It is an additive model because all colors are created from these primary colors combining them in various ways. The amount of each primary color gives its intensity. If all components are of highest intensity, then the color white results. This is the most extended and used color format because it is the one used in displays technology [19]. In YCbCr model, colors are specified in terms of luminance (Y channel) and chrominance (Cb and Cr channels). Cb represents the value for the blue component (B - Y) and Cr saves the information for the red color component (R - Y). YCbCr is widely used for digital video encoding [19]. In [20], we presented a rescaling formula for YCbCr called JPEG-YCbCr which used in the JPEG image format, with Y, Cb and Cr in [0,1]. UUV model defines a color space in terms of one

luminance (Y channel) and two chrominance components (UV channels). U represents the color difference between blue signal and luminance (B – Y) and V represents the difference between red and luminance (R – Y). YUV is used for analog television such as PAL or NTSC. Human vision is much more sensitive to luminance variations than chrominance variations. YUV codification takes advantages of this fact, giving more bandwidth to luminance so that the color space goes closer to the human perception (although YUV is not as close as HSV color space). YUV signals are created from an original RGB (red, green and blue) source. The weighted values of R, G and B are added together to produce a single Y signal, representing the overall brightness, or luminance, of that point. The U signal is then created by subtracting the Y from the blue signal of the original RGB, and then scaling; and V by subtracting the Y from the red, and then scaling by a different factor. Therefore, conversions between RGB and YUV formats can be done through linear transformations [19]. YIQ color space is similar to YUV model. The Y component represents the luminance information, while I (in-phase) and Q (quadrature) represent the chrominance information. In YUV, the U and V components can be thought as X and Y coordinates within the color space. I and Q can be thought as a second pair of axes on the same graph, rotated 33°, therefore IQ and UV represent different coordinate systems on the same plane. ‘I’ is in the range orange–blue, and ‘Q’ in the range purple–green. Therefore, transformations between RGB and YIQ color spaces are linear too [19]. HSI is the most frequently used applications oriented color space. HSI color space is based on the human visual perception theory and is suitable for describing, and interpreting color. H, S and I represent hue, saturation, and intensity respectively. The supposed separation of the luminance component from chrominance information is stated to have advantages in applications such as image processing. Embedding the watermark in the intensity component of HSI color space can resist the filtering, sharpening etc. attacks effectively [21]. HSV color space represents colors in terms of Hue (or color-depth), Saturation (or color-purity) and intensity of the Value (or color-brightness). Hue refers to color type, such as red, blue, or yellow. It takes values from 0 to 360 (but it is normalized to 0–100% in some applications). Saturation refers to the vibrancy or the purity of the color. It takes values from 0 to 100%. The lower the saturation of a color, the more “greyness” the color is represented and the more faded the color will appear. Finally, Value component refers to the brightness of the color. It takes the same range as the saturation (0–100%) [19]. In 1976, the International Commission on Illumination (CIE) recommended the CIE $L^*a^*b^*$, or CIELab color space for color quality estimation. The color space CIELab is a perceptually uniform color space created by nonlinear transformations of tristimulus XYZ values to overcome the non-uniformity of color spaces which had been discussed by Macadam. The main intention was to provide a standard and approximate uniform color space which can be used to compare the color values easily. In this color space the differences between points plotted in the color space correspond to visual differences between the colors plotted. The CIE recommended to use XYZ coordinate system to transform RGB to $L^*a^*b^*$ [14].

III. THE PROPOSED WATERMARKING ALGORITHM

As it was mentioned before, the most watermarking algorithms which are based on non correlation techniques use LSB insertion method. In [22], the author has proposed a LSB-based method, called the inverted pattern (IP) LSB substitution in which each section of secret images is determined to be inverted or not inverted before it is embedded. Also, in embedding process, the calculation of mean square error between cover image and message image is used to embed the secret message bits or the inverted message bits. In [23], a watermarking algorithm has proposed in which the neighboring symbol’s mean value of selected pixels is calculated to embed the watermark bits into LH2 coefficients of wavelet decomposition. In [24], the original watermark is generated by a halftoned version of the host image and inserted into least significant bits of host image.

In this paper, the properties of DWT2, ATM, CCM and also applying exclusive OR are combined in order to embed a highly imperceptible and robust watermark image for secured transmission of data. This scheme is blind and so there is no need to original watermark in the extraction process. The Figs.1 and 2 show the block diagrams of Watermark Embedding and Watermark Extraction, respectively.

A. Watermark Embedding

Watermark embedding is used to insert the watermark into the host image to obtain the watermarked image.

Steps:

- 1) Scramble original and watermark images by applying Arnold transform map, respectively, for N and M times.
- 2) Use Cross Chaos equation on scrambled watermark image.
- 3) Encrypt the watermark sequence by applying the exclusive OR.
- 4) Convert RGB channels of scrambled original image into the intended channels.
- 5) Make three-level wavelet decomposition to the first channel of converted image and use the coarsest subband LL_3 is taken as the target subband for embedding watermarks.
- 6) Save the signs of selection coefficients in a sign matrix.
- 7) Quantize the absolute values of selection coefficients.
- 8) Embed encrypted watermark instead of the second least significant bits of target coefficients that have smallest quantization errors.
- 9) Effect the sign matrix to the embedded coefficients.
- 10) Make a wavelet reconstruction of all changed and unchanged DWT coefficients of the first channel.
- 11) Combine the watermarked channel with the other two channels.
- 12) Reconvert the image form intended color space to RGB color space.
- 13) Re-scramble the reconverted image by applying ATM for N times to generate the watermarked image.
- 14) Save the iterative numbers of Arnold transforms on the original and watermark images (N and M), the initial value of

cross chaotic map, indexes of changed selection coefficients and index of the embedded subband as the authenticated keys.

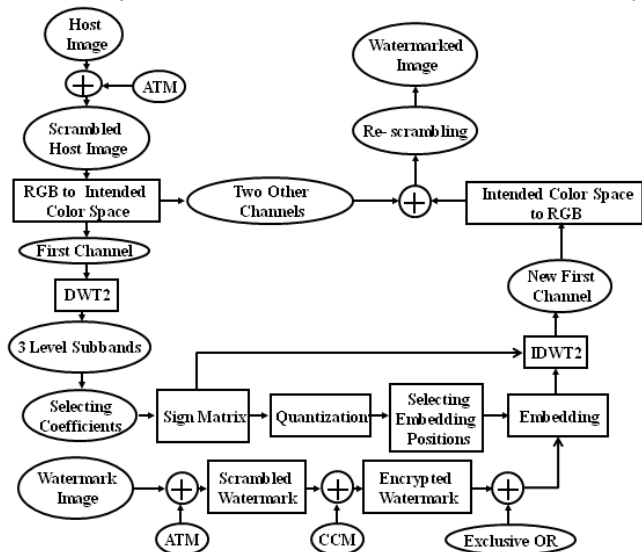


Fig. 1. The watermark embedding block diagram of the proposed approach

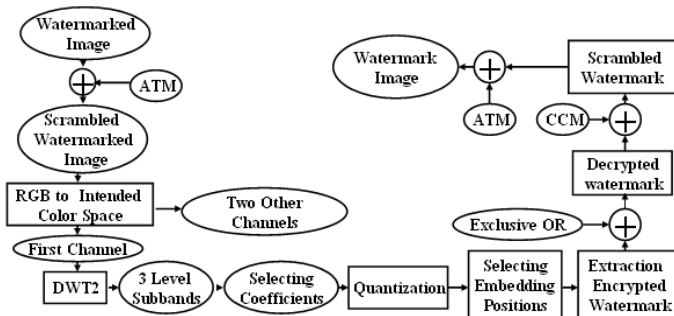


Fig. 2. The watermark extraction block diagram of proposed approach

B. Watermark Extraction

The extracting algorithm of watermarking is the converse process of embedding. As it is a blind scheme, there is no need to the original image for extracting the watermarks. The detailed steps are described in details in the following steps:

- 1) Re-fetch the stored authenticated keys.
- 2) Scramble the watermarked imaged by applying the Arnold's transformation for N times.
- 3) Convert RGB channels of a scrambled watermarked image into the intended color space.
- 4) Decompose the first channel into three levels with ten DWT subbands. The subband LL3 is taken as the target subband for extraction watermarks.
- 5) Quantize the absolute values of LL3 subband.
- 6) Extract the encrypted watermark from the second least significant bits of changed coefficients.

7) Decrypt the extracted watermark image by applying exclusive OR.

8) Take the initial value of cross chaotic map to make the extracted watermark image.

9) Re-generate the watermark image by applying Arnold's transform for M times.

IV. ANALYSIS OF PROPOSED ALGORITHM AND EXPERIMENTAL RESULTS

In order to evaluate the algorithm MATLAB R2007a software was used as a platform and the proposed algorithm is applied to three 256×256 famous images: Lena, Peppers and Baboon shown in Fig. 3(a-c) and a 27×27 binary watermark image shown in Fig. 3(d). Fig. 4 shows the scrambling process of watermark image by ATM method, too. Also, for computation of the wavelet transforms, 7-9 biorthogonal spline (B-spline) wavelet filters were used. Cause of using B-spline function wavelet is that, B-spline functions, do not have compact support, but are orthogonal and have better smoothness properties than other wavelet functions [25].

As we know, there is a direct relationship between processing time and computational complexity. So the processing times (in seconds) of the proposed scheme in all color spaces are calculated to consider the computational complexities of each one of them. The obtained results are shown in Table 1.

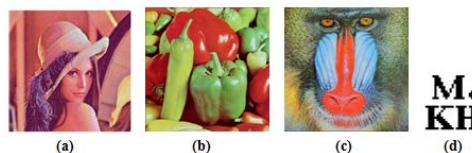


Fig. 3. (a-c) The host Lena, Peppers, and Baboon images and (d) The watermark image

As it is seen, RGB need the minimum times for embedding and extraction processes of an image. Afterwards, respectively YCbCr, YUV, YIQ, JPEG-YCbCr, HSI, HSV and finally CIE Lab need the minimum processing times. It was expectable. Because, in RGB, there is no need to color space conversion; in opposite, CIE Lab has the most complicated formulas to color space conversion. Therefore, we can conclude that in non correlation based watermarking techniques, the maximum computational complexities respectively belong to CIE Lab, HSV, HSI, JPEG-YCbCr, YIQ, YUV, YCbCr and RGB.

After a brief discussion on computational complexities, the similarity of original host images and watermarked images is measured by the standard correlation coefficient (Corr) as [26]:

$$Correlatio = \frac{\sum (x - x')(y - y')}{\sqrt{(x - x')^2} \sqrt{(y - y')^2}} \quad (5)$$

TABLE I. PROCESSING TIMES THE PROPOSED SCHEME IN DIFFERENT COLOR SPACES

Color Space	Lena		Peppers		Baboon	
	Embedding Time (S)	Extracting Time (S)	Embedding Time (S)	Extracting Time (S)	Embedding Time (S)	Extracting Time (S)
RGB	1.1948	0.3432	1.2216	0.3588	1.3444	0.3604
YCbCr	1.4976	0.6690	1.5012	0.5701	1.5468	0.6083
JPEG-YCbCr	1.6224	0.7188	1.6681	0.7276	1.6925	0.7744
YIQ	1.5288	0.5890	1.5444	0.6011	1.5600	0.6185
YUV	1.5176	0.5741	1.5262	0.5844	1.5590	0.6119
HSV	1.7316	0.7976	1.7559	0.8163	1.8222	0.8295
HSI	1.7160	0.7900	1.7401	0.8077	1.8006	0.8238
CIELab	2.1620	1.0900	2.3664	1.5133	2.6088	1.8777

Moreover, the peak signal-to-noise ratio (PSNR) is used to evaluate the quality of the watermarked images as [27]:

$$PSNR = 10 \log_{10} \frac{255^2}{MSE} \quad (dB) \quad (6)$$

where mean-square error (MSE) is defined as [27]:

$$MSE = \frac{1}{mn} \sum_{i=1}^m \sum_{j=1}^n (h_{i,j} - h'_{i,j})^2 \quad (7)$$

where $\{h_{i,j}\}$ and $\{h'_{i,j}\}$ are the gray levels of pixels in the host and watermarked images, respectively.

Also the normalized correlation (NC) coefficient is used to measure the similarity between original watermarks W and the extracted watermark W'. It was defined as [27, 28]:

$$NC = \frac{\sum_i \sum_j w_{i,j} * w'_{i,j}}{\sum_i \sum_j w_{i,j}^2} \quad (8)$$

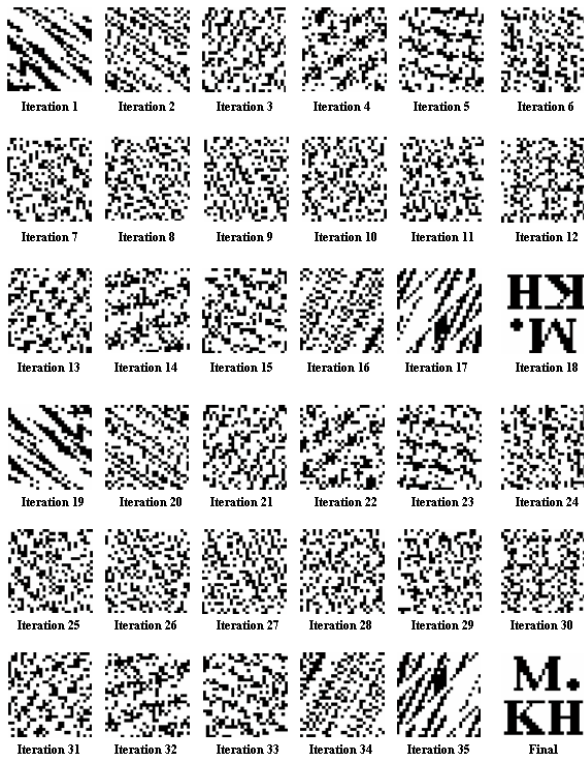


Fig. 4. The scrambling process of watermark image by ATM method

A. Imperceptibility and Transparency Experiments

According to literature, if the PSNR be greater, the image quality is better. In general, a watermarked image is acceptable by human perception if its PSNR is greater than 30 dBs. In other words, the correlation is used for evaluating the robustness of watermarking technique and the PSNR is used for evaluating the transparency of watermarking technique [26, 28]. Fig. 5 shows the watermarked images in all color spaces and table 2, shows the obtained results of PSNR, correlation and normalized correlation values.

TABLE II. THE PSNRs, CORRELATION (CORR) AND NORMALIZED CORRELATION (NC) VALUES OF WATERMARKED IMAGES IN DIFFERENT COLOR SPACES

Color Space	PSNR (dB)		
	Watermarked Images		
	Lena	Peppers	Baboon
RGB	54.95	54.73	54.43
YCbCr	54.65	54.02	54.05
JPEG-YCbCr	55.68	55.94	55.47
YIQ	55.36	55.80	54.71
YUV	55.36	55.80	54.71
HSV	55.44	54.99	55.36
HSI	55.43	54.98	55.35
CIELab	54.74	54.67	54.31

Color Space	Corr		
	Watermarked Images		
	Lena	Peppers	Baboon
RGB	0.9999	0.9999	0.9999
YCbCr	0.9997	0.9998	0.9996
JPEG-YCbCr	0.9998	0.9999	0.9997
YIQ	0.9998	0.9999	0.9997
YUV	0.9998	0.9999	0.9997
HSV	1.0000	1.0000	1.0000
HSI	1.0000	1.0000	1.0000
CIELab	0.9996	0.9992	0.9990

Color Space	NC		
	Watermarked Images		
	Lena	Peppers	Baboon
RGB	1.0000	1.0000	1.0000
YCbCr	1.0000	1.0000	1.0000
JPEG-YCbCr	1.0000	1.0000	1.0000
YIQ	1.0000	1.0000	1.0000
YUV	1.0000	1.0000	1.0000
HSV	1.0000	1.0000	1.0000
HSI	1.0000	1.0000	1.0000
CIELab	1.0000	1.0000	1.0000

The table results show, in all color spaces, the proposed scheme yields satisfactory watermark imperceptibility and transparency. All of the PSNRs are greater than 54 dB and correlation values are all greater than 0.999. In addition, normalized correlations are all equal to 1. It means the embedded watermark images are extracted completely, without

any error. It should be mentioned that the proposed scheme increases security in comparing with the similar algorithms because of three times encoding (i.e. using ATM, CCM and exclusive OR) on watermark image. In addition, encoding host images by ATM method and embedding watermarks in the first channel of each color space are the other criteria for increasing security.

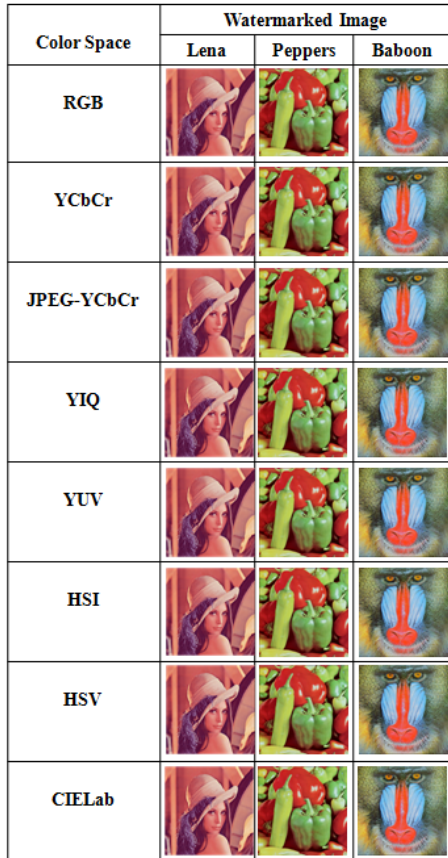


Fig. 5. The watermarked images in different color spaces

By comparing different color spaces, it could be found that in based non correlation watermarking algorithms not only our proposed formula called JPEG-YCbCr color space in [20] improves correlation and PSNR values of YCbCr color space, rather like previous work for based correlation watermarking algorithms in [14], it satisfies imperceptibility more than the other color spaces. After that, respectively, HSV, HSI, YUV & YIQ, RGB, CIE Lab and YCbCr color spaces have the most ability to satisfy imperceptibility requirement. Also, PSNR and correlation values in HSI and HSV color spaces are so near together, while in YIQ and YUV they are exactly the same and can satisfy imperceptibility and transparency properties like each other.

B. Robustness Experiments

Most standard watermarking techniques do not survive wavelet based compression and may also not be compatible with the scalability feature of wavelet based compression. Therefore, to assess the robustness of the proposed algorithm to compression attack, wavelet based compression was chosen. So that, watermarked images were compressed by wavelet

compression method in different thresholds (Thr): 1, 3, 6, 9, 12 and 15, and then the watermark image was extracted from the compressed watermarked images. The extracted watermarks and their percentage of error bit rates (EBR %) of all color spaces are shown in Figs. 6 and 7, respectively. They indicate the proposed scheme can successfully withstand wavelet based compression, as well as standard watermark attacks. The maximum percentage of error bit rate for extracted watermarks in all color spaces under high compression thresholds 12 is lower than 25.86% in baboon (for YCbCr), which demonstrates the appropriate quality of the images under wavelet compression attacks.

Color Space		RGB	YCbCr	JPEG-YCbCr	YIQ	YUV	HSI	HSV	CIE Lab
Threshold									
1.0	Lena	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Peppers	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Baboon	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
3.0	Lena	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Peppers	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Baboon	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
6.0	Lena	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Peppers	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Baboon	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
9.0	Lena	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Peppers	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Baboon	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
12.0	Lena	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Peppers	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Baboon	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
15.0	Lena	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Peppers	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH
	Baboon	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH	M. KH

Fig. 6. The extracted watermarks from compression experiment in different color spaces

By comparing all color spaces, it can be found that YUV & YIQ and also HSI & HSV color spaces have the same robustness against wavelet compression. The lowest error bit rate belongs to HSI & HSV. So, these color spaces are more robust than the other color spaces against compression attacks. Then, respectively, our proposed formula in [20] called JPEG-YCbCr, RGB, YUV & YIQ, RGB, CIE Lab and finally YCbCr color spaces yield most robustness against wavelet compression. The interesting point is our proposed formula called JPEG-YCbCr in [20] not only increases YCbCr robustness, rather it yields better results in comparing RGB, YUV & YIQ, RGB and CIE Lab color spaces.


















































































After achieving satisfactory results in compression experiment, the robustness of the proposed algorithm against cropping attacks was tested. So that, three different areas of

watermarked images were being under cropping attacks and then the extraction process of watermark on the cropped areas was performed. Table 3 shows the extracted watermarks from cropped watermarked images and their percentage of error bit rates (EBR %). The table results show, the maximum percentage of error bit rate in all color spaces is lower than 47.32% (in YCbCr) and the watermarks are still recognizable after cropping attacks as well.

By comparing different color spaces, it can be found that YUV & YIQ and HSI & HSV color spaces have the same

robustness against cropping attacks. The lowest percentage of error bit rate belongs to CIE Lab. So, this color space yields more robustness than the others. After that, JPEG-YCbCr, RGB, HSI & HSV, YIQ & YUV and finally YCbCr color spaces lead to most robustness against cropping attacks. It is necessary to note that, like compression experiment, our presented formula in [20] called JPEG-YCbCr not only enhances the YCbCr robustness, rather it leads to better results in comparing HSI & HSV, YIQ & YUV and RGB color spaces.

TABLE III. THE EXTRACTED WATERMARKS FROM CROPPED WATERMARKED IMAGES AND THEIR PERCENTAGE OF ERROR BIT RATES

Image	Cropped Area 1			Cropped Area 2			Cropped Area 3		
	<i>Lena</i>	<i>Peppers</i>	<i>Baboon</i>	<i>Lena</i>	<i>Peppers</i>	<i>Baboon</i>	<i>Lena</i>	<i>Peppers</i>	<i>Baboon</i>
Cropped Image									
RGB									
Extracted Watermark									
EBR %	44.91	44.54	44.36	45.09	44.52	44.67	43.67	43.54	42.22
YCbCr									
Extracted Watermark									
EBR %	47.32	45.54	45.96	46.73	45.74	45.74	44.85	45.54	46.39
JPEG-YCbCr									
Extracted Watermark									
EBR %	44.80	43.27	43.71	44.39	43.71	43.18	42.62	42.44	41.07
YIQ									
Extracted Watermark									
EBR %	47.18	45.13	45.81	46.63	45.54	45.67	44.71	45.13	45.26
YUV									
Extracted Watermark									
EBR %	47.18	45.13	45.81	46.63	45.54	45.67	44.71	45.13	45.26
HSV									
Extracted Watermark									
EBR %	45.67	45.09	45.09	45.95	44.81	45.26	44.12	43.89	43.07
HSI									
Extracted Watermark									
EBR %	45.67	45.09	45.09	45.95	44.81	45.26	44.12	43.89	43.07
CIE Lab									
Extracted Watermark									
EBR %	44.40	42.63	43.30	44.22	43.54	42.22	41.71	41.45	40.75

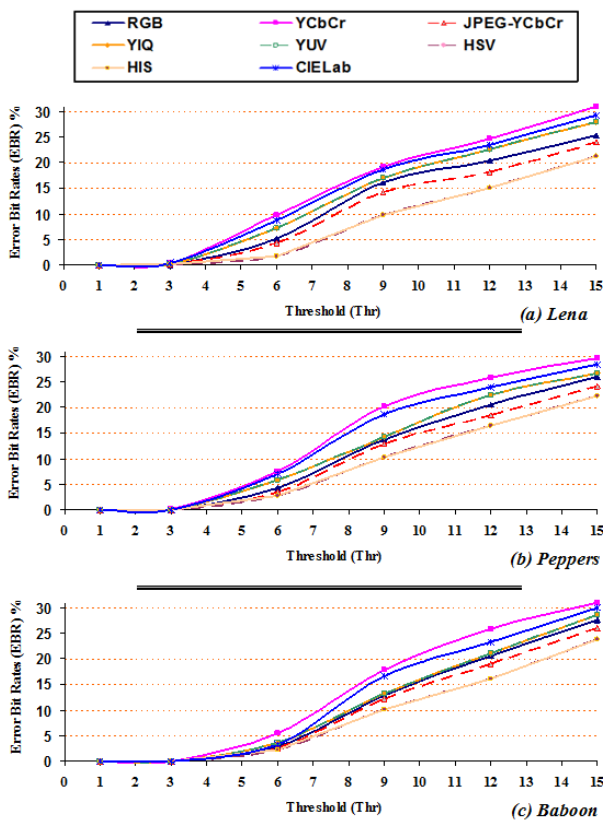


Fig. 7. The percentage of error bit rates (EBR %) of the extracted watermarks from compression experiment in different color spaces

V. CONCLUSION

In this paper, a non correlation based image watermarking scheme using DWT2 proposed and tested in eight historical color spaces: RGB, YCbCr, YIQ, YUV, HSI, HSV, CIELab and improved YCbCr color space, called JPEG- YCbCr which is presented by our formula in [20], to develop the results of previous work in [14] and achieve a comprehensive investigation on the effects of different color spaces on both based correlation and non correlation watermarking techniques. The watermarking security is more satisfactory than the similar works. Because, the watermark W becomes encoded three times by ATM, CCM and exclusive OR and embedded into wavelet decomposition coefficients of a scrambled original image. By the way, scrambling the host image by ATM and embedding encoded watermarks in the first channel of intended color space increase security, more. The experimental results show that in all color spaces, the proposed approach has achieved satisfactory imperceptibility and excellent robustness from three tested benchmark images, simultaneously. In different color spaces, All of the correlation values are greater than 0.999, the lowest PSNR is greater than 54 dB (in YCbCr) and all of the normalized correlations are equal to 1. Also, the maximum percentage of error bit rates for extracted watermarks from huge cropped watermarked images and compressed watermarked images in compression threshold 12 are respectively equal to 47.32% and 25.86% (in YCbCr); which it means the extracted watermarks from cropping and compression experiments are recognizable as well, even if cropped area is big or compression threshold is high.

The proposed approach has the following results in non correlation based watermarking techniques:

1) Our presented formula in [20] called JPEG-YCbCr color space, not only enhances YCbCr imperceptibility and robustness, rather it leads to most results in comparing the other color spaces. In addition, it yields more robustness against compression attacks in comparing RGB, YUV & YIQ, RGB, CIELab color spaces and cropping attacks in comparing RGB, HSI & HSV, YIQ & YUV color spaces.

2) YIQ & YUV color spaces exactly satisfy imperceptibility and robustness against compression and cropping attacks, equally. Also, imperceptibility results in HSI & HSV color spaces are very near together; while they exactly lead to same robustness against wavelet compression and image cropping.

3) After JPEG-YCbCr, respectively, HSV, HSI, YUV & YIQ, RGB, CIELab and YCbCr color spaces yield most imperceptibility.

4) HSI & HSV satisfy robustness against compression more than the other color spaces. Then, respectively, JPEG-YCbCr, RGB, YUV & YIQ, RGB, CIELab and finally YCbCr lead to most robustness.

5) CIELab yields most robustness against cropping attacks in comparing the other color spaces. After that, respectively, JPEG-YCbCr, RGB, HSI & HSV, YIQ & YUV and YCbCr color spaces have the most robustness.

6) In non correlation based watermarking, the minimum computational complexities and processing time in embedding and extraction processes belong to RGB. Afterwards, respectively, they belong to YCbCr, YUV, YIQ, JPEG-YCbCr, HSI, HSV and CIELab spaces.

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Semi-Automatic Segmentation System for Syllables Extraction from Continuous Arabic Audio Signal

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Abstract—The paper describes a speaker independent segmentation system for breaking Arabic uttered sentences into its constituent syllables. The goal is to construct a database of acoustical Arabic syllables as a step towards a syllable-based Arabic speech verification/recognition system. The proposed technique segments the utterances based on maxima extraction from delta function of 1st MFC coefficient. This method locates syllables boundaries by applying the template matching technique with reference utterances. The system was applied over a data set of 276 utterances to segment them into their 2544 constituent syllables. A segmentation success rate of about 91.5% was reached.

Keywords—Arabic speech syllables; automatic segmentation; boundaries detection; delta-MFCC features

I. INTRODUCTION

Speech and natural language processing (SNLP) is a vital topic in recent research. Computer Aided Language Learning (CALL) systems have received considerable attention in recent years. CALL system are used to improve learning and to evaluate pronunciation quality of speakers. Arabic is the spoken language in 60 countries around the world, so it is the second most spoken language in terms of the number of speakers [1]. Quran is the basic reference of Arabic language. One of the most important issues in the Arabic world is the learning of Quran recitations [2].

A robust language learning system should have a vocabulary database in order to recognize uttered speech, localize and identify pronunciation mistakes and provide meaningful feedback to help users to improve their performance. Building an acoustical Arabic database of syllables, phonemes, etc., according with the requirements of the used application is the base for these researches. In this paper, a new method for the automatic segmentation of Arabic audio signal into its syllables is introduced. Classical Arabic is the old form of the Arabic language used in the Quran. Modern Standard Arabic “MSA” is a formal language commonly used in all Arabic-speaking countries.

Our system seeks to perform accurate allocation of syllables boundaries from continuous speech as a step towards building an Arabic database that contributes in developing many applications, such as:

- a) *Diagnosis and treatment of speaking pathology.*
- b) *Teaching the recitation rules of the Holy Quran.*

c) *Training systems for correct Arabic pronunciation for children and non-native speakers.*

d) *Facilitate the man-machine communication and help its progress.*

This paper is organized as follows; section 2 presents segmentation. Section 3 defines the Arabic syllable segmentation. Section 4 introduces the implementation of the proposed system. Section 5 discusses the segmentation experimental results. In section 6, conclusion and future work are presented.

II. RELATED WORK

A. Selecting a Template

Due to the importance of the subject, intensive studies have been conducted on speech segmentation employing different features. A theoretical framework for MSA speech segmentation using dynamic level building was introduced by El Arif et al. [3]. Gody presented a speech segmentation approach depending on wavelet transform and spectral analysis; the accuracy was 95% [4]. Tolba used wavelet transform achieving 81% accuracy [5]. Yacine Yekache et al. reported a step toward developing Quranic reader using sphinx4 framework [6]. Wang et al. [7], Fu et al. [8] introduced zero crossing rate “ZCR”, pitch and energy profile as features for the segmentation of speech. In [9], a survey on Punjabi speech segmentation into syllables is presented using negative derivative of Fourier transformations. In [10], a syllable based recognition system based on pseudo articulatory method is presented which contributes of more plausible style of speech recognition and new modeling of speech behavior. In [11], a group delay based approach is proposed which the short-term energy is processed for determining segment boundaries. An attempt is made by Sarada et al. [12] to automate the syllable transcription task for Indian languages. The method does not require any manual segmentation and a new feature extraction strategy is explored using multiple frame sizes and rates for both training and testing datasets. A technique based on short term energy was implemented in [13] for the automatic segmentation of speech signals in Punjabi speech into syllables. In [14], biologically inspired auditory attention cues are proposed for syllables segmentation from continuous speech. The method achieved 92.1 % accuracy of syllable boundary detection at frame level when tested on TIMIT. In [15], a time-frequency representation and the fusion of intensity and voicing

measures were introduced for the segmentation of speech into syllables. A practical method for blind segmentation of continuous speech is presented by Villing et al. [16] using amplitude onset velocity and coarse spectral makeup to identify syllable boundaries. Mijanur Rahman et al. [17] developed a system that automatically segments words from the continuously spoken Bangla sentences. Mijanur Rahman et al. [17] developed a system that automatically segments words from the continuously spoken Bangla sentences. Our prior works of [18], [19] presented an algorithm for segmenting a subset of emphatic and non-emphatic sounds automatically from continuous spoken Arabic, where achieved a segmentation accuracy of up to 90 %.

III. THE PROPOSED SEGMENTATION SYSTEM

A. Arabic syllables

Speech units can be phonemes, letters, syllables, words, etc. The segmentation problem may be viewed as an unlabeled splitting problem where the input sequence needs to be split into subsequences. This study focuses on the isolation of syllables; the syllable consists of nuclear vowel plus neighboring consonants. The vowel should be preceded by a consonant and followed by zero, one or two consonants. Thus Arabic language has five standard types of syllables: {CV, CV%, CVC, CV%C, CVCC} where “C”: consonant, “V”: vowel, “V%”: long vowel.

B. System block diagram

The block diagram of the proposed system is shown in Fig. 1.

It consists of five modules developed to allocate the syllables boundaries in Arabic utterance

- 1) Data collection, preprocessing, feature extraction and forming reference template
- 2) Test data entry and preprocessing.
- 3) Features Extraction
- 4) Automatic allocation of syllables boundaries through matching process.
- 5) Evaluation of the resulting isolated syllables.

Our system is based on the Holy Quran. The recordings of twelve speakers; each recited 23 continuous sentences (verses), constitute a dataset of 276 utterances. The texts of the collected data with its IPA mapping for according to [20] are reported in table 4 which includes (2544) syllables to be detected and segmented. The recordings from the reader “Mahmoud Khaleel El-Hosary” has been selected to form the template dataset which will be used as reference throughout the matching process, this choice is based on the well-known of his good realization for the rules of recitations of Quran verses.

Wide variability of speech may affect the accuracy of its analysis. So, good setup of pre-processing phase improves the performance of speech segmentation. The audio signal is divided into fixed length frames with overlapping to insure continuity [21]. MIR© software toolbox [22] is used for applying the pre-processing steps, which are: trim silence at beginning and end of the audio signal, normalize the recorded

data, form fixed length frames of 30 ms with 60% overlapping and smooth frame boundaries using Hamming window.

C. Extraction of features from template utterance

The selected tool for allocating syllables boundaries is the feature vector of local maxima picked from Delta function of the first Mel Frequency Cepstrum coefficient [23].

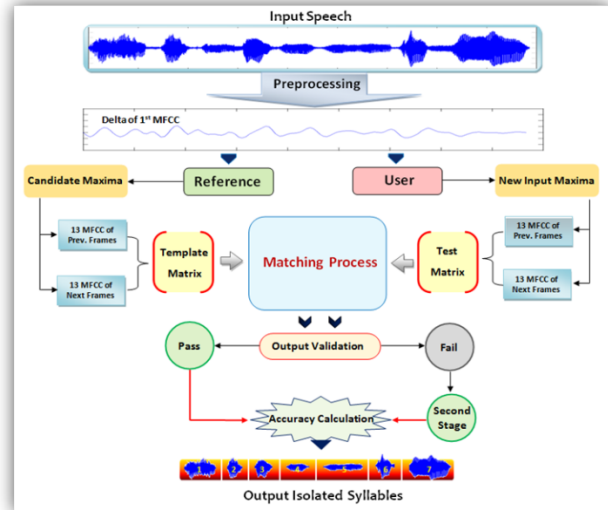


Fig. 1. Schematic diagram of the proposed system

Fig. 2 shows the procedure of obtain MFCC from the audio signal and Fig. 3 illustrates how local maxima are extracted from delta function of 1st MFCC along speech signal.

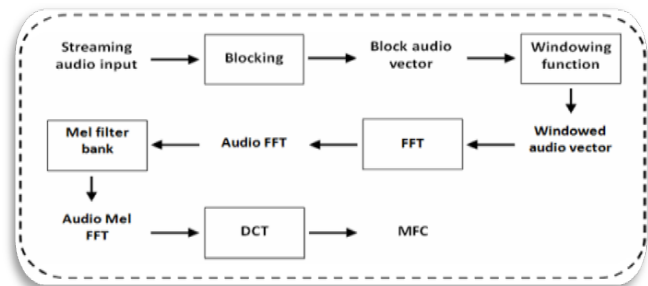


Fig. 2. Obtaining the feature vector from speech stream

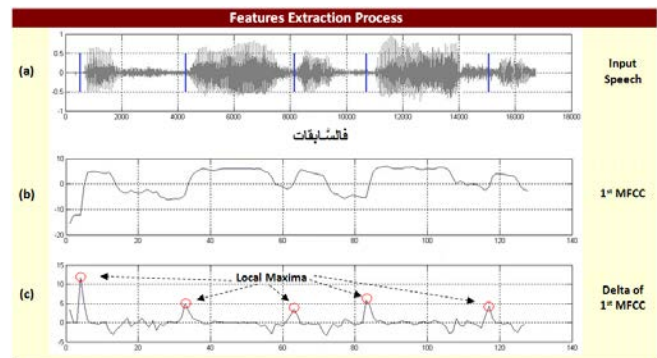


Fig. 3. Local maxima extraction from delta function of 1st MFCC. (a) Input speech signal. (b) 1st MFCC along whole signal. (c) Delta function of 1st MFCC with a selected local maxima

Fig. 4 clarifies the template matrix creation through selection of candidates. There are two factors affecting the accuracy of getting candidate that identify a syllable boundary at the template utterance, the frame length and/or the percentage of frames overlapping and the ML value “number of holding local maxima”, as illustrated in Fig. 5.

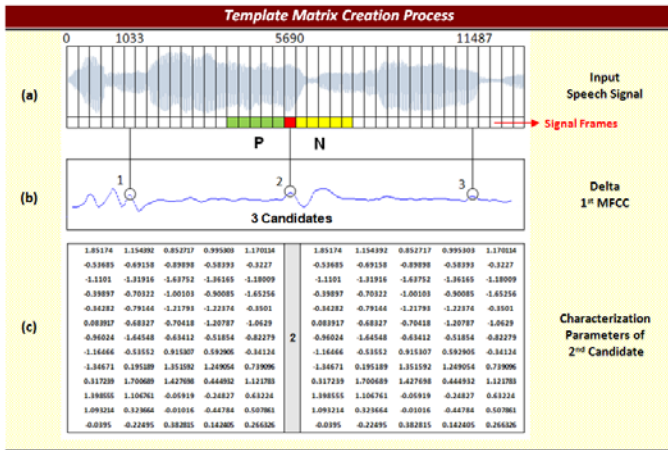


Fig. 4. Template matrix of candidates’ parameters (a)Input signal. (b) Delta function of 1stMFCC with candidate maxima. (c) Characterization parameters of the 2nd candidate

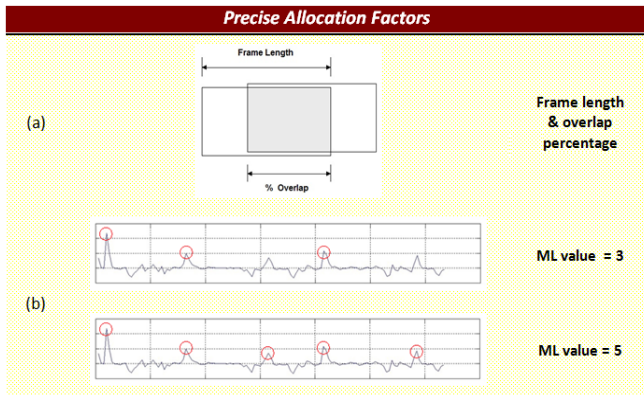


Fig. 5. Factors affect the precise allocation of syllables through hand tuning (a) Frame length and percentage of frames overlapping, (b) Delta function of 1st MFCC with “ML value=3” and “ML value=5”

D. Process the new input utterance

This module is responsible for processing the new input utterance that needs to be segmented. Each verse (utterance) has a fixed “ML value” to extract the local maxima from (PD-1st-MFC), depending on the syllabic structure of the utterance. The limiting value should be more than twice the number of existing syllables in the audio signal. Lower value of this limit results in less boundaries detection and more occurrences of connected syllables and vice versa. According to the pre-specified picked local maxima from user input, characterization parameters are obtained constituting the test matrix. The matching process is now ready to be applied between Template Matrix “output of the second module” and Test Matrix “output of the third module”.

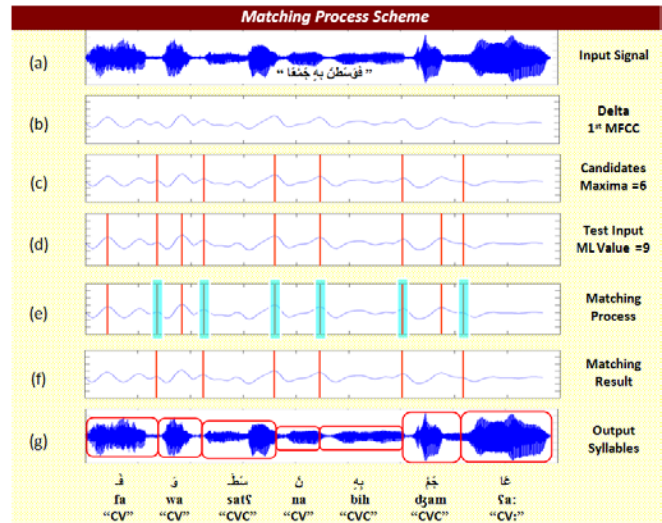


Fig. 6. Segmentation of speech syllables from input signal through template matching (a) Input Signal. (b) Smoothed Delta function of 1stMFCC along whole signal. (c) Candidate maxima selection using hand tuning. (d) Peaks configuration of the test input with ML value=9. (e) Matching process between (c), (d). (f) Resultant maxima from matching process. (g) Output segmented syllables with Arabic/IPA labels

E. Automatic identification of syllables boundaries through matching process

Identification of syllables boundaries of the user utterance is carried out by comparing its characterization parameters with the stored ones of the reference utterance using the matching technique schemed as shown in figure 6. Several methods can be used for the formulation of the rules in matching process based on distance measures techniques like Euclidean distance [24], Mahalanobis distance and Saito divergence [25]. The Euclidean distance is used for measuring closeness throughout the matching of this module. The final allocation of boundaries is obtained by passing the output of matching process through two stages of decomposition as will discussed in the next two subsections.

a) Primary Allocation of Syllables:

Table 1 shows the matching result as distances measure between user and template parameters. The closest local maxima are identified according to the minimum distance as displayed at the last row of table 1. These maxima represent locations of the target boundaries.

TABLE I. MATCHING RESULT BETWEEN TWO IDENTICAL UTTERANCES FROM THE SPEAKER (HSARY)

112 001		(Input utterance from "HSARY")											ML value = 13	
		1	2	3	4	5	6	7	8	9	10	11		
(Template utterance) Candidate Maxima =6	1	37	0	30	29	37	43	42	37	34	46	50	38	42
	2	34	30	0	23	19	31	30	31	29	29	34	39	40
	3	38	37	19	28	0	29	29	32	32	30	31	37	38
	4	43	34	29	22	32	35	28	25	0	34	43	39	47
	5	44	50	34	40	31	36	37	40	43	26	0	37	34
	6	37	38	39	33	37	29	32	33	39	43	37	0	29
BP		2	3		5				9		11	12		

TABLE II. MATCHING RESULT BETWEEN TWO UTTERANCES OF DIFFERENT SPEAKERS (HSARY & AUOOB)

112 001		(Input utterance from "AUOOB")													ML value = 13	
		1	2	3	4	5	6	7	8	9	10	11	12	13		
(Template utterance) Candidate Maxima = 6	1	31	42	40	44	46	47	46	48	43	38	38	37	--		
	2	44	35	48	47	46	48	41	38	42	41	41	38	--		
	3	49	39	43	40	30	39	43	39	47	45	46	43	--		
	4	42	35	39	37	37	40	32	41	42	39	40	38	--		
	5	55	42	42	40	37	39	43	35	43	33	47	46	--		
	6	53	49	45	48	45	41	45	42	32	35	37	30	--		
BP		1	2			5		7			10		12			

The first case in table 1 is the matching result between two identical utterances “Qul huwa ?allahu ?ahad , فُلُّنْ هُوَ اللهُ أَحَدٌ ,” for same speaker (HSARY), therefore the closest distances are zeroes, this ensures the efficiency of the algorithm.

Each utterance from the test dataset is processed through the matching module with a template of the reference speaker. As reported in the last row of table 2, the boundaries of targeted syllables are represented by the local maxima of indexes (1st, 2nd, 5th, 7th, 10th & 12th) where these maxima had the best closeness among the picked thirteen maxima from (PD-1st-MFC) of the reader (AUOOB) with the six candidate maxima from the template utterance of the reader (HSARY).

b) Connected Syllables Breakup:

In some cases the output has one or more connected syllables, as shown in Fig. 7. This phenomenon appears as a result of one of the following interpretations:

- a. Adjusted “ML value” is not enough to get the ideal number of local maxima, so that the maximum which represents the missing boundary was not taken it into account.
- b. Framing duration is big rather while one or more of speech articulations disappeared inside one frame, since the frame is the smallest unit in the speech signal and

- should be selected precisely to avoid merged or spited syllables.
- c. The recorded audio has a composite noise at this point, where the selected feature as a tool for segmentation is unable to detect the boundary between syllables in this area.

In this situation, detection of the missed boundary is performed in a semi-automatic manner. Local maxima are picked with a number equal to the missing boundaries.

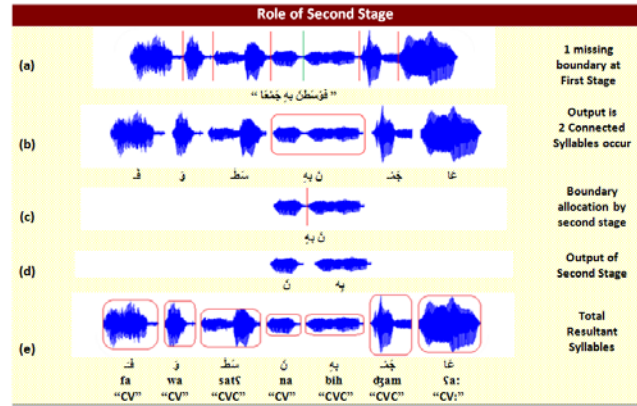


Fig. 7. The Role of second stage in the fourth module (a)One missing boundary at the first stage. (b) Two connected syllables occurrence. (c) Missing boundary allocation by the second stage. (d) Output isolated syllables by the second stage. (e) Total resultant syllables

IV. SEGMENTATION RESULTS

Since the system is speaker independent, utterances from different speakers were tested with an overall accuracy of 91.5 % as shown in table 3.

As an example, the results of applying the algorithm for segmenting the utterance “Qul huwa ?allahu ?ahad , فُلُّنْ هُوَ اللهُ أَحَدٌ” from different speakers are tabulated in table 4.

TABLE III. ACCURACY RESULTS OF THE TEST SAMPLES

No.	Verse Code	Verse Text Arabic/IPA	Syllables Count			Accuracy
			Reference	Correct	Error	
1	110001	إِذَا جَاءَ نَصْرُ اللَّهِ وَالْفَتْحُ ʔiðɑ: dʒɑ:ʔɑ nasrullahi walfath	120	115	5	95.83 %
2	110002	وَرَأَيْتَ النَّاسَ يَدْخُلُونَ فِي دِينِ اللَّهِ أَفْوَاجًا waraʔajta ʔalnasa yadxulu:na fi: di:nillahi afwadʒɑ:	216	204	12	94.44 %
3	110003	فَسَبِّحْ بِحَمْدِ رَبِّكَ وَاسْتَغْفِرْهُ إِنَّهُ كَانَ تَوَّابًا fasabbih bihamdi rabbika wastayfirhu ʔinnaui kana tawwaba:	252	201	51	79.76 %
4	111001	تَبَّتْ يُدَا أَبِي لَهَبٍ وَتَبَّ tabbat jada: ʔabi: lahabin watab	120	111	9	92.50 %
5	111002	مَا أَغْنَىٰ عَنْهُ مَالُهُ وَمَا كَسَبَ ma: ʔaynā ʔanhu ma:luhu wama:kasab	144	140	4	97.22 %
6	111003	سَيَصْلَىٰ نَارًا ذَاتَ لَهَبٍ sajasʕlā na:ran ðɑ:ta lahab	108	107	1	99.07 %
7	111004	وَأَمْرًا ذَاتَ حَمَالَةَ الْخَطَبِ wamraʔatuhu hammalata ʔalhatab	108	104	4	96.30 %
8	111005	فِي جِيدِهَا حَبْلٌ مِّن مَّسَدٍ fi: dʒi:diha hablun min masad	108	106	2	98.15 %
9	112001	قُلْ هُوَ اللهُ أَحَدٌ qul huwa ʔallahu ʔahad	84	81	3	96.43 %

10	112002	اللَّهُ الصَّمَدُ ʔallahu ʔalsʕamad	60	40	20	66.67 %
11	112003	لَمْ يَلِدْ وَلَمْ يُولَدْ lam jalid walam ju:lad	72	71	1	98.61 %
12	112004	وَلَمْ يَكُنْ لَهُ كُفُوًا أَحَدٌ walam jakun lahu kufuwān ʔahad	108	97	11	89.81 %
13	113001	قُلْ أَعُوذُ بِرَبِّ الْفَلَقِ qul ʔaʕu:ðu birabbi ʔilfalaq	108	101	7	93.52 %
14	113002	مِنْ شَرِّ مَا خَلَقَ min ʕarri ma: xalaq	72	65	7	90.28 %
15	113003	وَمِنْ شَرِّ حَاسِقٍ إِذَا وَقَبَ wa min ʕarri ʔa:siqin ʔiða: waqab	120	112	8	93.33 %
16	113004	وَمِنْ شَرِّ النَّفَّاثَاتِ فِي الْعُقَدِ wa min ʕarri ʔalnaffa:ða:ti fi: ʔilʕuqad	120	113	7	94.17 %
17	113005	وَمِنْ شَرِّ حَاسِدٍ إِذَا حَسَدَ wa min ʕarri ha:sidin ʔiða: hasad	120	113	7	94.17 %
18	114001	قُلْ أَعُوذُ بِرَبِّ النَّاسِ qul ʔaʕu:ðu birabbi ʔilna:s	96	91	5	94.79 %
19	114002	مَلِكِ النَّاسِ maliki ʔilna:s	48	35	13	72.92 %
20	114003	إِلَهِ النَّاسِ ʔilahi ʔilna:s	48	40	8	83.33 %
21	114004	مِنْ شَرِّ الْوَسْوَاسِ الْخَنَّاسِ min ʕarri ʔilwaswa:si ʔilxanna:s	96	88	8	91.67 %
22	114005	الَّذِي يُوسِّسُ فِي صُدُورِ النَّاسِ ʔllaði: juwaswisu fi: ʕuduri ʔilna:s	132	125	7	94.70 %
23	114006	مِنَ الْجِنَّةِ وَالنَّاسِ min ʔaldjennati waʔalna:s	84	81	3	96.43 %
			2544	2341	203	91.48%

TABLE IV. SPEAKER INDEPENDENCY TEST RESULTS

Test Utterance : قُلْ هُوَ اللَّهُ أَحَدٌ
 Utterance Type : Quranic Verse ML value : 13
 Utterance Code : 112001 Actual Syllables : 7 (Bulk:0)
 Reference Reader: HSARY Segmentation Stages: 2

No.	Reader Code	Start/ End Trim	# O/P Syllables					Calculated Accuracy	Total Accuracy
			First Stage			Second Stage			
			Matched	Merged	Split	Matched	Un-matched		
1	HSARY	0/0	7	-	-	-	-	100	96.5 %
2	AKHDR	1/0	2	3,2	-	7	-	100	
3	AUOOB	0/1	6	-	1	6	1	86	
4	BASIT	1/1	3	2,2	-	7	-	100	
5	GHMDY	0/1	5	2	-	7	-	100	
6	HZYFI	1/1	2	2,3	-	7	-	100	
7	JBRIL	0/1	5	2	-	7	-	100	
8	MAQLY	1/1	6	-	1	6	1	86	
9	QASIM	1/1	5	2	-	7	-	100	
10	REFAY	0/1	2	5	-	6	1	86	
11	SHTRY	0/1	3	2,2	-	7	-	100	
12	SHRIM	0/2	5	2	-	7	-	100	

V. CONCLUSION

The main purpose of this paper is to implement precise semi-automatic speaker independent system for building a database of Syllables banks from continuous Arabic uttered speech. The developed method employs the vector of local maxima picked from peaks of the delta function of first Mel Frequency Cepstrum Coefficient as cutting tools that predict

possible locations of syllables boundaries inside the continuous speech. The final boundaries are allocated by taking into account the number of segments predicted and the closeness between the predicted and the reference segment boundaries. The results have shown that the system was able to break up a set of 276 Arabic utterances into its syllables with up to 91.5 % accuracy.

VI. LIST OF ABBREVIATIONS

MFC: Mel Frequency Cepstrum.

MFCC: Mel Frequency Cepstrum Coefficient.

SNLP: Speech and Natural Language Processing.

CALL: Computer Aided Language Learning.

MSA: Modern Standard Arabic.

ZCR: Zero Crossing Rate.

PD-1st -MFC: Peaks of Delta function of the first Mel Frequency Cepstrum Coefficient.

ML value: Maximum value of picked Local Maxima from the utterance for processing.

IPA: International Phonetic Alphabet.

BP: Boundary Position.

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Power-Controlled Data Transmission in Wireless Ad-Hoc Networks: Challenges and Solutions

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Abstract—Energy scarcity and interference are two important factors determining the performance of wireless ad-hoc networks that should be considered in depth. A promising method of achieving energy conservation is the transmission power control. Transmission power control also contributes to the mitigation of interference thereby promotes throughput by means of rendering multiple hosts to communicate in the same neighborhood simultaneously without impairing each other's transmissions. However, as identified previously in the literature, traditional hidden terminal problem gets deteriorated when transmission power control mechanism is intended to be applied. In this article, we discuss the primary details about the power usage and throughput deficiency of the traditional 802.11 RTS/CTS mechanism. Improvements by means of power control are introduced as well as the solutions to the challenges likely to emerge because of the usage of diverse power levels throughout the network.

Keywords—ad-hoc networks; energy conservation; power control; throughput

I. INTRODUCTION

Wireless ad-hoc networks (WANETs) have gained increasing popularity because of the ubiquitous communication needs with satisfying costs. Data communication is irrevocable for many people and applications in fields such as military, medical and industry [1]. Moreover, the easy, low cost deployment and set-up of WANETs have played an important role in the proliferation of that technology in fields such as hotels, airports, fabricates, surveillance areas, etc. [2]. A WANET consists of mobile wireless devices that can start communication immediately when get together in a reasonable proximity which is also called the transmission distance. In doing so, the lack of need for a centralized coordinator to administrate the communication and also an infrastructure for data transmission are the most prominent features of WANETs [3] that made them a globally prevalent technology especially for the last mile. Furthermore, instead of the need of a direct communication establishment to a central coordinator (no connection unless the distance is inside the transmission range) in order to connect to the internet, hosts can maintain data transmission by means of relaying each other's data through the gateway point which is called multi-hop communication. Besides, any group of nodes in the network can start communication with each other without redundantly sending their data to an access point [4].

Though it is indisputable that WANETs facilitate many processes in a variety of application fields, they unfortunately

embody serious characteristic challenges and problems to be concerned with. The two important challenges of wireless communications is the bandwidth scarcity and limited energy supplies of the mobile devices used. Since mobile devices are battery enabled and the communication unit drains a significant portion [5-6], it is intended to develop highly energy-efficient protocols and architectures to be utilized in WANETs. Those researches that are focused on energy and bandwidth efficiency mainly developed for the second layer of the protocol stack that is Medium Access Control (MAC). MAC layer is of particular importance for WANETs because it coordinates the multiple accesses to the common transmission medium among hosts [7], and also maintains the reliability of the transmissions.

MAC layer contributes to the energy conservation in a way by coordinating also the transmission power levels of the hosts thereby preventing the redundant energy consumption because of the data transmission at maximum power level. That is, instead of a node to transmit its data to the targeted host with a constant high power unnecessarily, it is possible to use a reasonable lower power level to achieve a desired data communication which is called power controlled data transmission (PCDT). By employing PCDT mechanism, hosts can arrange their power levels according to the distance to the next hop of the packets dynamically. Since it is possible to define different routes for a stream of data packets, next hops that will relay them through the ultimate receiver may also change. That is, conveying data to a closer point requires lower energy consumption.

Utilizing dynamic power levels dynamically by means of PCDT also exploits interference reduction. Since lower energy signals do not propagate to further distances, the size of the area they affect in terms of interference also reduces. By this way, more hosts possibly stay out of the interfering range of the node and can start communication without being affected by the ongoing transmission which consequently results with throughput improvement in the network.

One of the primary challenges in wireless communications to be considered is the hidden terminal problem that is firstly identified by Tobagi and Kleinrock [8-9]. Since it is not possible for a sender to detect a collision at the receiver side, hosts lay at the opposite side of the receiver are hidden to the sender. Thus, the sender will not be able to recognize an ongoing receive process at the receiver and might start to transmission which will result as a collision at the receiver. The fundamental solution offered to remedy the hidden-node

problem is the Multiple Access with Collision Avoidance (MACA) [10] that utilizes a handshake mechanism by exchanging signaling messages called Request-to-Send (RTS) / Clear-To-Send (CTS). Taking reference the MACA method, many research activities have been performed to achieve improvements and amendments.

Unfortunately, PCDT resurrects the hidden-terminal problem which has been obviated previously, because using a dynamic power level lower than a definite one leads to an expansion of the interference range at the receiver side which turns out new hidden terminal candidates to emerge that previously not. Many research activities have been performed in order to mitigate this newly emerged hidden-terminal challenge induced by the power control mechanism. In this article, we discuss the primary details about the power usage and throughput deficiency of the traditional 802.11 RTS/CTS mechanism. Improvements by means of power control are introduced as well as the solutions to the challenges likely to emerge because of the usage of diverse power levels throughout the network.

The rest of this paper is organized as follows. Section II gives details about the fundamental concepts such as traditional hidden and exposed terminal problems, spatial reusing as well as some of the solutions offered so far in the literature. Section III introduces power controlled transmission challenges and solutions. Finally, Section IV concludes the paper and provides an outline of future directions.

II. PRELIMINARIES

One of the most important layers of the communication protocol stack is the portion of the second layer that is called MAC. As is known, MAC layer coordinates the access to the common transmission medium as fair as possible among the sharing hosts. IEEE 802.11 standard [11] is the de facto standard preferred for local area communications. MAC protocols are roughly classified into two categories as: deterministic schedule-based protocols and stochastic contention-based protocols [12].

Schedule-based protocols such as 802.11 Point Coordination Function (PCF) depends on strict time synchronization; which is difficult to maintain in terms of resource consumption that is very critical for some types of networks such as Wireless Sensor Networks (WSNs). Each node in the network is assigned a dedicated slot permanently or dynamically depending on the type and Quality-of-Service (QoS) requirements of the application or network. However, once this complex synchronization challenge is handled successfully, guarantees for the demands such as delay, bandwidth, etc. can be achieved easily. In contrast, contention-based protocols such as IEEE 802.11 distributed Coordination Function (DCF) do not give guarantees like delay or bandwidth. There is not any central coordinator in the network that allocates resources among the hosts. Every node senses and competes for the common transmission medium and starts transmission unless it detects any signal that is already on the way. Otherwise, it refrains itself from transmission and makes attempt later some time. The lack of the need for the synchronization of the hosts and presence of a central coordinator makes contention-based protocols relatively easier

to impose [13-14]. Thus, substantial portion of the efforts have been devoted to the development of new ideas or improvement of the former contention-based protocols.

As mentioned in the previous section, the fundamental approach applied in contention-based protocols is the RTS/CTS mechanism. The RTS/CTS mechanism remedies one of the most crucial challenges in wireless communication that is hidden terminal problem. Details about the hidden terminal problem and RTS/CTS mechanism are introduced in the following section.

A. Hidden Terminal Problem

The hidden terminal problem is one of the most important challenges of the wireless communication to be handled carefully. As shown in Fig. 1, say host A starts a transmission to node B. Meanwhile, host C also intends to start communication with host B. Since host C is out of the transmission range of host A, is unaware of the ongoing transmission. Signals of both host A and C will collide at host B unwittingly which is called the hidden terminal problem [15-16].

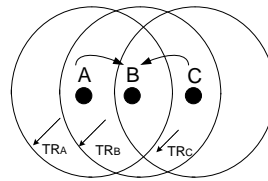


Fig. 1. An illustration of the hidden terminal problem

In the above-mentioned scenario as depicted in Fig. 1, it is not possible to prevent a possible collision by means of a Physical Carrier Sensing (PCS) mechanism. Thus, an alternative approach primarily proposed to prevent the hidden terminal is the MACA protocol which employs Virtual Carrier Sensing (VCS) mechanism rather than PCS. In this mechanism, the sender firstly sends a RTS control message including the amount of data it intends to send. Neighbors of the sender refrains themselves long enough from attempting to access the common transmission medium in order the CTS control message sent by the receiver to be acquired without any distortion at the sender side.

Meanwhile, when the receiver gets the RTS message, replies back with a CTS message also including the information about the duration of the communication. Neighbors that are in the vicinity of the receiver overhear the CTS message and defer their (if) intended transmissions during the data transmission period that will take place. As discussed in [9], many followers, amendments and new alternatives to the MACA have been proposed in the literature to date. However, the primary standard 802.11 is derived from the MACA protocol [17].

Depending on their receivers, message types in Local Area Networks (LANs) can be roughly split into two main categories:

- Broadcast messages that are not targeted to a specific receiver, i.e. every node in the vicinity of the sender are desired to get the packet.

- Receiver specific messages that are destined to a specific host in the vicinity of the sender.

In infrastructure types of LANs, since there is a central coordinator, (i.e. access point), the broadcast packet generated by the sender is sent firstly to the access point by utilizing RTS/CTS mechanism. Access point then broadcasts it through the network. Unicast packets are also sent in the same way from the sender to the access point.

In ad-hoc type LANs, however, the mechanism is different for broadcast and unicast messages. As is known, for broadcast messages, there is no need to identify a receiver id in the packet. However, for the unicast packets, the receiver address should be included. Since there is not any central coordinator in an ad-hoc network, unicast packets are directly destined to the receiver host by a preceding RTS/CTS message exchange, whereas broadcast messages are directly pumped into the medium without any processor message exchange. That is because, for a broadcast message, it is not definite for the sender that how many CTS messages should be retrieved in order to start the data transmission. Thus, in ad-hoc networks, no RTS/CTS type control mechanism is applied during broadcast message transmission [18].

B. Spatial Reusing & Exposed Terminal Problem

Another drawback of the wireless communication is the limited bandwidth resource. Since the frequency interval utilized for WLANs have become short of the capacity for the excessing demands of the users, researchers have been pursuing for the new solutions to mitigate this capacity shortage problem. The endeavor performed about capacity facilitation has been done at different layers of the protocol stack such as new data compression methods at the application layer, modulation techniques at the physical layer and MAC protocols at the data link layer.

The network capacity is highly related with the spatial reusing which can be described as a measure of the degree of the reuse of the spectrum per space or also can be expressed as the number of concurrent transmissions that can occur in the network without interfering each other [19-20]. By increasing the number of simultaneous transmissions that proceed in the network by means of spatial separation improves spatial reuse which inherently increases the overall network throughput. To achieve spatial reuse maximization, MAC protocol must coordinate the hosts to access the common medium depending on their distances to avoid interfering another ongoing transmission whilst not to prevent an upcoming nondestructive transmission but declared as possibly interferer because of the traditional PCS and VCS mechanisms [21].

4-phase MAC protocol of 802.11 alleviates highly the hidden terminal problem, whilst posing another challenge called exposed terminal problem which is a critical issue to be considered in order to improve spatial reuse [22]. As depicted in Fig. 2, although a transmission from node C to node D does not affect a transmission from node B to A; node C defers its transmission to node D unnecessarily because of the overheard RTS message sent by node B. A variety of research activities (e.g. [23-24]) have been performed to alleviate the exposed terminal problem thereby to increase the spatial reusing in the

network which inherently results with the overall network throughput. An alternative solution offered by the community is the power control during transmissions that is discussed in detail in the following section.

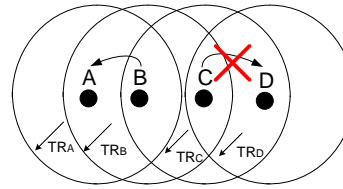


Fig. 2. An illustration of the exposed terminal problem

C. Power Control

As mentioned previously, limited capacity and power supply of the mobile devices are the primary drawbacks of wireless networks. Methods are being considered to improve spatial reuse and energy conservation for the sake of capacity and energy prudence. Power control is one of the choices offered for this purpose. Power saving can be achieved in two ways:

- Changing states of the devices to low-power mode during idle periods which the 802.11 Power Saving Management (PSM) is based on. Research studies to date have revealed that the highest energy consumption is attributed to the data communication unit [25-26]. However, if the hosts stay in the idle state longer times when compared with the data transmission period, it is not the best way purely striving to save power by transmission power control [27]. It is identified that a significant amount of energy is consumed by the devices at idle states even though they do not make any transmission [28]. Therefore, it is deduced that it is better for devices to change their states to passive position at which they require minimum energy rather than holding at a constant level [29].

- Original IEEE 802.11 applies constant power level during transmissions that is one of the primary prohibitive factors for the spatial reuse. Beyond the exposed terminal problem, it is sometimes possible for two transmissions to take place simultaneously without distorting each other by means of power control despite the RTS/CTS mechanism as illustrated in Fig. 3.

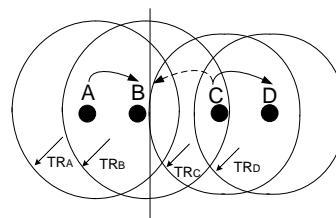


Fig. 3. An illustration of the power controlled data transmission

As depicted in Fig. 3, during a transmission between nodes A-B, a concurrent transmission can also take place between C-D, if node C can arrange its transmission power to the level that signals arriving to node B will not collide with the ones generated by node A.

III. POWER CONTROLLED TRANSMISSION CHALLENGES AND SOLUTIONS

As stated in the previous section, power controlled transmission can yield to significant energy savings as well as spatial reuse improvements that ultimately results with overall throughput enhancement. However, new challenges emerge with the power-controlled transmission, such as Hidden Terminal Jamming and Power Control Induced Hidden Terminal Problems. Following sections introduce these crucial issues that can be handled by different methods which will be discussed subsequently.

A. Hidden Terminal Jamming Problem

One of the challenges emerge by the power-controlled transmission is the Hidden Terminal Jamming that occurs during two concurrent transmissions of which different power levels employed for each.

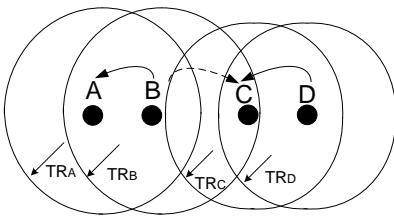


Fig. 4. Hidden Terminal Jamming Problem

In the scenario depicted in Fig. 4, say node C starts a transmission to node D. Meanwhile, node B starts a new transmission which is destined for node A. It is assumed that, all the control messages (RTS/CTS) and data packets are exchanged with the same power level. That is, if node sends a RTS message with a power level P_t , it continues to send data packet with the same power level. Since this power level is lower enough that none of the RTS/CTS messages of node C arrive at node B; B is unaware of the ongoing transmission and starts to send its signals with a power level higher that covers also node C. Therefore, signals sent by node D collide with the signals sent by node B, which is called the Hidden Terminal Jamming Problem [27, 30]. An idea offered for mitigating the Hidden Terminal Jamming Problem is that, instead of using the same power level also for control messages, a higher power can be applied for these packets to make it possible to transmit them to every neighbor in the vicinity to notify them about the upcoming transmission. That is, RTS/CTS control message transmission should be performed at the maximum power level possible, but DATA/ACK packets can be transmitted at a minimum power level related with the distance [27, 31-32] as shown in Fig. 5.

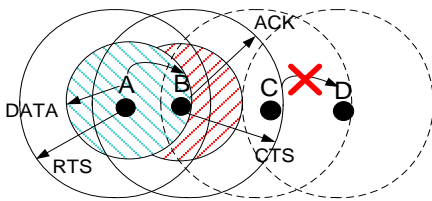


Fig. 5. Power level differentiation to mitigate the hidden terminal jamming problem

B. Spatial reuse performance analysis of 802.11 RTS/CT

Ye et.al [35] analysis the spatial reuse performance of IEE 802.11 provided that all the hosts in the network use the maximum power level during transmissions. In order to understand the challenges fairly, some definitions are introduced below [33-35]:

Transmission Range (RT): is the range within that a packet transmission is performed successfully and retrieved by the receiver correctly under the assumption that any other radio interference occurs.

Interference Range (RI): is the range within that the receiver host can be interfered by a sender(s) of another irrelevant transmission(s).

Most of the research studies [36-38] base on the two-way ground reflection model [39-40] as follows:

$$P_r(d) = \frac{P_t G_t G_r h_t^2 h_r^2}{d^{nL}} \quad (1)$$

where $P_r(d)$ is the signal power at the receiver, P_t denotes the transmitter power, G_t and G_r express the antenna gains respectively, h_t and h_r are the antenna heights, d is the distance, and L is the system loss. n is the path loss exponent ranging from 2 to 6 depending on the pattern and environment that the communication takes place. A list of well accepted values for different types of environments is given in Table 1:

TABLE I. VALUES OF THE PATH LOSS EXPONENT n FOR DIFFERENT ENVIRONMENTS

Environment	n
Free space	2
Urban area	2.7-3.5
Shadowed urban area	3-5
Indoor	4-6

By omitting the other parameters as done in almost all previous work and only considering the distance between the communicating pairs, Equation (1) takes the form:

$$P_r(d) = P_t d^{-n} \quad (2)$$

The relationship between the received (P_r) and interfering power levels at the receiver (P_i) is given as follows:

$$SIR = \frac{P_r}{P_i} = \left(\frac{R_i}{d}\right)^n \geq SIR_{Thr} \quad (3)$$

where SIR denotes the signal-to-interference ratio (ambient noise is neglected) and SIR_{Thr} is the threshold capture value that is the ratio of the essence signal to the interfering signal arrived from another irrelevant transmitter. SIR_{Thr} is the reference value that the SIR value should be above in order the retrieved signal to be justified as meaningful and valuable. In their study, the values assigned to SIR_{Thr} and n are 10 and 4 respectively; say that the indoor environment is considered. When the abovementioned values are put properly, equation (3) takes the form:

$$SIR = \left(\frac{R_i}{d}\right)^4 = 10 \quad (4)$$

$$R_i = 10^{1/4} d \cong 1,78 d \quad (5)$$

As clarified in equation (5), the interference range (R_i) only depends on the distance between the sender and receiver; i.e. there is not any relationship between R_i and R_t .

Regarding to their comments, when the maximum transmission power level is utilized permanently, three situations emerge that should be considered separately:

- *Underactive Scenario:* In this scenario, the value of d lays at the interval: $(R_t/1,78) \leq d < R_t$. They express that in ns-2 and WaveLAN, R_i is determined as 250m for the maximum transmission power value of P_t . Thus, the values of the interval becomes as : $(0,56R_t) \leq d < R_t = 141 \leq d < 250$
 R_i is larger than R_t in this scenario as illustrated in Fig. 6:

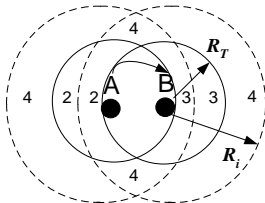


Fig. 6. $0,56 * R_t \leq d < R_t$.

A potential collision will be induced by the hosts deployed in the regions denoted by 4, since these nodes are out of the transmission range (R_t). Thus, they are unaware of the ongoing transmission and might start to send their own signals destined to a different receiver which will collide with the former.

- *Moderate Scenario:* $R_i \leq R_t$ in this scenario, of which d lays at the interval $(R_t/2,78) \leq d < (R_t/1,78) = 90 \leq d < 141$
in Lucent's WaveLAN and ns-2.

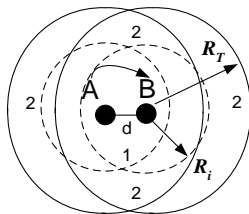


Fig. 7. $(R_t/2,78) \leq d < (R_t/1,78)$.

Despite the hosts deployed in the regions denoted by 2 do not interfere they will defer their transmissions regarding the reception of RTS/CTS control messages. Thus, 802.11 RTS/CTS VCS mechanism gives erroneous alarm and confines the throughput of the network.

- *Overactive Scenario:* $R_i < R_t$ in this scenario which models the real-life office, building or house LANs with approximate range of 100 m coverage. $d < (R_t/2,78) = d < 90$.

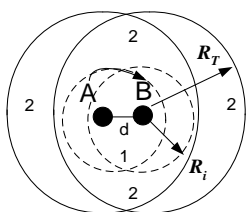


Fig. 8. $d < (R_t/2,78)$.

As depicted in Fig. 8, though it is possible for concurrent transmissions to take place; unproductive region size increases. Hosts deployed in the regions expressed by 2 are not allowed to transmit because of the prohibitive behavior of 802.11 RTS/CTS VCS mechanism.

In essence, Scenario 2-3 concludes with the same result which is a reduction of the spatial reuse. In Scenario 1, the hidden terminal problem emerges this time. These scenarios clarify the deficiency of 802.11 RTS/CTS mechanism for the hidden terminal and spatial reuse issues. Besides the performance analysis of 802.11 RTS/CTS mechanism for spatial reusing, Ye et.al [35] propose a method to achieve improvement for the overactive case. That is, hosts deployed in region 2 in Fig. 7 that hear either a RTS or CTS but not both, can start their transmissions because they will not affect the ongoing transmission that take place between A-B.

C. Power Control Induced Hidden Terminal Problem

Significant improvements in energy saving are intended by means of power-controlled data transmission; especially for the type of networks in which data transmission periods takes considerably comparable with the idle state intervals in contrast to the wireless sensor network. Moreover, by decreasing the power of the transmitted signals, the range of the area that will be interfered shrinks as well. Hence, a possible concurrent communication in vicinity can take place without any distortion. However, the methods suggested previously such as utilizing different power levels for control and data packets can induce a new problem that is called as the Power Control Induced Hidden Terminal Problem (POINT) [33].

In the previous section, it is identified that, since the maximum transmission power is used permanently by all of the hosts in the network, R_i does not depend on the transmitting power (P_t). Shih et.al [33] discuss that, in order for a transmission to be valid, two conditions given below should be satisfied:

- The strength of the received signal should be greater than a threshold value: $P_r \geq P_{Thr}$.
- The ratio of the received signal strength to the total interfering signals strength that is the SIR should be also greater than the SIR_{Thr} .

$$SIR = \frac{P_r}{P_i} \geq SIR_{Thr} \quad (6)$$

$$P_\alpha = \max(P_{Thr}, P_i * SIR_{Thr}) \quad (7)$$

In order to satisfy the abovementioned conditions, $P_r \geq P_\alpha$. Thus, basis on equation (2), transmitting range can be calculated as follows:

$$R_t = \left(\frac{P_t}{P_\alpha}\right)^{1/n} \quad (8)$$

Since the transmit power levels of the hosts in the network are not fixed and identical, R_i depends upon P_i as well as the remaining tolerable interference level of the receiver which is calculated as follows:

$$P_i = P_{ci} + P_{ri} \quad (9)$$

where P_{ci} , P_{ri} indicate the current strength of the total interfering signals and remaining tolerable interfering signal strength at the receiver. To satisfy the condition expressed in equation (6):

$$P_{ri} \leq \frac{P_r}{SIR_{Thr}} - P_{ci} \quad (10)$$

In the vicinity of the receiver, say the closest node (with the distance d_{XR} to the indicated receiver) denoted by X starts a transmission with the maximum transmitting power (P_{max}). The signals emerged by that node will be sufficient to result a collision and corrupt the transmission at the receiver. Thus, R_i can be calculated by considering only the signals originate from node X. As a result, equation (10) can be rewritten as follows:

$$\frac{P_{max}}{(d_{XR})^n} \leq \frac{P_r}{SIR_{Thr}} - P_{ci} \quad (11)$$

Thus, the distance d_{XR} actually identifies the minimum range that a neighbor should be in proximity to start a non-interfering transmission with the worst case of using the maximum transmission P_{max} . By this way, R_i which in essence substitutes for d_{XR} should satisfy the following condition:

$$R_i(P_t) = \left(\frac{P_{max}}{\left(\frac{P_r}{SIR_{Thr}} - P_{ci} \right)} \right)^{1/n} \quad (12)$$

In the following section, the description of the POINT problem is identified and how it can be mitigated by arranging the transmission power inferred from the equations introduced thus far. Assume a scenario as illustrated in Fig. 9, in which host A sends RTS control message with P_{max} and host B replies back with CTS message with P_{max} also.

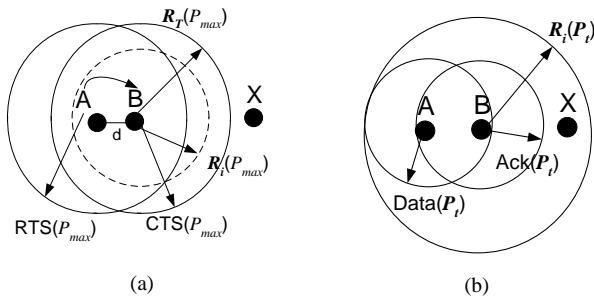


Fig. 9. A sample scenario to illustrate the POINT problem

X is unaware of the ongoing transmission since none of the RTS/CTS packets arrive it. As depicted in Fig. 9(a), X is obviously out of the interfering range and can start its own transmission. However, when host A attempts to make a transmission with a power level P_t that is lower than P_{max} , host X which was formerly out of R_i involves in the interference range. The reason is that the interference range (R_i) extends by the reduction in the transmission power (P_t).

Shih et.al [33] propose the Collision Avoidance Power Control (CPAC) MAC protocol to mitigate the POINT problem that emerges especially in the situations that $d < (R_i/2.78)$. In CPAC, instead of using the exact power level related with the distance, utilizing an appropriate power level will prevent unexpected interferers to emerge. The aforementioned appropriate power level P_{app} is calculated as follows. Since R_i and $R_i(P_{max})$ are intended to be equal in CPAC, P_{app} can be derived by considering equation (8) and (12) as follows:

$$R_i(P_{max}) = \left(\frac{P_{max}}{P_a} \right)^{1/n} = \left(\frac{P_{max}}{\left(\frac{P_r}{SIR_{Thr}} - P_{ci} \right)} \right)^{1/n} = R_i(P_{max}) \quad (13)$$

P_{app} is derived as follows:

$$P_{app} = (P_a + P_{ci})d^n SIR_{Thr} \quad (14)$$

Hence, POINT problem can be alleviated by arranging the applied transmitting power level as discussed in CAPC [33].

IV. CONCLUSIONS

The throughput and delay performance of WANETs highly depends on their ability of spatial reusing. Power control is a way of improving spatial reuse, as well as a promising solution for energy conservation. 802.11 RTS/CTS has some drawbacks from the point of view of spatial reuse and power efficiency. By applying varying transmission power level, it is desired to improve the spatial reusing. This time, new problems such as Hidden Terminal Jamming and POINT emerge that should be exhaustively considered. In this article, we discuss the primary details about the power usage and throughput deficiency of the traditional 802.11 RTS/CTS mechanism. Improvements by means of power control are introduced as well as the solutions to the challenges likely to emerge because of the usage of diverse power levels throughout the network.

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Face Recognition Based on Improved SIFT Algorithm

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Abstract—People are usually identified by their faces. Developments in the past few decades have enabled human to automatically do the identification process. Now, face recognition process employs the advanced statistical science and matching methods. Improvements and innovations in face recognition technology during 10 to 15 past years have propelled it to the current status. Due to the wide application of face recognition algorithms in many practical systems, including security control systems, human-computer interaction systems, etc., algorithms with high success rate are highly interested in research areas in recent years. Most of suggested algorithms are about correctly identifying face photos and assigning them to a person in the database. This study focuses on face recognition based on improved SIFT algorithm. Results indicate the superiority of the proposed algorithm over the SIFT. To evaluate the proposed algorithm, it is applied on ORL database and then compared to other face detection algorithms including Gabor, GPCA, GLDA, LBP, GLDP, KGWRM, and SIFT. The results obtained from various tests show that the proposed algorithm reveals accuracy of 98.75% and run time of 4.3 seconds which is shorter. The new improved algorithm is more efficient and more accurate than other algorithms.

Keywords—face detection; improved SIFT descriptor; KGWRM; GPCA; GLDA

I. INTRODUCTION

Faces play the major role in identifying people and showing the feelings in the image. The human ability to recognize faces is remarkable. This ability stands against changes in visual conditions such as facial expressions, age, as well as changes in glasses, beard or hair style. Face recognition is a key subject in applications such as security systems, credit card control, culprit identification, mugshot matching (photos taken from the face), surveillance and control of access, personal identification, as well as applications for forensic pathology and constitution enforcement. In this case, a database containing facial photos for a group of people is employed.

Then, based on unrecognized photo, we want to answer the question: "does the photo belongs to a person in the database?" Many algorithms are proposed to solve this problem [2]. In this work, a new method is proposed based on SIFT [1]. Then, it is compared to Eigen-faces and Fisher-faces algorithms [3],[4],[5]. The results obtained by the proposed algorithm show improvements compared to other algorithms.

II. LITERATURE

Eigen-faces algorithm [6] is based on reducing the size of Principal Component Analysis (PCA). The basic idea is that each image is considered as a vector in a high-dimensional space. As the input image, the target image is first transformed using Eigen vectors matrix or vectors comprising the subspace. Then, it is compared with existing data in the dimension-reduced space and the most similar image is selected as the identified one. First, the data covariance matrix and then the vectors matrix and eigenvalues are calculated. Eigen vectors matrix is composed of orthogonal vectors forming the feature subspace. Data will be independent by transforming to this subspace. An input image can be displayed as a linear combination of Eigen-faces by transforming to the new Eigen-faces space. After transforming the input image to the new Eigen-faces space, the most similar image is selected as the identified image using the nearest neighbor. To cluster the approach, two types of nearest neighbor are considered. The first input image is compared with all images in the database. Then, the nearest cluster center is obtained. Calculations are done using each cluster (the face of a person) and the cluster is selected for comparison.

Fisher-faces approach [3],[4] is based on the expansion of linear resolution analysis (LDA) in face recognition. The objective is generating a space with minimum intra-class dispersion and maximum inter-class dispersion, finding vectors providing the best classes for separation, as well as trying to maximize the difference between the classes and minimize the number of classes. Here, the mechanism is similar to that of PCA, i.e. an image is considered as point in a high-dimensional space. Then, LDA is applied on new basic vectors. This method is called Fisher-faces. Accordingly, the images are used for matching process. Again, the two nearest neighbors can be employed for the discussed process.

III. SIFT ALGORITHM

Scale Invariant Feature Transform (SIFT) is an algorithm employed in machine vision to extract specific features of images for applications such as matching various view of an object or scene (for binocular vision) and identifying objects [6]. The obtained features are invariant to scale and rotation, and partially invariant to change in lighting. Such features are extracted in four steps.

In first step, using a DOG filters, all points in the image at all scales are searched to find features invariant to the orientation and scale. In the next step, an extended model is used for any potential point to determine the appropriate points based on different sustainability criteria. In third step, based on the local gradient of the image at target points, one orientation is assigned to each feature point. Finally, the information contained in the gradient function around the feature points is encoded to be used as the point's characteristic for future works such as feature matching. The name of descriptor is taken from its mechanism, i.e. converting the image data algorithms into scale-invariant coordinates in accordance with local features. Each feature is considered as a vector in a 128-dimensional space identified over the key point neighborhood.

IV. THE PROPOSED ALGORITHM

Image matching is a basic process in image processing and plays an important role in photogrammetry and remote sensing processes. SIFT is an efficient matching algorithm which has successfully been used to match the remote sensing images automatically. However, feature extraction operator is still a major weakness of the algorithm. Despite the high computational complexity, it can be partially controlled in feature extraction from remote sensing images. To alleviate the computational complexity while improving the efficiency, this study limit the feature extraction operators in SIFT as well as key-points extraction operators to a certain number to develop an efficient way to match images. The main feature of this method is taking the advantages of a specific strategy to select the best key points. After extracting the key points and making their descriptors, matching process starts considering the Euclidean distance between the descriptors as a multilevel process using a one-to-many procedure as done. Then, using least squares model, the accuracy of the initial matching-pairs are checked to find the compatible matching. Finally, the two images are matched by determining the parameters of a segment transformation function. Practical results obtained by applying the method on different face images indicate the great efficiency of the proposed method in comparison with the standard SIFT algorithm. In the proposed method, a consistent feature is considered to be compared with another feature for which the distance is less than a certain deductions (in this project 0.79) of a distance to the next nearest feature. This will reduce the number of incorrect matches. For incorrect matches, there will be a number of other near features due to the high-dimensional existing features. On the other hand, if there is a correct match, it is unlikely to find other features due to the distinctive nature of the SIFT descriptor.

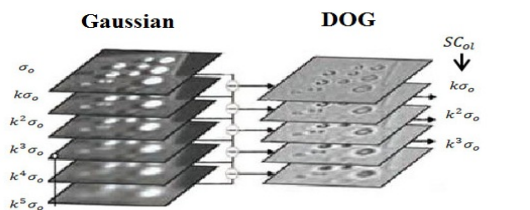


Fig. 1. The structure space is composed of a set of Gaussian images and DOG (Difference of Gaussian)

In key point extraction, the first step is to create a scale space. The structure space is a display of image structures at different scales. The structure space is composed of a set of Gaussian images and DOG (Difference of Gaussian) stored in a structure called the octave as Figure 1 shows.

To create the Gaussian images in scale space for any octave, the image must be convolved with the kernel Gaussian with varying kernel. Therefore, the Gaussian image $L(x, y, \sigma)$ is created by convolving the image $I(x, y)$ with Gaussian kernel $G(x, y, \sigma)$ through equation (1):

$$L(x, y, \sigma) = G(x, y, \sigma) \otimes I(x, y) \quad (1)$$

Where \otimes denotes the convolution operator. In Eq. 1, σ represents the scale with the initial value of 1.6 repeated increasingly using K at different levels of the octaves. Then, DoG images are calculated based on the difference between the two adjacent Gaussian images in the scale space (scale of the smaller Gaussian image is considered as the scale of the DoG image). The octave Gaussian image with the value twice the initial scale is selected, the dimension is halved by resampling, and it is considered as the primary image for the next octave. In order to establish a stable condition, the space scale of each pixel in the intermediate levels of DoG images in each octave is compared with the 8 neighboring pixels for the target DoG image and 9 neighboring pixels for each upper and lower DoG images. If it is an extreme (maximum or minimum), it will be selected as the key point. The number of scale levels in each octave (i.e. the number of searched to extract key points) is a parameter affecting the number of extracted key points is. According to Lowe, the number of Gaussian image in each octave and the number of DoG images is considered 6 and 5, respectively. In the proposed algorithm, σ is calculated as follows Eq. (2):

$$\sigma = k^{(k_1 + 2 * k_2)} * 1.6 \quad (2)$$

This change makes the number of Gaussian images per octave 4 instead of 6, thus decreasing the SIFT run time. Besides, the proposed method considers the number of key points fixed as 50 points for each image with a rating higher than others. This ultimately increases the speed of the algorithm. The number of octaves in the scale space also varies depending on the image size which is considered 3 after testing different numbers.

Scale factor of each scale levels is calculated through Eq. (3) as:

$$SC_{0l} = \sigma \cdot 2^{(o-1 + \frac{l}{LN})} = \sigma \cdot K^{LN(o-1) + l} \quad (3)$$

$$o = 1, 2, \dots, ON; l = 1, 2, \dots, LN; k = 2^{1/LN}$$

Where SC_{0l} is the scale factor for level l of octave o . Finally, using the Euclidean distance, key points between two images are selected.

V. COMPARING ALGORITHMS IN TERMS OF ACCURACY AND SPEED

ORL is considered as the input data for this study including 400 face images belonging to 40 different people taken from different angles as well as light intensity. For the first test, 100 images were randomly selected and 1 image of each person was considered as the training image (three persons were considered for training). As an example of the algorithm output, we assumed the three images in Fig. 2 as training images. Then, a randomly selected image from the three target persons was tested. A function removing all unnecessary part of the image such as background was called. Then, employing improved SIFT algorithms, key points in the training images using sift were extracted.

corresponding key points in training images, the Euclidean distance between the points was calculated and finally, the image with smaller total value was selected as the final image. Repeating the test on 100 randomly selected image resulted in 86 correct answer was. The second test was conducted to compare the proposed method with others. Here, 80% of all data was considered as training data and the remaining 20% as test data (i.e. 320 images for training and 80 ones for test). After applying Gabor, GPCA and GLDA, LBP, GLBP, KGWRCM [7] and SIFT on ORL data set, it was concluded that that the accuracy of the proposed algorithm is 6.25% more than SIFT with the runtime 1.9 seconds less than the original SIFT algorithm.

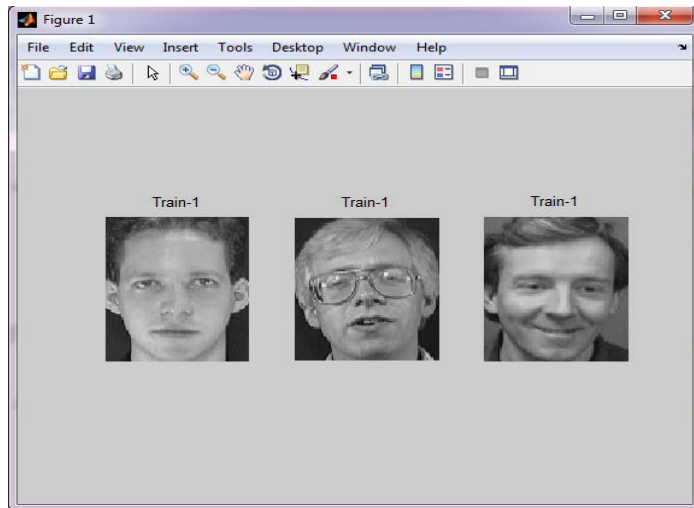


Fig. 2. As an example of the algorithm output

In the next step, an image from the three training persons was tested. Operations done on training images was repeated for the test image as shown in Figure 3. Fig. 3 shows the main test image, the test image after removing unnecessary areas, and the extracted key points.

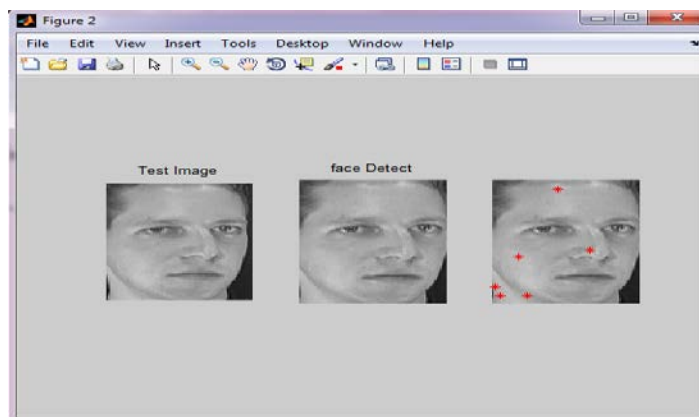


Fig. 3. Shows the main test image, the test image after removing unnecessary areas, and the extracted key points

Finally, the key points calculated for the test image were compared with the key points calculated for the training images. The comparison was so that after finding the

TABLE I. ACCURACY OF RECOGNITION ON ORL DATABASE

Algorithm	Accuracy
1)Gabor	81.34
2)GPCA	89.87
3)GLDA	97.50
4)LBP	89.61
5)GLBP	93.42
6)KGWRCM	99.32
7)SIFT	92.5
8)OUR METHOD	98.75

The graph comparing the proposed method with other methods is shown in Figure 4.

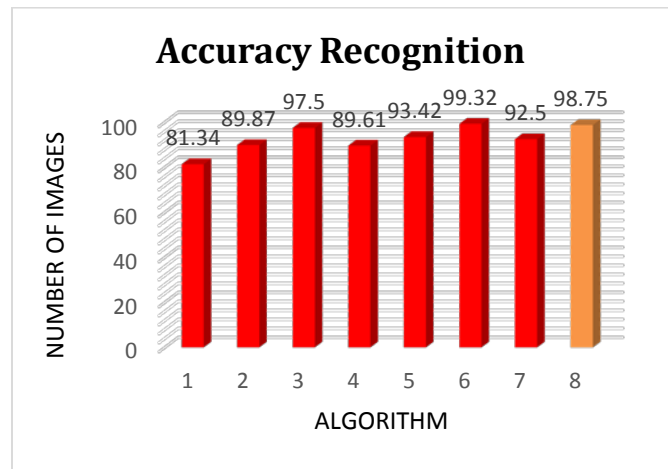


Fig. 4. Comparing the proposed method with other methods

The next comparison was conducted based on the run time indicating the significant advantage of the proposed algorithm compared to others shown in Table 2.

TABLE II. PROCESSING TIME FOR 300 IMAGES (IN SECONDS)

Algorithm	Accuracy
1)Gabor	1.8
2)GPCA	1.5
3)GLDA	7.2
4)LBP	6.9
5)GLBP	3.7
6)KGWRCM	4.9
7)SIFT	5.3
8)OUR METHOD	5.3

The graph comparing the run time of the proposed method with other methods is shown in Figure 5.

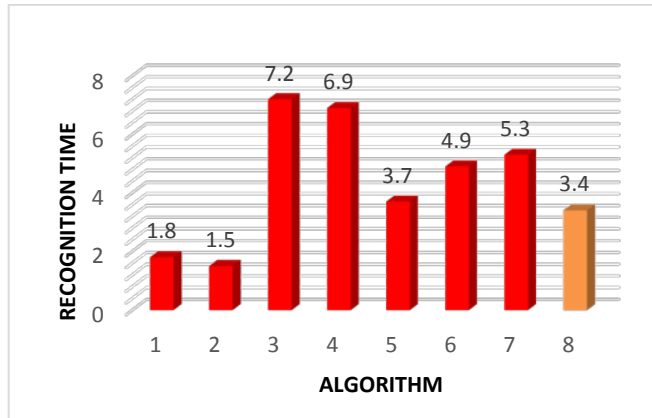


Fig. 5. The run time of the proposed method with other methods

VI. CONCLUSIONS

People are usually identified by their faces. Developments in the past few decades has enabled human to automatically do the identification process. Early face recognition algorithm used simple geometric models. Now, face recognition process employs the advanced statistical science and matching methods. Improvements and innovations in face recognition technology during 10 to 15 past years have propelled it to the current status. Face recognition is applicable for both investigation and identification. This study provided a new face recognition method based on SIFTS features. Improving

SIFT algorithm, this study focused on face recognition. Results indicated the superiority of the proposed algorithm over the SIFT. To evaluate the proposed algorithm, it was applied on ORL database and then compared to other face detection algorithms including Gabor, GPCA, GLDA, LBP, GLDP, KGWRCM, and SIFT. The results obtained from various tests showed that the proposed algorithm, with 98.75% accuracy and run time of 4.3 seconds, is more efficient and accurate than other algorithms yet the run time was less than others.

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Prediction of Mental Health Problems Among Children Using Machine Learning Techniques

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Abstract—Early diagnosis of mental health problems helps the professionals to treat it at an earlier stage and improves the patients' quality of life. So, there is an urgent need to treat basic mental health problems that prevail among children which may lead to complicated problems, if not treated at an early stage. Machine learning Techniques are currently well suited for analyzing medical data and diagnosing the problem. This research has identified eight machine learning techniques and has compared their performances on different measures of accuracy in diagnosing five basic mental health problems. A data set consisting of sixty cases is collected for training and testing the performance of the techniques. Twenty-five attributes have been identified as important for diagnosing the problem from the documents. The attributes have been reduced by applying Feature Selection algorithms over the full attribute data set. The accuracy over the full attribute set and selected attribute set on various machine learning techniques have been compared. It is evident from the results that the three classifiers viz., Multilayer Perceptron, Multiclass Classifier and LAD Tree produced more accurate results and there is only a slight difference between their performances over full attribute set and selected attribute set.

Keywords—Mental Health Diagnosis; Machine Learning; Prediction; Feature Selection; Basic Mental Health Problems

I. INTRODUCTION

Mental illness is rising at epidemic rates around the world and WHO predicted that one in four people in the world will be affected by mental and neurological disorders at some point in their lives. Depressive disorders will become the second leading cause of the global disease burden by 2020, behind ischaemic heart disease but ahead of all other diseases [1]. The growth in number of professionals who treat mental illness is very less when compared with the growth in number of people suffering from mental illness.

Mental health diagnosis involves many steps and it is not a straightforward process. Actually, the diagnosis starts with specially designed interview with questions about the symptoms and medical history and sometimes performing a physical examination. Various psychological tests are also conducted to make sure that the symptoms are caused only by the mental health and not by any other problems. Various assessment tools are available to evaluate a person for a mental illness.

As similar factors and symptoms lead to different mental health problems, the diagnoses have become a complicated

task and sometimes they may be misdiagnosed. The co-operation of the patient is very much needed in diagnosing the problem. Diagnosing children with mental health problems is more problematic than diagnosing adults. Hence, care must be taken to diagnose the mental health problem accurately.

Artificial Intelligence (AI) is the intelligence exhibited by machines. It is a field of study on creation of computer software that is capable of intelligent behavior. AI researchers have developed many machine learning techniques that help computers to imitate the human reasoning to solve puzzles or make logical deductions. Methods have also been found to deal with uncertain or incomplete information, employing concepts from probability and other fields.

This research has identified eight machine learning techniques that diagnosed mental health problems correctly over a sample data set of ten cases. A comparison is made on those eight machine-learning techniques and identified the best three which can be utilized to assist mental health professionals in diagnosing mental health problems. Five basic mental health problems of children viz., Attention problem, Academic Problem, Anxiety Problem, Attention Deficit Hyperactivity Disorder (ADHD) and Pervasive Developmental Disorder (PDD) have been considered. The factors, symptoms and various test results that are observed by the professionals are given as input to the techniques and the psychological problem diagnosed is retrieved as their output.

Attention problems in children refer to failing to pay close attention to details or making careless mistakes, not listening to when spoken directly, do not sustain attention or struggling with basic reading and math skills and so on. **Academic problems** can be very common and range from difficulties with studying and reading to anxieties about exams than practical or technical skills. **Anxiety Problems** are characterized by chronic excessive worry accompanied by restlessness, fatigue, concentration problems, irritability, muscle tension and sleep disturbance **Attention Deficit Hyperactivity Disorder (ADHD)** is one of the most common childhood disorders which may continue through adolescence and adulthood. Some of the symptoms include difficulty staying focused and paying attention, difficulty controlling behavior and hyperactivity. **Pervasive Developmental Disorder (PDD)** refers to a group of conditions that involve delays in the development of many basic skills, like, ability to socialize with others, ability to communicate and ability to use imagination.

The aim of this research is to predict basic mental health problems using machine learning techniques. Section 2 presents a literature review on mental health diagnosis using computers. Section 3 gives an overview of various machine learning techniques. Section 4 presents the methodology and the data sets used in this research to predict the five basic mental health problems. Section 5 evaluates the techniques using training and test data sets. Section 6 provides conclusion and future work.

II. LITERATURE REVIEW

The research on applying machine learning techniques in mental health diagnosis have started in nineteen eighties. DTREE [2] is an expert system that diagnoses DSM-IV Axis I disorders using 'Decision Tree' techniques. INTERNIST/AUTOSCID [3] is a computerized Structured Clinical interview for DSM-IV Axis II personality disorders. Yap R.H. and Clarke D.M. [4] developed an expert system called MILP (Monash Interview for Liaison Psychiatry) using constraint-based reasoning for systematic diagnoses of mental disorders based on DSM-III-R, DSM-IV and ICD-10. Constraint Logic Programming (CLP) language was used to develop the system. Kipli et al. [5] introduced new approaches for Data-mining and classification of mental disorder using Brain Imaging data.

Masri R.Y. and Jani H.M., in [7] proposed a Mental health Diagnostic Expert System to assist the psychologists in diagnosing and treating their mental patients. Three artificial intelligence techniques viz., Rule-Based Reasoning, Fuzzy Logic and Fuzzy-Genetic Algorithm were used for diagnosis and suggestion of treatment plans. Luxton, David D. [8] analyzed how artificial intelligence has been utilized in psychological practice. The current trend and future applications and their implications have been discussed.

Razzouk D., et al [9] developed a decision support system for the diagnosis of schizophrenia disorders with an accuracy of 66-82%. Chattopadhyay S., et al. [10] developed a neuro-fuzzy approach to grade adult depression. The supervised (Back Propagation Neural Network: BPNN and Adaptive Network-based Fuzzy Inference System : ANFIS) and unsupervised (Self-Organizing Map) neural net learning approaches have been utilized and then compared. It was observed that ANFIS being a hybrid system performed much better than BPNN classifier. L.C.Nunes et al. [11] introduced a hybrid method to handle diagnosis of Schizophrenia. They have integrated structured methodologies in decision support (Multi-Criteria Decision Analysis: MCDA) and structured representation of knowledge into production rules and probabilities (Artificial Intelligence: AI).

Basavappa S.R. et al. [12] used depth first search method with backward search strategy to diagnose depression or dementia. They developed an expert system using the patient's behavioral, cognitive, emotional symptoms and results of neuropsychological assessments. Rahman, Rashedur M. and Farhana Afroz [13] compared various classification techniques (Bayesian Network, Multilayer Perceptron, Decision Trees, Single Conjunctive Rule Learning and Fuzzy Inference Systems and Neuro-Fuzzy Inference System) using different data mining tools like WEKA, TANAGRA and MATLAB for

diagnosing diabetes. It was observed that different techniques yielded different accuracy levels with different accuracy measures like Kappa Statistic and Error rates.

David Gil A. and Magnus jhonson B. in [14], proposed a system based on Artificial Neural Networks(ANN) and Support Vector Machines(SVM) that diagnoses Parkinson's Disease. The system has shown an increase in accuracy and a decrease in cost. In [15], Gomula, Jerzy et al., tried to find efficient methods for classification of MMPI profiles of patients with mental disorder. They identified that Attribute Extension Approach improves classification accuracy in the case of discretized data. Flavio Luiz Seixas and Bianca Zadrozny in [16] proposed a Bayesian Network (BN) Decision Model for diagnosis of dementia, Alzheimer's disease and Mild Cognitive Impairment. The BN model was considered as it is well suited for representing uncertainty and causality. Network parameters were estimated using a supervised learning algorithm from a dataset of real clinical cases. Model was evaluated using Quantitative methods and Sensitivity Analysis and it showed better results when compared to most of the other well-known classifiers. Anchana Khemphila and Veera Boonjing in [17] used Multi-Layer Perceptron (MLP) with Back Propagation Learning to diagnose Parkinson's disease effectively with reduced number of attributes. Information Gain of all attributes is used as a measure to reduce the number of attributes. Pirooznia, Mehdi, et al., in [18] utilized data mining approaches for Genome-wide Association of Mood Disorders. Six classifiers namely Bayesian Network, Support Vector Machine, Logistic Regression, Random Forest, Radial Basis Function and Polygenic Scoring Approach were compared. It was identified that simple polygenic score classifier performed better than others and it was also found that all the classifiers performed worse with small number of Single Nucleotide Polymorphisms in the brain expressed set compared to whole genome set.

In [5], Kipli, Kuryati, Abbas Z. Kouzani, and Matthew Joordens detected depression from structural MRI scans to diagnose the mental health of patients. They investigated performances of four Feature Selection algorithms, namely, OneR, SVM, Information Gain (IG) and ReliefF. Finally, they concluded that the SVM Evaluator in combination with Expectation Maximization (EM) classifier and the IG evaluator in combination with Random Tree Classifier have achieved the highest accuracy. It had also been found that the small simple sizes limit the ability to draw firm conclusions.

Dabek, Filip, and Jesus J. Caban [19] developed a Neural Network (NN) Model with an accuracy of 82.35% for predicting the likelihood of developing psychological conditions such as anxiety, behavioural disorders, depression and post-traumatic stress disorders. In [20], Tawseef Ayoub Shaikh, compared the performance of Artificial Neural Networks, Decision Tree and Naïve Bayes in predicting Parkinson's disease and Primary Tumour Disease and found that the accuracy is high in ANN for predicting Parkinson's disease and Naïve Bayes for Primary Tumour Disease. Accuracy of Decision Tree and Naïve Bayes have further improved after reducing the size of feature set by applying Genetic Algorithm to the actual data set.

The literature review shows that, on one side, a number of research works are going on in computerizing the diagnosis of mental health disorders. On the other side, efforts are taken to diagnose the mental health problems using different machine learning techniques in an efficient way. Various combinations of machine learning techniques (Hybrid techniques) are being employed to improve the accuracy of diagnosis with reduced set of features from profiles of patients. This research is to analyse selected machine learning techniques in predicting the possibility of primary mental health problems like Attention Problems, Anxiety Problems, Academic Problems, ADHD and PDD. Early diagnosis of these types of problems among children allows early treatment and improves their quality of life. Comparison has also been made with reduced number of attributes.

III. OVERVIEW OF MACHINE LEARNING TECHNIQUES

As of now, various machine learning techniques are available and still researches are going on to techniques that produce optimal results. According to Wolpert et al. [6], there is no single learning algorithm that universally performs best across all domains. Hence, in this research, eight techniques are selected as they produced accurate results for a small dataset. The techniques selected are AODEsr, Multi Layer Perceptron (MLP), RBF Network, IB1, KStar, Multi-Class Classifier (MCC), FT, LADTree.

A. AODEsr

Averaged One-Dependence Estimator (AODE) is a probabilistic classification learning technique. It is advantageous over Naïve Bayes Classifier as it addresses the problem of attribute-independence at the cost of increased amount of computation. It supports incremental learning and handles situations where some data are missing.

B. Multilayer Perceptron

Multilayer Perceptron is a feed forward artificial neural network model that maps set of input data onto a set of appropriate outputs. It is a supervised learning technique that uses back propagation for training the network and it is used to distinguish data that are not linearly separable.

C. RBF Network

Radial Basis Function Network is a feed forward artificial neural network that uses radial basis functions as activation functions for classification. It is a supervised learning technique that trains much faster than back propagation networks.

D. IB1

IB1, an instance based classifier uses a simple distance measure to find the training instance closest to the given test instance and predicts the same class as this training instance. The training is very fast and suits to problems in which the target function is very complex, but can still be described by a collection of less complex local approximations.

E. K* or KStar

K*, an instance based classifier, is similar to K-Nearest Neighbor. It uses entropy as a distance measure for

classification and handles real valued attributes and missing values.

F. Multiclass classifier

This algorithm classifies instances into one of the more than two classes i.e. each training instance belongs to one of N different classes.

G. FT

Functional Trees are classification trees that could have logistic regression functions at the inner nodes and/or leaves. It deals with binary and multi-class target variables, numeric and nominal attributes and missing values.

H. LAD Tree

Logical Analysis of Data tree is the classifier that generates multiclass alternating decision tree using logistics strategy. It combines ideas and concepts from optimization, combinatorics and Boolean functions. The patterns or rules play a vital role in classification. It has been successfully applied to data analysis problems in different domains, including biology and medicine.

IV. METHODOLOGY

As the first step for the research, the problem of diagnosing basic mental health was identified and an Interview was held with a clinical psychologist to identify the mental health problems that occur more often among children. Then, observation was made on how the diagnoses were performed by the professionals. A model was built that uses machine learning techniques to diagnose five common mental health problems effectively. This model assists the professionals to identify the problem if the known evidences of the patient are given as input.

Figure 1 shows the methodology of the research. The data set for predicting mental health problem is taken from a clinical psychologist. The data set has 60 instances in text document format. From the documents, 25 attributes including the class label have been identified manually and checked with the psychologist.

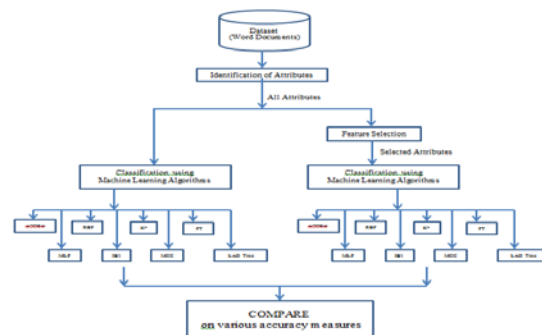


Fig. 1. Methodology of the Research

The attributes identified from the data set are listed in table 1. All the attributes are of nominal type. As only a few attributes are relevant for classification and prediction of the problem, data set is pre-processed by eliminating irrelevant and redundant attributes using Best First Search technique.

TABLE I. ATTRIBUTES IDENTIFIED FROM THE DATA SET. ATTRIBUTES SELECTED WITH FEATURE SELECTION ALGORITHMS ARE SHOWN IN BOLD

No.	Attribute	Meaning	Values	No.	Attribute	Meaning	Values
1	Age	Age group of the child, Infant / Early Childhood / Middle Childhood / Adolescent	{I,E,M,A}	14	Attention Sustained	Whether the child is able to direct and focus cognitive activity on specific stimuli	{Y,N}
2	Family History	Presence / Absence of psychological disorder in the family	{Y,N}	15	CBCL Score	A checklist filled by teacher/parent/self-report to identify problem behavior in children	{BC, EC, AC}
3	Pregnancy Complication	Presence / Absence of Complication during pregnancy	{Y,N}	16	IQ Test Score	A standardized test to assess the intelligence level of the child. The level may be Below Average, Average, Above Average	{BA,A,AA}
4	Delayed Speech	Presence / Absence of delay in development of speech	{Y,N}	17	ADHD Positive	Screening test score for Attention-Deficit/Hyperactivity Disorder	{Y,N}
5	Under Medication	Whether the child is under any type of medicinal control	{Y,N}	18	ODD Positive	Screening test score for Oppositional Defiant Disorder to measure argumentative and defiant behavior of the child	{Y,N}
6	Academic Performance	Whether the performance of the child is adequate/ inadequate in academics	{A,I}	19	Manic Episode Test Score	Screening test score for manic episode to check extreme elation or irritability in the mood of the child	{Y,N}
7	Relationship Formation	Whether the child has adequate ability to socialize with peers, relatives and teachers	{A,I}	20	General Anxiety Disorder	Screening test score to measure the anxious level of the child	{Y,N}
8	Behavioral Problem	Whether the child has any problem in behavior, Ex. Cheating, telling lies	{Y,N}	21	Major Depressive Episode	Screening test score to measure the major depression level of the child	{Y,N}
9	Concentration	Whether the child has adequate ability to focus his attention on a particular object or activity	{A,I}	22	CDI Score	Screening test score to measure the depressive symptoms of the child	{AC,EC,BC}
10	Restless	Whether the child is able to relax or involving in constant activity	{Y,N}	23	PDD Score	Screening test score to measure the level of pervasive developmental disorder of the child	{MODERATE,NO,MILD}
11	Seizures	Whether the child has a sudden surge of electrical activity in the brain which may cause unconsciousness, rigidity, muscle stiffness, uncontrollable movements, etc.	{Y,N}	24	Autism Score	Screening test score to identify whether the child has difficulty in communicating and forming relationships with other people and in using language and abstract concepts	{MODERATE,NO,MILD}
12	Learning Difficulty	Whether the child has any difficulty in acquiring age appropriate knowledge and skills	{Y,N}	25	Problem	The class attribute specifying the mental health problem of the child.	{ATT_ACA, ANX_SYMP, ANX_DEP_SYMP, PDD, DEV_DELAY, AUT_SYMP, ACA, ATT_BEH, ATT_EMO}
13	Attention Aroused	Whether the child is psychologically alert, awake and attentive	{Y,N}				

V. EXPERIMENTAL RESULTS

In this section, the performance analysis of eight classification algorithms is carried out with a common dataset using WEKA tool. First, the classifiers were executed by including all the attributes (25) identified from the text documents and then they were executed by including only the attributes (13) selected by the feature selection algorithms. The WEKA tool provides various measures to understand the classification. Among the various measures, three measures are very important to compare the accuracy level of classifiers. They are Kappa Statistic, Accuracy and ROC Area.

Kappa Statistic measures the agreement of prediction with the true class. A measure of 1 signifies complete agreement and a measure of 0 signifies complete disagreement between prediction and the true class. Figure 2 depicts that the kappa statistics of three classifiers, namely, Multilayer Perceptron, Multilayer Classifier and LAD Tree are higher than other classifiers and selected attributes show a higher kappa value than classifier with all attributes.

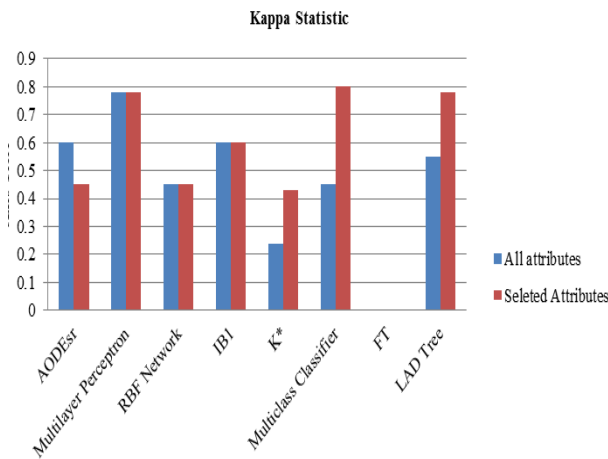


Fig. 2. Kappa Statistic for all classifiers

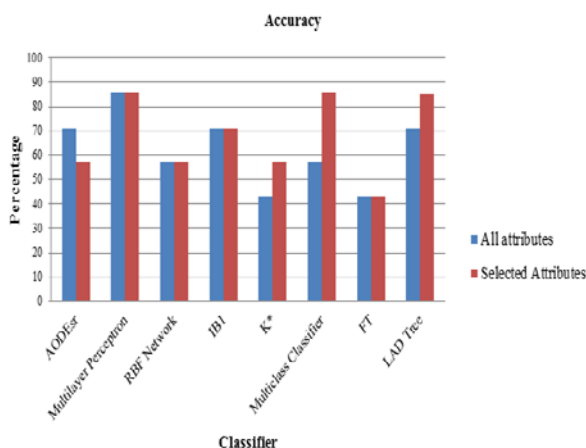


Fig. 3. Accuracy measures for all classifiers

Accuracy of a classifier on a given test set is the percentage of test set instances that are correctly classified by the classifier. Figure 3 depicts that the three classifiers, namely, Multilayer Perceptron, Multilayer Classifier and LAD Tree are

more accurate than other classifiers and the selected attributes show a higher accuracy value than classifier with all attributes.

The accuracy of the classifier depends on how well the classifier classifies the data set being tested. It is measured by the area under the Receiver Operating Curve or ROC Area. An area of 1 represents a perfect test and an area of .5 represents a worthless test. The Figure 4 shows the graph on ROC Area values of the eight classifiers. The ROC Area of four classifiers, namely, Multilayer Perceptron, K*, Multiclass Classifier and LAD Tree are between 0.8 and 0.9 and so these classifiers are more accurate than other classifiers in predicting the basic mental health problems among children.

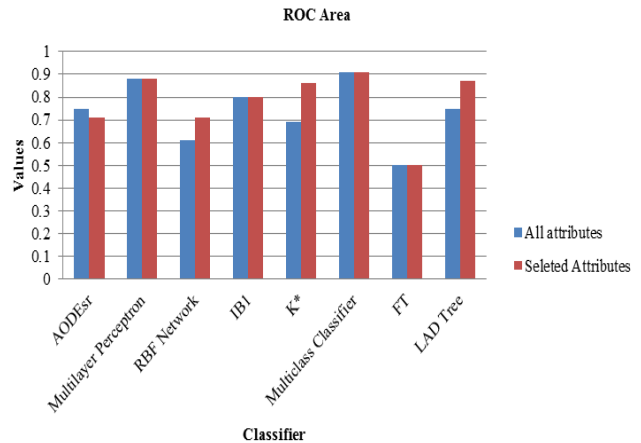


Fig. 4. ROC Area values for all classifiers

VI. CONCLUSION

Nowadays, a number of expert systems are utilized in medical domain to predict diseases accurately at an early stage so that treatment can be made effectively and efficiently. Also, expert systems are developed in the mental health domain to predict the mental health problem at an earlier stage. As a number of machine learning techniques are available to construct expert systems, it is necessary to compare them and identify the best that suits the domain of interest. The research has compared eight machine learning techniques (classifiers) on classifying the dataset to different mental health problems. It is evident from the results that the three classifiers viz., Multilayer Perceptron, Multiclass Classifier and LAD Tree produce more accurate results than the others. The data set is very minimal and in future, the research may be applied for a large data set to obtain more accuracy. The classifiers need to be trained prior to the implementation of any technique in real prediction.

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VLSI Design of a High Performance Decimation Filter Used for Digital Filtering

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Abstract—With the rapid development of computers and communications, more and more chips are required to have small size, low-power and high performance. Digital filter is one of the basic building blocks used for implementation in Very Large Scale Integration (VLSI) of mixed-signal circuits. This paper presents a design of decimation filter used for digital filtering. It consists of Cascode Integrated Comb (CIC) filters, using Finite Impulse Response (FIR) filters and Infinite Impulse Response (IIR) filters structure. This architecture provides small area and low power consumption by avoiding the use of multiplication structure. This design presents the way of speeding up the route from the theoretical design with Simulink/Matlab, via behavioral simulation in fixed-point arithmetic to the implementation on either ASIC. This has been achieved by porting the netlist of the Simulink system description into the Very high speed integrated circuit Hardware Description Language (VHDL). At the first instance, the Simulink-to-VHDL converter has been designed to use structural VHDL code to describe system interconnections, allowing simple behavioral descriptions for basic blocks. A comparison of several architectures of this circuit based on different architectures of most popular filter is presented. The comparison includes: supply voltage, power consumption, area and technology. This approach consumes only 2.94 mW of power at a supply voltage of 3V. The core chip size of the filter block without bonding pads is 0.058 mm² by using the AMS 0.35 μ m CMOS technology.

Keywords—Digital circuit design; CIC decimation; Cascaded integrator comb filter (CIC); IIR-FIR structure

I. INTRODUCTION

Generally analog filters are cheaper, faster and have large dynamic range in both amplitude and frequency. Digital filters in comparison are vastly superior performance level that can be achieved. Quality is better than analog filters to digital filters can achieve performance unique [1]. The filtering problem is approached makes a dramatic difference [2]. With analog filters, emphasizing precision and stability, such as resistors

and capacitors in electronics, controls have limitations. In comparison, digital filters are often ignored in order to better filter performance [3]. The emphasis shifts signal constraints and the processing of theoretical issues. The digital processing performance of signals and communication systems is generally limited by the quality of the input signal. The increasing use in telecommunication digital technique, like audio applications, supported the analog to digital converter use [4]. However, it proved that the traditional converters do not reach high performances on a small surface. Current research in the field of telecommunication progress exceeds the current technological limits and improves the concept of the future system performances considerably: better integration rate, low power, personal communication apparatus size reduced and the capacity of multiplying the communication number. For the signal transmission, a using of a conversion technique called modulation such as the coded pulse modulation. The basic operation in digital signal processing is filtering [5]. This operation is widely used in many electronic devices to cancel part of signal that is redundant or damages the signal. The digital filter is described by difference equation in time domain and by transfer function in frequency domain [6]. There are two basic types of digital filters: Finite impulse response (FIR) filters and infinite impulse response (IIR) filters [7]. The both types of filters have some advantages and disadvantages are summarized in Table I.

The main disadvantage of a FIR filter is high order of the filter in comparison with IIR filter with approximately the same frequency response. So the size, consumption of power and computing time of a FIR filter are higher than of IIR filter. The one of the solutions of this problem is use Interpolated FIR (IFIR) filter that significantly reduces order of the filter.

Structure of IFIR filter is described below. More details about features of FIR and IIR filters can be found in. Different structures of Digital filters are presented such as: decimation filter, comb filter, half band filter.

TABLE I. ADVANTAGES AND DISADVANTAGES OF DIGITAL FILTERS

Type of Filter	Advantages	Disadvantages
FIR	-Linear phase -Behavioral Stability -Low quantization noise -Simple implementation	High order filter
IIR	Low order filter	Stability Complicated implementation Limit cycles

The paper is organized as follows. Firstly, a presentation of decimation filter by combining FIR and IIR filters. Secondly, a description of Cascaded Integrator Comb (CIC) Filter and Half band filter are presented with behavior simulation using MATLAB. Then a presentation of synthesis process of decimator filter which shows the standard cells netlist extracted from Verilog file in order to obtain an integrated circuit design called the layout of decimation filter using Cadence. Finally, all main parameters of the described decimation filter are indicated with a full comparison of the most popular designs.

II. DECIMATION FILTER

One of the most important component which combines FIR and IIR filters is decimation filter. Its function is to remove all of the out-of-band signals and noise and to reduce the sampling rate by M, where M is the over-sampling ratio (OSR). By averaging M values of the coarsely quantized sigma-delta output, the filter gives a high resolution output at the low rate.

In the decimation [8], the frequency on the outlet side of the sample frequency F_s filter is a submultiples frequency of entry F_e . There is $F_s/F_e = 1/M$. It is then enough to filter the signal to keep only the contents of the band $[0, F_s/2]$ corresponding to the new sampling rate, then not to keep that a sample on M. This filtering must be with linear phase so that a signal originally in the band $[0, F_s/2]$ is not deteriorated by filtering. The filtering multi-stages reduce the complexity of the decimation filter of the converter. The first step in designing a decimation filter is to decide which types of filters will be used and where decimation will occur. It is possible to remove the undesired channels and noise with a single filter and then decimate to the output rate. The number of taps in an FIR filter is proportional to the stop band rejection and to the sampling rate divided by the transition band. The power is proportional to the number of taps and the rate at which they operate. By decimating in stages, the total number of taps in the filters is reduced and subsequent filters operate at lower sampling rates, further reducing the power consumption [8].

III. CASCADED INTEGRATED COMB (CIC) FILTER

Different structures of decimation filter are used such as: polyphase, IIR-FIR and Non-recursive structure. A polyphase structure [9], needs maximum area and minimum power consumption. Then IIR-FIR structure requires highest power consumption and minimum area on chip. In this paper we are using a hardware implementation of a multistage decimation filter using Cascaded Integrator Comb filter (CIC) shown in Fig. 1 required IIR-FIR structure. Where IIR filter works at a sampling frequency F_s and the FIR filter works at Nyquist rate F_s/M .

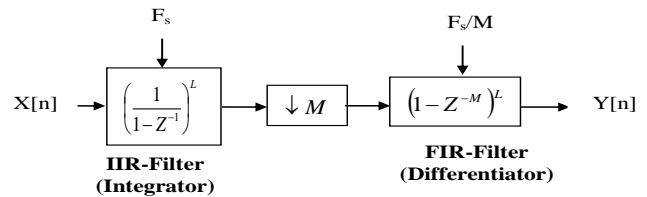


Fig. 1. Block Diagram of CIC decimation using IIR-FIR Filter Structure

Where M is the Over sampling Ratio. Moreover IIR filter can be implemented as digital integrator and FIR filter can be implemented as digital differentiator. The transfer function of the CIC filter is giving by:

$$H(z) = \left(\frac{1 - Z^{-M}}{1 - Z^{-1}} \right)^L \quad (eq 1)$$

Where M is the oversampling ratio, L is the order of the filter. The numerator $(1 - Z^{-M})^L$ represents the transfer function of a differentiator and the denominator $1 / (1 - Z^{-1})^L$ indicates the transfer function of an integrator. The decimation filter is made up of several stages.

A. Half band Filter Design

A Half-Band filter [10] is a special type of FIR which is very suitable for decimation by 2. It consists of two FIR filters. The decimation factor for each FIR filter is two. The first FIR filter in a chain corrects the distortion implied by the first two Sinc blocks. This is also a low-pass filter and though it has to provide a noise shaping outside the filter bandwidth. The last FIR filter is used as an element with a very high selectivity. FIR filters design is based on a CSD (Canonical Signed Digit) representation and they have a hardware implementation of their coefficients. For the filter coefficients design, there are number of advantages provided by CSD code over the ordinary binary representation.

First, since there are no adjacent nonzero digits in CSD code, one may expect less nonzero digits in CSD code than in the binary equivalent, which simplifies filter implementation. For the similar reason, CSD code is expected to be less sensitive to truncations which can further help in reducing add-shift operations within the multipliers while preserving initial filter properties.

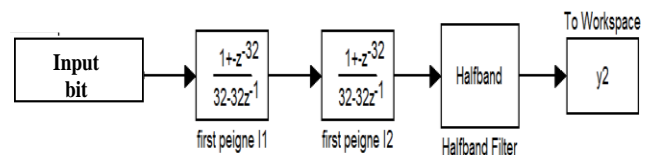


Fig. 2. MATLAB model of the Halfband Filter

CIC filter as the first stage and half-band filter as the second stage used to down sample the sampling frequency to Nyquist rate, as shown in Fig 2.

B. Design and simulations methods

a) Simulations results of the filter:

The objective is to design a decimation filter with several stages where the intermediate stages have more flexible specifications. The first stage is divided into two blocks with

decimations by 8 then by 4 to decrease the sinc filter order. The first filter is a fourth order whereas the second filter is a third order. This division is made to reduce the filter Orders and consequently their complexity. The last half-band filter has strict specifications, while the first half-band filter presents broad transition band. After simulation using MATLAB tool [11], the following curves is obtained: Fig. 3 shows the frequency response of the cascaded sinc filter.

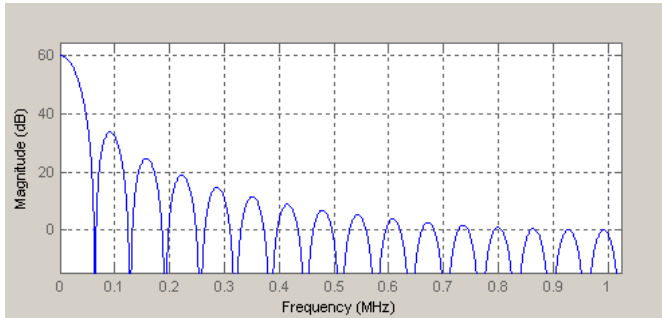


Fig. 3. Sinc filter response

Fig. 4 is the superposition of the sinc filter loss in amplitude curve and the ideal response of FIR filter.

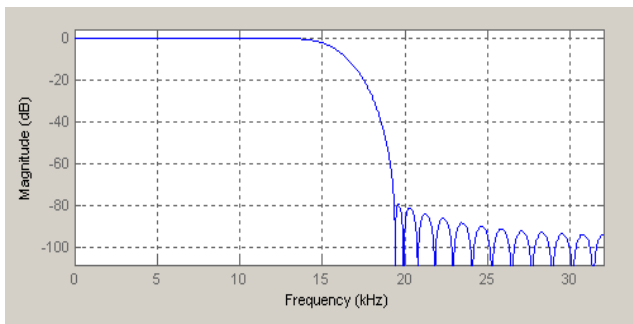


Fig. 4. Frequency response normalized and Composed after HB1

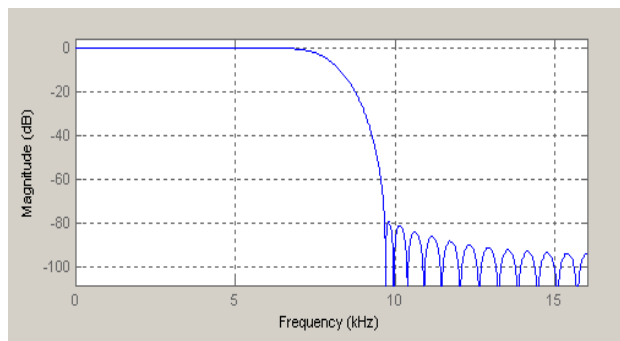


Fig. 5. Frequency response normalized and Composed after HB2

Finally, Fig. 5 presents the frequency response normalized and composed after HB2. It is the total output of the decimation chain

b) Synthesis of filter :

As shown in Fig. 6, during synthesis process, standard cells netlist was extracted and in a form of Verilog file loaded back to VHDL simulator [12]. The simulation was now performed using Cadence NCsim tool, which is a tool for logical

verification, using the same test bench as before synthesis process [13]. Results of this simulation are digital samples from the filters' outputs. They are again loaded back to Matlab and processed with FFT functions. Obtained frequency this way, a design verification process was completed successfully, since the requirements are met. Using Cadence program Silicon Ensemble, floor planning, placement and routing were performed, as well as clock and reset trees generation. At the end, Verilog file is extracted and brought back to NCsim simulator where final check of the total digital part of the chip was performed.

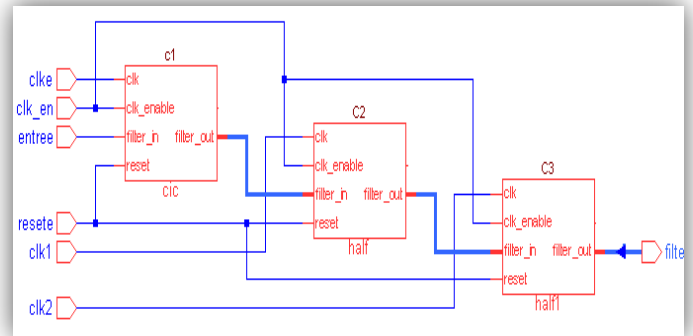


Fig. 6. Synthesis of filter

c) Layout of filter:

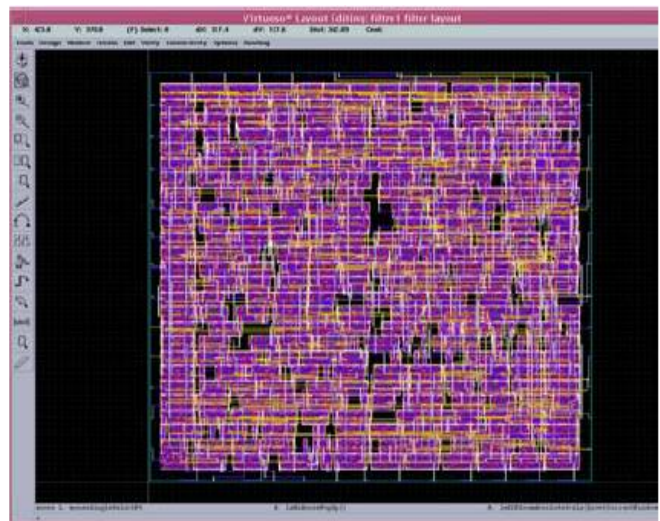


Fig. 7. Layout of decimator filter

After the step of synthesis process by extracting the cells netlist from verilog file, it is necessary to obtain an integrated circuit design. So physical design is a step in the standard design cycle which follows after the circuit design. At this step, all devices and interconnects of the design are converted into geometric representation of shapes which called integrated circuit layout, as shown in Fig 7 the layout of filter decimation obtained.

In Table II, a comparison of the most popular designs which also compares the published works with the current work. It can be seen that the current work consumes less power (2,94 mW) than most published work and also have a small

area (0,058 mm²). In the other hand, it achieves a higher speed of 10.24 MHz.

TABLE II. COMPARISON TABLE OF MOST POPULAR FILTER BLOCKS WITH CURRENT WORK (*)

	[14]	[15]	This Work*
Supply Voltage Vdd (V)	1.8	1.8	3
Power consumption (mW)	1.73	8.1	2.94
Area (mm ²)	-	0.2	0.058
Sampling Frequency (Mhz)	2.56	1	10.24
Technology	CMOS 0.18µm	CMOS 0.18µm	AMS 0.35µm

IV. CONCLUSION

Design of decimation chain made up of digital filters was successful. The multi-stages filtering reduces the complexity of the converter decimation filter. It was verified in the standard verification environment of the Cadence laboratory and is ready to use. The design is suitable for applications that require low power consumption and a small occupied silicon space. The first stage is a sinc filter carrying out decimation by 8. The second stage is also a sinc filter carrying out decimation by 4. The last two stages are half-band filters carrying out each one decimator by two. The main advantage of the design is that they can be implemented in which ever FPGA; meaning that they are not dependent on the platform. Using AMS 0.35µm CMOS Technology a Cascode Integrated Comb (CIC) filters structure consumes only 2.94 mW of power at a supply voltage of 3V. In the future work, a design of the different blocks constituting complete analogue to digital converter (ADC) by adding a block of modulator to the decimator filter will have been studying, in the other hand a Discussion about multi-bit discrete-time Sigma-Delta ADC and methodology will be described.

In the future work, we will study and design the fabrication of the proposed filter and the measurement results [16].

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Proposal and Implementation of MPLS Fuzzy Traffic Monitor

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Abstract—Multiprotocol Label Switched Networks need highly intelligent controls to manage high volume traffic due to issues of traffic congestion and best path selection. The work demonstrated in this paper shows results from simulations for building optimal fuzzy based algorithm for traffic splitting and congestion avoidance. The design and implementation of Fuzzy based software defined networking is illustrated by introducing the Fuzzy Traffic Monitor in an ingress node. Finally, it displays improvements in the terms of mean delay (42.0%) and mean loss rate (2.4%) for Video Traffic. Then, the result shows an improvement in the terms of mean delay (5.4%) and mean loss rate (3.4%) for Data Traffic and an improvement in the terms of mean delay(44.9%) and mean loss rate(4.1%) for Voice Traffic as compared to default MPLS implementation.

Keywords—Multiprotocol Label Switched Networks; Fuzzy Traffic Monitor; Network Simulator; Ingress; Traffic Splitting; Fuzzy Logic Control System; Label setup System; Traffic Splitting System

I. INTRODUCTION

Multiprotocol Label Switching Network (MPLS) is one of the promising architecture of the future internet. It has proven to be highly reliable. MPLS provides traffic engineering and virtual private network (VPN) services that lead to solving problems relating to traffic congestion. Currently, these network services depend upon protocols like Resource Reservation Protocol (RSVP), Constraint-based Label Distribution Protocol (CR-LDP), Open Shortest Path First (OSPF) and the extensions for the resource reservation protocol (ERSVP) [1-4].

However, on the same note, all routers must enable new protocols to support any new network services. Extensive simulation using NS-3 and contemporary research for optimizing the high speed network. The tests were conducted to reach optimal conditions for large volume traffic network scenarios. Since, these scenarios are known to be computationally and resource intensive, there is a continuous need for innovation achieving faster response time in managing traffic splitting and congestion avoidance.

However, the main stress is on traffic splitting, which is mainly done dynamic as per traffic conditions and available paths. Multiprotocol architecture provides various features such as Quality of Service (QoS), Architectural Scalability, Differentiated Services (DiffServ), Integrated Services

(IntServ), Explicit Routing and high quality End-to-End (E2E) service for Virtual Private Networks (VPNs).

In this research work, it is illustrated with the architecture and study of improving the performance of transmission through ingress node [7,8]. Ensuring 100% E2E QoS that makes the network to select the most feasible path in terms of load, delay, utilization rate and link capacity.

The framework proposed till now is lacking in high dynamic traffic scenario. So, there is a need to overcome the main limitations discovered by researchers [9-19]. A deep investigation for analysis of high volume traffic and improving existing algorithms on traffic splitting and congestion avoidance is done.

Constraint based routing or multiple path selection can fail due to no decision making component in software defined networks. It gives high node to node QoS for video rendering applications. The need of the path selection process considers the combination of QoS and Traffic Engineering (TE) constraints dynamically. Therefore, this paper presents experience and fuzzy-based rules for traffic management in Software Defined Networks. This architecture ensures high QoS for video applications.

A virtual environment was created to perform simulations for understanding the bottlenecks in traffic management and building a solution based on this analysis. The framework consists of NS-3 and Fuzzylite.

It presents the integration of the Fuzzy Logic Control System with NS-3 to simulate and model assumed network model. The MPLS module was extended to achieve highly optimized traffic routing. The parameters are defined for standard simulations performed to analyze the MPLS network formation. The Fuzzy logic Control System (FLCS) is sited at ingress node. FLCS is composed of two sub fuzzy control systems ie. Label Switched Path setup System (LsS) and Traffic Splitting System (TSS) as shown in the Fig.1.

LsS selects LSPs according to two input fuzzy metrics (delay and load) [16, 17] and gives output fuzzy metric LsS_value. Load and Delay are suitable QoS factors on which selection of appropriate path depends. TSS performs the traffic splitting among the computed paths, according to the two input fuzzy metrics (utilization rate and link capacity) and gives output fuzzy metric TSS_value.

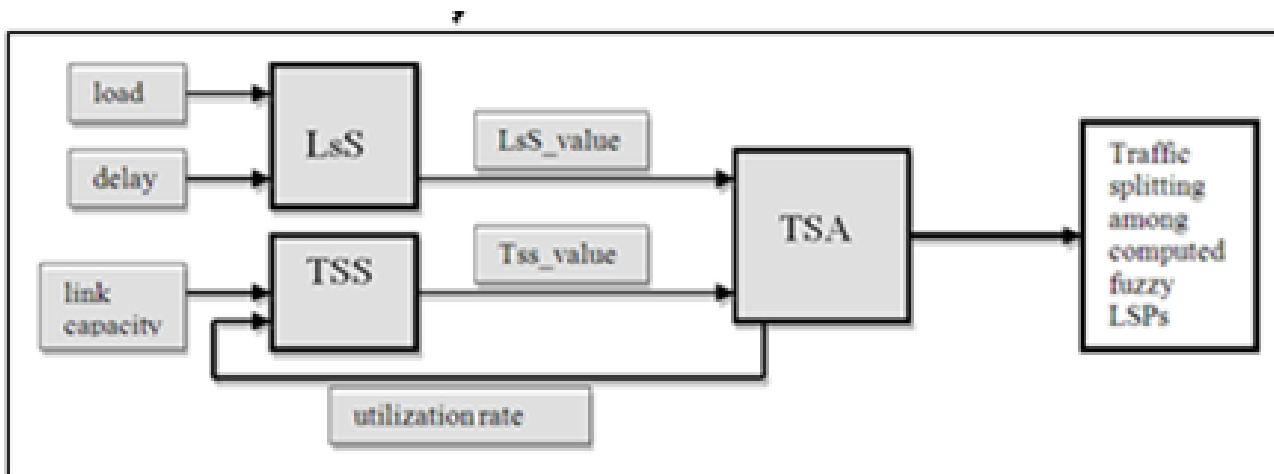


Fig. 1. The Fuzzy Logic Control System consists of LsS and TSS as fuzzy controllers

An appropriate decision of traffic splitting is performed among the computed number of LSPs required by Traffic Splitting Algorithm (TSA). The computed fuzzy based LSPs for forwarding packets are obtained to avoid the situation of underutilization and over utilization of paths. None of the paths remains idle for longer time and proper utilization of resources takes place. Hence, it is better for congestion to be prevented rather than corrected. Implementation of FL using Mamdani Fuzzy Inference System [31] evaluates final decisions as LsS_value and TSS_value. The available number of rules in the Rule Base matrix LsS and TSS represents intermediate situations and provides the control mechanism with a highly dynamic action.

The section II discusses the current scenario of the problems related to traffic splitting and motivation for design of an intelligent system in MPLS networks. We describe the model of our proposed approach in detail in the section III. Section IV, studies the performance evaluation, and finally section V concludes the research paper. We strongly believe that our proposal will considerably helps in solving the congestion problem in MPLS networks. Since, the measurements of the input parameters (ie. Load, delay, link capacity and utilization rate) can be imprecise due to dynamic nature of traffic. In fact, the proposed scheme provides a way forward to a more robust and efficient form of multimedia streams, which is highly needed as the demand grows for better services.

II. THE CURRENT SCENARIO

Currently, network QoS attributes are most important for real-time applications. These are not addressed in the path selection process.

As, MPLS is not able to provide intelligent routing, which is a fundamental component of high volume multimedia traffic [4]. The implementation of RSVP with MPLS successfully enables the reservation of bandwidth on a router for each LSP. However, it does nothing to solve the variable demands of multimedia services. With these solutions, MPLS is not able to provide 100 % guarantee on the QoS with respect to high volume incoming traffic. Choosing the best suitable routing algorithm for network is the key factor determining the

network lifetime [5]. Moreover, heterogeneity of the nodes in the network also plays a vital role in the network operations [6]. Hence, there is a need to look for a better solution.

A. Technology background

OSPF is opted for Multiprotocol Label Switching with traffic engineering (MPLS-TE). It is introduced in RFC3630 [7] which work using the OSPF opaque Link State Advertisement (LSA). It builds an extended TE link state database (TEDB), for signaling Extended Resource Reservation Protocol-Traffic Engineering (RSVP-TE). It supports intra domain, hard state connection oriented – set up / take down, provisioning, recovery and many other features. The LSA spread the attributes of TE link among the network. RFC4090[8] performed work regarding routing protocols and redesign to give MPLS Fast-Reroute (MPLS-FRR) which is still not a reliable solution for such a large scale data center networks.

From the mentions of the literature survey [9-15], the analysis of related work shows that although the applicability of this fuzzy approach needs to be put to extensive tests. We can further conclude that it has gained much attention recently that till date OSPF has not been replaced with any better algorithm which can produce better results.

III. MODEL FORMULATIONS

After conducting systematic literature survey and studying associated matter related to the problem area. In this work, we propose a fuzzy based solution that does not change the existing conventional algorithms but uses operations of them in order to provide a system that will improve performance. The selection of the fuzzy logic method was based on the simplicity and the fact that since it processes experts defined rules governing the routing system. It can be modified & tweaked easily to improve the performance. Analysis of the state of the art in it reveals that there is no protocol designed and evaluated. Hence, in this paper, we use the abilities of FL to propose a technique for designing a fuzzy based traffic conditions. In a dynamic network environment, it is difficult to build choice in choosing a path where traffic congestion will be least possible. There are many fuzzy factors responsible to get the best one for a flow. The goal and the

constraints are fuzzy in nature. Path selection and traffic splitting among selected paths is a decision making problem in a fuzzy network environment. To deal quantitatively with an impression, we employed fuzzy theory to define fuzzy goals or fuzzy constraints precise as fuzzy sets and get fuzzy decisions. Therefore, Fuzzy Logic Control System (FLCS) is proposed to select paths and then traffic splitting among them. Since, there are many available paths for the same source node and destination nodes. So, this approach first gives the list of paths in preference order. Secondly, the traffic will be transmitted among the number of paths required, which is obtained from the controller itself.

IV. ARCHITECTURE

A. Background and Concepts

Fuzzy Traffic Manager (FTM) is a novel an intelligent controller resides in an ingress node of MPLS domain as shown in Fig. 2. It works in the LSP selection component. The Label Distribution Protocol (LDP) takes the output of the proposed controller and transmits it to routing protocol (RSVP-TE, OSPF).

The architecture has the following features:-

1) High speed network

Domain consists of two kinds of nodes edge nodes and core nodes. An edge node, is called as label edge router. The ingress label edge router sends packets to the domain and egress label edge router receives packets from the domain. The intermediate nodes are core nodes which perform packet forwarding.

2) Fuzzy Logic Control System

The proposed fuzzy traffic manager resides in the ingress node works on the execution of Fuzzy Logic Control System (FLCS). The Fig. 1 is holistic view of work done, as seen FLCS consists of two fuzzy controllers LsS and TSS. LsS develops intelligent rules using QoS constraints of traffic flow. It gives the SLSPs as the matrix of LSPs in preference order to be used for routing. TSS develops intelligent rules using TE constraints of traffic flow. It computes the NLSPs as the matrix of number of required LSPs. [16] defines completely mathematically how fuzzy logic will be applied to proposed controllers.

LsS rule base is a set of inputs (delay (D), load(L)) with their five linguistic values (Very Low (VL), Low (L), Medium (M), High (H), Very High (VH)) and output LsS_value with their seven linguistic values (Zero (Z), Tiny (T), Very Small (VS), Small (S), Big (B), Very Big (VB), High (H)) forms at most 5*5 = 25 possible combinations. Hence, optimized decisions are made using developed 25 rules (R1, R2, and R3...R25) as shown in Fig. 3, which are maximum possible cases for the selection of LSPs under different situations.

This FMM (LsS-Fuzzy Rule Base Matrix) provides large number of options to obtain best routing decisions.

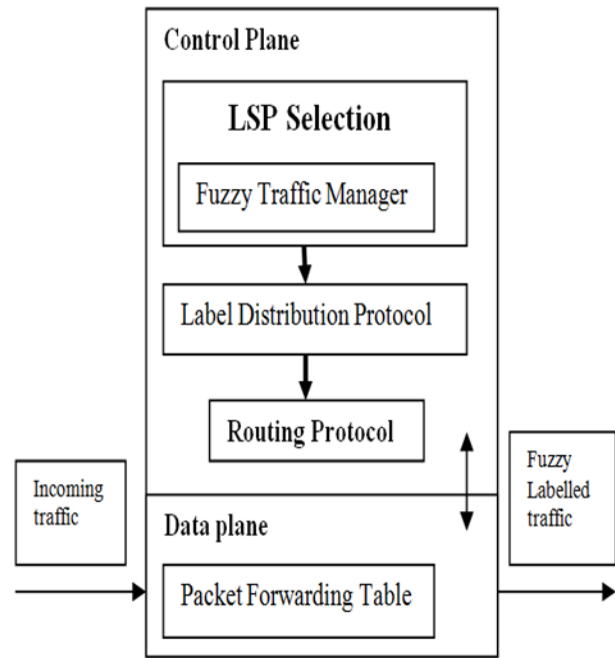


Fig. 2. The Fuzzy Traffic Manager (FTM) in the LSP selection component of an ingress node as FLCS

delay load	VL	L	M	H	VH
VL	Z R1	T R2	VS R3	S R4	B R5
L	T R6	VS R7	S R8	B R9	B R10
M	T R11	VS R12	S R13	B R14	B R15
H	VS R16	S R17	B R18	VB R19	VB R20
VH	S R21	B R22	B R23	VB R24	H R25

Fig. 3. Fuzzy Rule Base Matrix- LsS

It provides efficient alternate paths in preference order that support Traffic Engineering and Path Optimization. The combination of rules is shown in TABLE I. The computed output response LsS_value identifies the paths.

It demonstrates the classification of LSPs and also defines the fuzzy set boundaries. Paths lying in region Z, T, VS, S, B, VB, H are best, very good, good, satisfactory, just acceptable, not acceptable and rejected respectively. Hence, in this way it arranges the paths in preferential order for the successful transmission of packets.

TABLE I. COMPOSITION OF RULES

Output linguistic Variables	
LsS	Z=R1
	T=R2+R6+R11
	S=R3+R7+R12+R16
	VS=R4+R8+R13+R17+R21
	B=R5+R9+R10+R14+R15+R18+R22+R23
	VB=R19+R20+R24
TSS	H=R25
	NC=R1+R2+R3+R4+R8+R9+R10+R15+R16+R22
	IO=R5+R6+R11+R12+R17+R18+R23
	IW=R7+R13+R19+R24+R25+R29+R30+R20
	IT=R14+R21+R26+R27+R31+R32
A=R28+R33+R34+R35	

TSS rule base is a set of inputs (utilization rate (UR), link capacity (LC)), with their seven linguistic values of UR (Zero (Z), Tiny (T), Very Small (VS), Small (S), Big (B), Very Big (VB), High (H)) and five linguistic values of LC (Very Low (VL), Low (L) Medium (M), High (H), Very High (VH)) and output TSS_value with their five linguistic values (Very Low (VL), Low (L) Medium (M), High (H), Very High (VH)) forms at most $7*5 = 35$ possible combinations. Hence, optimized decisions are made using developed rules 35 (R1, R2, R3... R35) as shown in Fig. 4.

For the simplicity and low computational complexity of operation, Triangular Membership Function (TMF) is applied to obtain the membership function of metrics as:-

$$\text{triangle}(x;a,b,c)=\max(\min(x-a/b-a,c-x/c-b),0), \quad (1)$$

It is defined by three parameters, where b indicates the point on which, the membership function value is 1, a and c indicate the left and right limits of the definition domain of the membership function.

On applying TMF as shown in formula (1), we obtain membership functions of input and output parameters as follows:-

$$\mu_{\text{load}} = \{\mu_{\text{VL}}, \mu_{\text{L}}, \mu_{\text{M}}, \mu_{\text{H}}, \mu_{\text{VH}}\},$$

$$\mu_{\text{delay}} = \{\mu_{\text{VL}}, \mu_{\text{L}}, \mu_{\text{M}}, \mu_{\text{H}}, \mu_{\text{VH}}\},$$

$$\mu_{\text{LsS_value}} = \{\mu_{\text{Z}}, \mu_{\text{T}}, \mu_{\text{VS}}, \mu_{\text{S}}, \mu_{\text{B}}, \mu_{\text{VB}}, \mu_{\text{H}}\},$$

$$\mu_{\text{utilization_rate}} = \{\mu_{\text{Z}}, \mu_{\text{T}}, \mu_{\text{VS}}, \mu_{\text{S}}, \mu_{\text{B}}, \mu_{\text{VB}}, \mu_{\text{H}}\},$$

$$\mu_{\text{link_capacity}} = \{\mu_{\text{VL}}, \mu_{\text{L}}, \mu_{\text{M}}, \mu_{\text{H}}, \mu_{\text{VH}}\},$$

$$\mu_{\text{TSS_value}} = \{\mu_{\text{NC}}, \mu_{\text{IO}}, \mu_{\text{IW}}, \mu_{\text{IT}}, \mu_{\text{A}}\}.$$

utilization_rate \ link_capacity	Z	T	VS	S	B	VB	H
	VL	NC R1	NC R2	NC R3	NC R4	IO R5	IO R6
L	NC R8	NC R9	NC R10	IO R11	IO R12	IW R13	IT R14
M	NC R15	NC R16	IO R17	IO R18	IW R19	IW R20	IT R21
H	NC R22	IO R23	IW R24	IW R25	IT R26	IT R27	A R28
VH	IW R29	IW R30	IT R31	IT R32	A R33	A R34	A R35

Fig. 4. Fuzzy Rule Base Matrix- TSS

TABLE II. DEFUZZIFICATION OF FUZZY CONTROLLERS

	Linguistic term	Linguistic center	
LsS	Z	Z _c	$\text{LsS_value} = \frac{(Z * Z_c + T * T_c + VS * VS_c + S * S_c + B * B_c + VB * VB_c + H * H_c)}{(Z+T+VS+S+B+VB+H)}$
	T	T _c	
	VS	VS _c	
	S	S _c	
	B	B _c	
	VB	VB _c	
	H	H _c	
TsS	NC	NC _c	$\text{TSS_value} = \frac{(NC * NC_c + IO * IO_c + IW * IW_c + IT * IT_c + A * A_c)}{(NC+IO+IW+IT+A)}$
	IO	IO _c	
	IW	IW _c	
	IT	IT _c	
	A	A _c	

TABLE V. ALGORITHM 3 EXECUTES TSS IN FLCS

<p>Input: Traffic Constraints values are to be fed in FLCS</p> <p>Precondition: LC[i][j] of links from LIB</p> <p style="text-align: center;">Main program :</p> <ol style="list-style-type: none">1. For i = 1 to n do2. For j = 1 to n do3. Read LC[i][j]4. EndFor5. EndFor6. For i = 1 to n do7. For j = 1 to n do8. Freq[i][j]= Lused[i][j]9. Freq[i][j]= Freq[i][j] + 110. UR[i][i]= Freq[i][j]/ Lused[i][j]11. EndFor12. EndFor13. TSS_value(i,j)= TSS(UR,LC)14. Return NLSPs15. EndFor

V. SIMULATION ANALYSIS AND RESULTS

For validating incremental progress related to solving the problem of traffic splitting, a comparison between classical approach (OSPF) and proposed work (FLCS) was conducted by multiple experiments using NS-3[18-26]. We have performed a group of simulations in the scenario composed of MPLS-TE together with constraint based routing. FTM performs the dynamic route selection. Taking this as scenario, our work demonstrates the improvements by adding FTM for LSP Selection. The topology used to evaluate the proposed work behavior consists of three different link capacities and propagation delay as follows in Fig 5:- a) 10 Mbps, 3ms (Red) b) 100 Mbps, 2 ms (Green) c) 1000 Mbps, 1ms(Black).

The incoming requests arriving follow an exponential distribution (λ) and the requested bandwidth is uniformly distributed between ranges [0~10] (Mbps), [10~100] (Mbps), [100~1000] (Mbps) to model Voice, Data and Video Traffic respectively [27, 28]. The holding time is randomly distributed with a mean of 300 sec (ON = 0.325 OFF = .64).

IPV4 network stack has been used and packets are generated using a hypothetical ON/OFF model using the Drop tail queue. Different traffic types are used in simulations like the FTP over the TCP to simulate data traffic and the constant bit rate (CBR) to simulate voice and video over the UDP. The packet size of the User Datagram Protocol (UDP) is 512 byte and 1024 byte for the Transport Control Protocol (TCP). Finally, the existing topology and an available resources database in ingress assumed perfectly updated. The statistical comparison of FLCS and OSPF is depicted in TABLE VI, which illustrates that standard deviation for different traffic is high in the case of OSPF is obtained with varying UDP traffic for 300 seconds.

In the Fig. 6 (a,b) the mean delay and the mean loss rate of video traffic is displayed, which is obtained with varying UDP traffic for 300 seconds. In the Fig. 6 (c,d) the mean delay and mean loss rate of data traffic is displayed, which is obtained with varying UDP traffic for 300 seconds. In the Fig. 6 (e, f) the mean delay and the mean loss rate of Voice traffic is displayed, which is obtained with varying UDP traffic for 300 seconds. Moreover, 42%, 15.4% and 44.4 % of Video packets, Data packets and Voice packets respectively are dropped due to buffer overflow at the congested node when using OSPF. FLCS on the other hand, as seen from Fig.6 by avoiding this congestion as mentioned earlier, maintains the traffic delivery performance.

By distributing the traffic over suitable preferable paths and maintaining low queue lengths, all the packets can be successfully delivered. The result is an increase in the queuing delay at a particular congested node, consequently leading congestion, and large waiting time at a few hops where the performance is poor. For few seconds in the above scenarios, OSPF outperforms due to low queuing delay. Even though FLCS utilizes multiple paths and it leads to less delay for the said Congested path. The obtained results of simulation experiments are displayed in as Fig. 6 and Table VI gives the better understanding of research work.

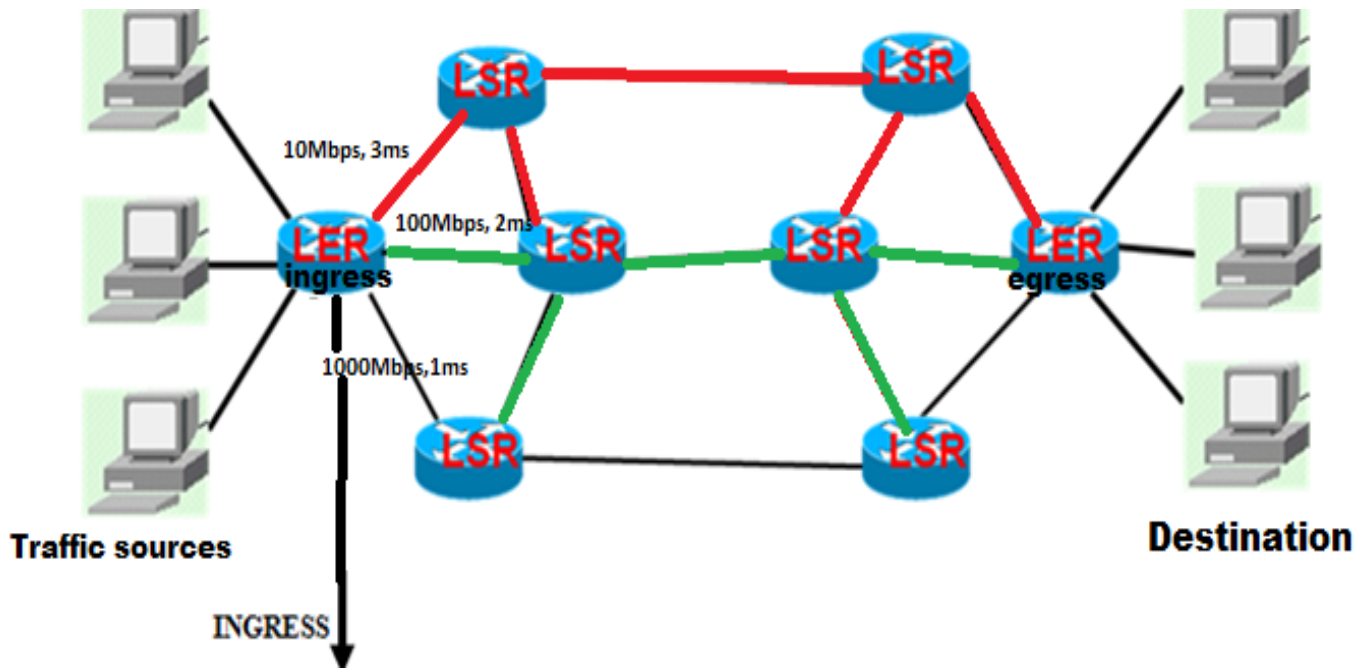


Fig. 5. Link capacities and propagation delays a) 10 Mbps, 3ms (Red) b)100 Mbps, 2 ms(Green) c)1000 Mbps, 1ms(Black)

TABLE VI. STATICAL STUDY OF THE RESULTS OBTAINED FOR OSPF (RED AND FLCS(BLUE))

Video Traffic			Data Traffic			Voice Traffic		
Mean Loss Rate			Mean Loss Rate			Mean Loss Rate		
MAX	33.6	29.4	MAX	0.16	8.604	MAX	19.1	14.9
MIN	0	0	MIN	0.0011	0.001	MIN	0	0
STDEV	9.66351192	8.984994	STDEV	2.958289	3.467	STDEV	4.503381	5.4168058
Mean Delay			Mean Delay			Mean Delay		
MAX	33.6	29.4	MAX	0	0.126	MAX	0.25	0.21
MIN	0	0	MIN	0.0011	0.001	MIN	0	0
STDEV	0.01939707	0.018885	STDEV	0.04777	3.399	STDEV	0.070779	5.8286941

VI. CONCLUSIONS

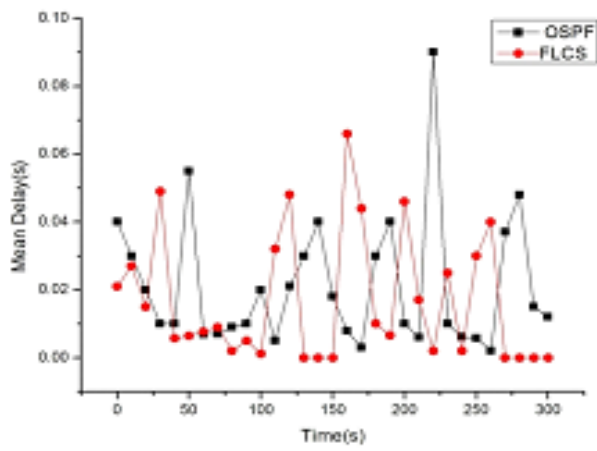
Our premise of this research is based on the fuzzy traffic manager which works within the realms of MPLS. The FTM is positioned at ingress node. The proposed fuzzy based approach is highly suitable for congestion control over MPLS networks. It applies QoS & TE constraints. It is designed for better video delivery, since it provides better throughput, average delay. We also empirically demonstrated the advantages of FLCS over OSPF in NS-3.

In current simulation, the network was using 2 edge routers and 6 switch routers. We achieved significant improvement as 42%, 15.4% and 44.4 % of Video packets, Data packets and Voice packets respectively are dropped due to buffer overflow at the congested node when using OSPF. It is applicable for any kind of traffic, where traffic volume is small and traffic type is normal. However, it is based on traffic scenarios, which have been implemented in this research paper. This work gives an opportunity to analyze the complex scenario in network traffic management, which would help future implementation in this context. As multipath routing is the case of uneven length of paths. The current approach demonstrated here, was able to track load, delay as well as link capacity for taking decisions.

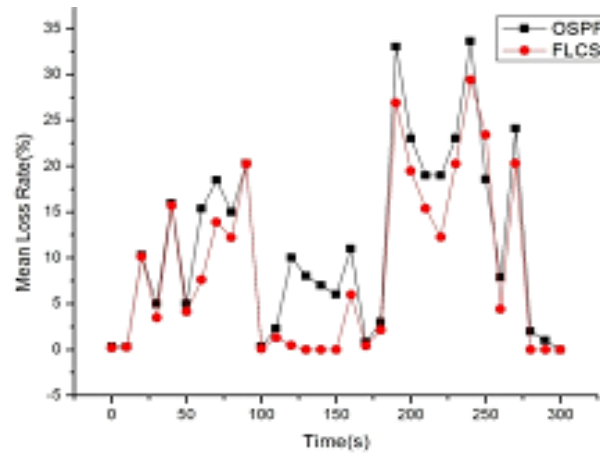
So the framework is capable of performing non-redundant search and tracking in order to perform congestion free traffic flow. It is an artificial intelligence based self-configurable framework with enhanced scalability. No such framework exists in literature that can simultaneously perform routing and traffic splitting using fuzzy predictions. It was found that fuzzy approach used here, is performing better as compared to previous work. In this way, we were able to build a “soft controller” supporting a “hardware controller”, as traffic splitting, traffic flow control is better, more responsive in nature. These findings predict that the fuzzy traffic monitor proved as more efficient solution and constitute a good alternative to some conventional methods of traffic engineering based on complex optimization.

VII. FUTURE DIRECTIONS

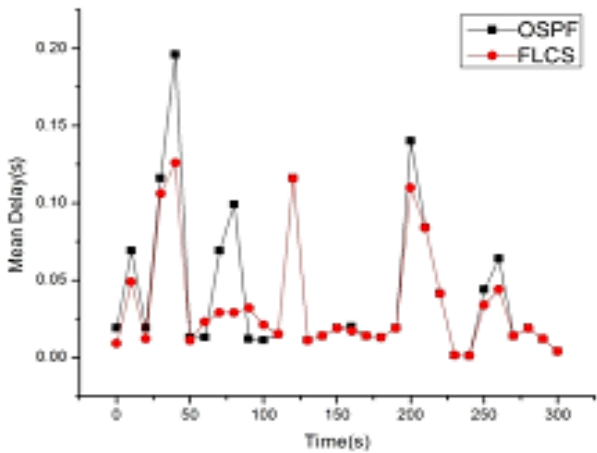
In Future, work can be carried out to validate the approach in more simulated dynamic scenarios. The interest of such a work has a big impact for the applications related to the networks, in particular those dedicated to the mission critical application.



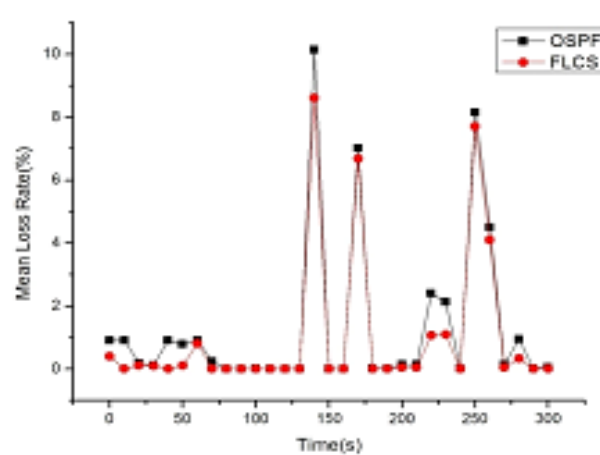
(a)



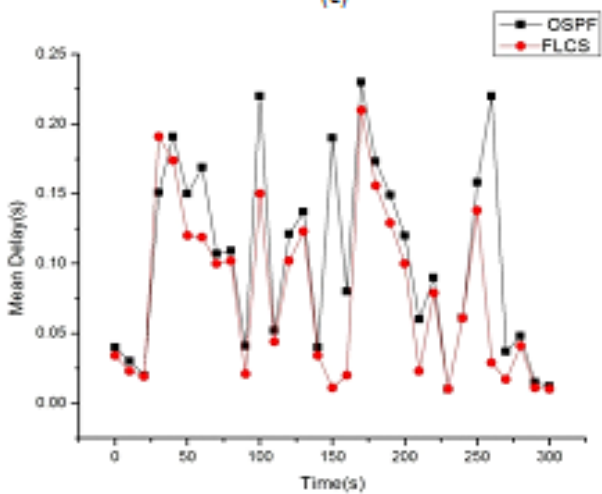
(b)



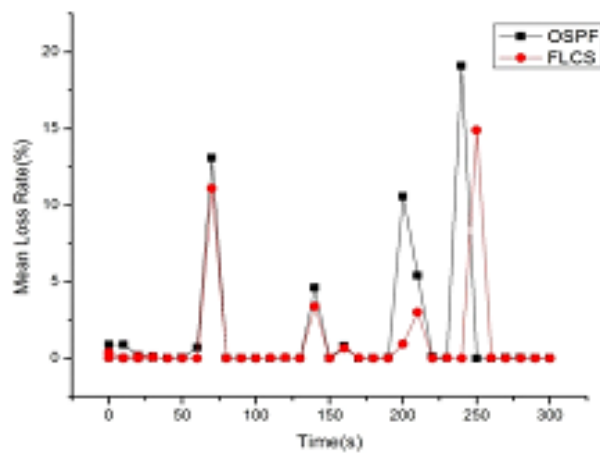
(c)



(d)



(e)



(f)

Fig. 6. (a)(b) Variation of Mean delay(s) and Mean Loss Rate (%) for Video Traffic with respect to time(secs)
(c)(d) Variation of Mean delay(s) and Mean Loss Rate (%) for Data Traffic with respect to time (secs)
(e)(f) Variation of Mean delay(s) and Mean Loss Rate (%) for Voice Traffic with respect to time (secs)

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Verification of Statecharts Using Data Abstraction

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Abstract—We present an approach for verifying Statecharts including infinite data spaces. We devise a technique for checking that a formula of the universal fragment of CTL is satisfied by a specification written as a Statechart. The approach is based on a property-preserving abstraction technique that additionally preserves structure. It is prototypically implemented in a logic-based framework using a theorem prover and a model checker. This paper reports on the following results. (1) We present a proof infra-structure for Statecharts in the theorem prover Isabelle/HOL, which constitutes a basis for defining a mechanised data abstraction process. The formalisation is based on Hierarchical Automata (HA) which allow a structural decomposition of Statecharts into Sequential Automata. (2) Based on this theory we introduce a data abstraction technique, which can be used to abstract the data space of a HA for a given abstraction function. The technique is based on constructing over-approximations. It is structure-preserving and is designed in a compositional way. (3) For reasons of practicability, we finally present two tactics supporting the abstraction that we have implemented in Isabelle/HOL. To make proofs more efficient, these tactics use the model checker SMV checking abstract models automatically.

Keywords—Statecharts; CTL; Data Abstraction; Model Checking; Theorem Proving

I. INTRODUCTION

The Statecharts formalism [1] combines a state based automata formalism with intuitive behaviour and hierarchical and parallel state composition. In addition, each state of a state chart contains data for the modeling of real world systems. This data can be naturally changed in transitions. Thus, the Statecharts formalism supports a concise and natural presentation of large models of reactive and embedded systems. It meets high acceptance in industry in particular compared to other formal methods like Z [2] or CSP [3]. The benefits of structured state and data contained in a Statechart come at a price. The validation of properties of Statecharts is a complex endeavor. The hierarchical state structure produces intricate semantic decision problems if several state transitions are enabled simultaneously at different hierarchical levels and in several parallel states. Moreover, the presence of data in states inherits a classical problem from concurrent programming: data races, i.e. successor states are ambiguous if parallel writes happen on one state variable since it is unclear which write occurs before the other.

These issues raised by the complexity of the formalism make a mechanical support for the development of Statecharts models imperative. A natural choice for verification tools for Statecharts are model checker [4] since they implicitly address state based models of transition systems. Frameworks for

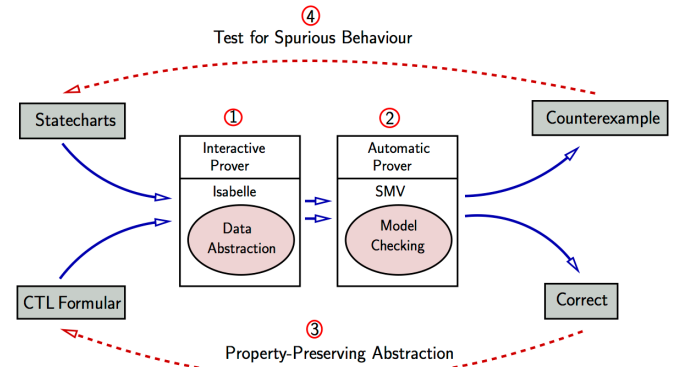


Fig. 1: Verification of Statecharts including infinite data spaces

model checking Statecharts exist, e.g. [5], but from our point of view they do not support the modeling of Statecharts well enough. One apparent reason is that many of these approaches omit the data contained in Statecharts to avoid the well-known state-explosion problem: since data domains may be infinite, e.g. integers, the state graph becomes infinite and model checking fails because it attempts a complete traversal. We propose a framework addressing this issue in our work that uses abstraction techniques making model checking applicable.

However from the point of view of a modeler, the use of Statecharts poses more pressing practical problems. Abstracting data from a Statechart adds behaviour since it omits detail about concrete decisions between transitions. This behaviour created in such an overapproximation is called spurious behaviour. It is necessary to include in order to produce faithful abstractions that allow checking the abstraction reliably while producing check results that are valid for the concrete model. The concrete models are flattened in this abstraction process and become unreadable for the model designer who does not recognise the original data conditions. For example, if a counterexample is found that is based on spurious behaviour, this is hard to understand for the user since it is behaviour that has not been there in the concrete model.

We counter this problem by proposing abstractions that are again Statecharts with readable data conditions (for example, as conditions on transitions). To retain the convenience of model checking while still supporting a realistic Statecharts formalism that contains real data, we employ Higher Order Logic theorem proving with Isabelle/HOL [6] in combination with model checking. As illustrated in Fig. 1, we input a full data containing Statechart and a property to verify on this chart

into Isabelle (1). We then apply data abstraction to enable a representation in the model checker SMV (2). Applying SMV we either find a counterexample to the property to be verified – in which case we test for spurious behaviour (4) – or SMV accepts the model and property – which means the original Statechart has the required property since we use a property preserving abstraction (3).

The additional use of a Statecharts formalisation in Isabelle/HOL provides a rigorous quality assurance: no hand-made abstraction – commonly used when model checking Statecharts – can endanger soundness of verification results. Moreover, we conservatively embed the theory of abstraction itself in Isabelle/HOL whereby the actual abstraction process guarantees property preservation. It is not practical (though perfectly possible) to abstract a data containing Statechart in Isabelle/HOL using the logical theory. Therefore, we offer additional tactic support for abstraction that automatically produces – given suitable abstraction predicates – the abstract Statechart’s representation plus the proof obligations that must be provided by the user to guarantee wellformedness (naturally being assisted by Isabelle’s powerful automated proof procedures).

We use a description of Statecharts by Hierarchical Automata (HA) an established representation of Statecharts [7], [5]. HA enable the structural decomposition of Statecharts into so-called Sequential Automata (SA). This structural decomposition makes efficient and concise proofs possible but also supports data modelling. The data modelling is the central aspect of our proposed framework. The accompanying abstraction needs to be *structure-preserving* – i.e. it must preserve the Statechart’s structure. This implies that the *data* abstraction must be respected as well by the abstraction of the Statechart. Abstraction has thus a compositional aspect: independent abstraction of the single SAs must be transferable to the abstraction of the HA containing these SAs. For example, the correctness condition *AbsCorrect* (see Definition 3.1 in Sec. III-D2) is a prerequisite for application of the model checking abstraction.

In this paper, after a detailed description of related work, we first present a motivating example of a safety injection system for nuclear power plants. We then give an overview of the formalisation of the abstract syntax and semantics of Statecharts in Isabelle/HOL (Sec. II). We next introduce in detail the formalisation of the abstraction theory in Isabelle/HOL (Sec. III). The abstraction theory is naturally divided in two parts: the abstraction process for a single SA (Sec. III-B) and how it is composed into an abstraction for the composite HA (Sec. III-D). Finally, we present the practical working with the framework for model checking Statecharts (Sec. IV). We introduce the tactic support and illustrate it on the running example. The Isabelle/HOL formalisation is part of the Archive of Formal Proof [8] and can be downloaded from there. The paper closes with conclusions and ideas for future work in Sec. V.

A. Related Work

The first formalisation of Statecharts in a theorem prover has been provided, to our knowledge, by Nancy Day in 1993 [9]. In this Master’s project, she translated Statemate Statecharts into the input language of the HOL-Voss tool [10]. The HOL-Voss system is an integrated tool consisting of the

theorem prover HOL [11] and a symbolic model checker. The focus of Day’s work is not so much on the efficient formalisation within the HOL-system; rather, she aims at providing a front end for the verification of CTL formulas using the model checker that has been integrated into HOL-Voss. Although this formalisation of Statecharts does not use HA, there is a strong connection to our work. In particular, Day’s approach supports the handling of data variables. The resulting effects – for example racing – are also touched on in the formalisation. However, the formalisation of data spaces has been addressed much more thoroughly in our work. By contrast, the work of Nancy Day contains no description of how temporal formula can be interpreted semantically with respect to a Statecharts specification. Day also fails to address data-abstraction concepts for Statecharts, for which our formalisation paves the way.

We are also familiar with a work on the formalisation of Statemate Statecharts in the proof assistant KIV [12]. KIV (*Karlsruhe Interactive Verifier*) [13] is an interactive theorem prover based on the Logic of Computable Functions (LCF) [14]. The authors define a sequent calculus for the verification of Statecharts specifications with infinite data spaces. However, unlike our work, the data spaces are described by a separate algebraic specification. Moreover, the approach differs in that an asynchronous macro-step semantics has been formalised for Statecharts, which differs from the synchronous step semantics used in our work.

Formalisations for UML Statecharts also exist for the theorem prover PVS [15]. In 2000, for example, Issa Traore presented a formalisation for UML Statecharts that was expressed in the input language of PVS [16]. In particular, he documents in his work how UML Statecharts can be verified using a model checker available in PVS. Based on this work, Demissie B. Aredo developed a similar – but more elaborated – formalisation [17]. In contrast to our work, this is a formalisation of UML Statecharts, which is not based on HA.

Concerning the model checking of Statecharts, we have to distinguish existing approaches according to which Statecharts semantics they use. We focus here on related work mostly with respect to the Statemate-Statecharts – for detailed information on how it relates to verification of UML state machines see [18]. However we discuss one recent work that addresses model checking of UML models, since it is fairly close to our work [19]. The authors present a CEGAR-like method for abstraction and refinement of behavioural UML systems. The behaviour of a system is there represented by the communication of a finite number of UML state machines. To verify properties effectively and automatically they use a so called model-to-model transformation, which means that - similar to our approach - the result of abstractions and refinements can be again an UML model. However the main difference to our work is that they abstract and refine the interfaces of the state machines only, which does not work for Statemate-Statecharts. Further they use a “don’t-know” value for data variables, which changes the semantics of UML state machines slightly.

Our approach to the model checking of Statecharts is much in the tradition of the work of Erich Mikk. He proposes in his work a semantics for Statecharts based on Extended HA [5] and uses this as a basis for model checking. He defines and implements one translation to the input language of the

CTL model checker SMV and a second one to the input language Promela [20] of the LTL model checker SPIN [21]. In contrast to our work, Mikk's work does not address Statecharts containing data.

Jan-Juan Hiemer suggests in his dissertation [22] to verify Statechart-Statecharts with the CSP model checker FDR [3]. In a work by Bill Roscoe and Zhenzhong Wu, the translation of Statechart-Statecharts to CSP is optimised – amongst other things – by a simple treatment of data [23]. However, this approach is restricted to finite data domains. Conflicts caused by concurrent write to the data space is represented by a special error-event unlike our solution with an interleaving semantics.

Udo Brockmeyer and Gunnar Wittich use a symbolic model checker produced by Siemens for the verification of Statechart models [24], [25]. They emphasise in their work the treatment of time-constraints specified in Timed CTL – an extension of CTL by discrete time. They do not support data spaces.

B. Relation to our Previous Work

Some parts of this journal paper have been published at two conferences [26], [27] and a workshop [28]. However, apart from the doctoral dissertation [18], which is written in German, there is no publication that presents these parts in an overall context. Furthermore, in this paper we give not only an idea on how single items can be related, we revisit our previous work and present our improved concepts based on a reworked formalisation [8].

A first formalisation of Statecharts by HA was presented in [26] and relates to Sec. II. Note, that this formalisation includes no data spaces. Further, this conference paper is focused on the comparison between the traditional set-based encoding of HA and an alternative optimised variant which exploits the tree-like structure of HA using Isabelle's datatypes and primitive recursive functions. This optimised version should only be used if the main focus is on proving Statecharts properties completely within Isabelle (and not outsourcing them to other automated tools). For practical applications of Statecharts we do not recommend this.

A first idea how to formalise data spaces of HAs was given in [28]. This workshop paper deals with the challenge how to formalise a partial update on a single data partition without influencing the rest of the data space. This is realised by so called generic update function that abstract over other partitions using a lambda abstraction. However from today's perspective we believe that this encoding is too technical and is solved more elegantly by a special type for partial update-functions [8]. Note, that only this new formalisation allows us to define the properties of an abstraction function in precise manner (cf. Definition 3.1), which we did not achieve before.

Finally a first version of our abstraction theory on HA is sketched in [27]. This paper introduces the so called abstraction operators to build a structure- and property-preserving abstraction of an HA inside of Isabelle. In this journal paper we omit the detailed description of the operators. Instead, for practical reasons we focus on constructing the abstraction outside of Isabelle using more efficient algorithms, which is only very roughly described in [27].

C. Example Specification: Safety Injection System

In this section we introduce a short example specification to model the behaviour of a safety-critical system. We consider

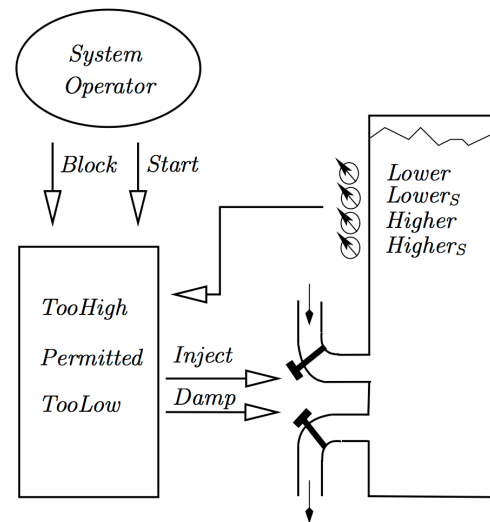


Fig. 2: Construction of a cooling system in a nuclear power plant

a reactive cooling system, which can be installed in a nuclear power plant. In the following sections we reuse this example to illustrate the verification technique.

Fig. 2 shows the construction of the system: on the upper left is the global system operator capable of starting or blocking the system. The control unit, depicted below, once started, observes the actual cooling loop on the right via four sensors and can manipulate it via two valves. Depending on the pressure in the loop of the cooling system the valves have to be opened by the control software. The sensors *Lower* and *Higher* detect big changes in pressure whereas *Lower_s* and *Higher_s* detect small changes. For example, in case that the water pressure is falling under a limit (*TooLow*), the control software must generate an event (*Inject*) that causes the upper valve of Fig. 2 to open. Thereby, new cooling water comes into the loop. Similarly, the lower valve of Fig. 2 must be activated by an event (*Damp*) if overpressure occurs.

A first description of this application was published by David L. Parnas [29]. The functionality there is restricted to injecting cooling fluid into the loop. After some years, this description was rendered more precise in an SCR requirements specification [30]. Furthermore, Bultan et al. have extended the example by reducing steam in the case of overpressure [31]. Based on this work, we have developed a Statecharts specification as depicted in Fig. 3. On the top level, the specification consists of three states, which are composed in parallel. In the following, we describe the behaviour for each of these parallel states. The specification contains a data space, which consists of the single integer variable *pressure*.

The controller for both valves is represented by State *DeviceCTRL*. Initially, the system is blocked. Event *Start* indicates that the controller has to be activated. At any time, this activation can be canceled by Event *Block*. Initially, the system is situated in Substate *Idle*. Depending on the activated substate in State *PressureMode*, the controller changes into Substate *Damp* or Substate *Safety-Injection*. If the water pressure is in a regular segment, we still are in State *Idle*. If the event

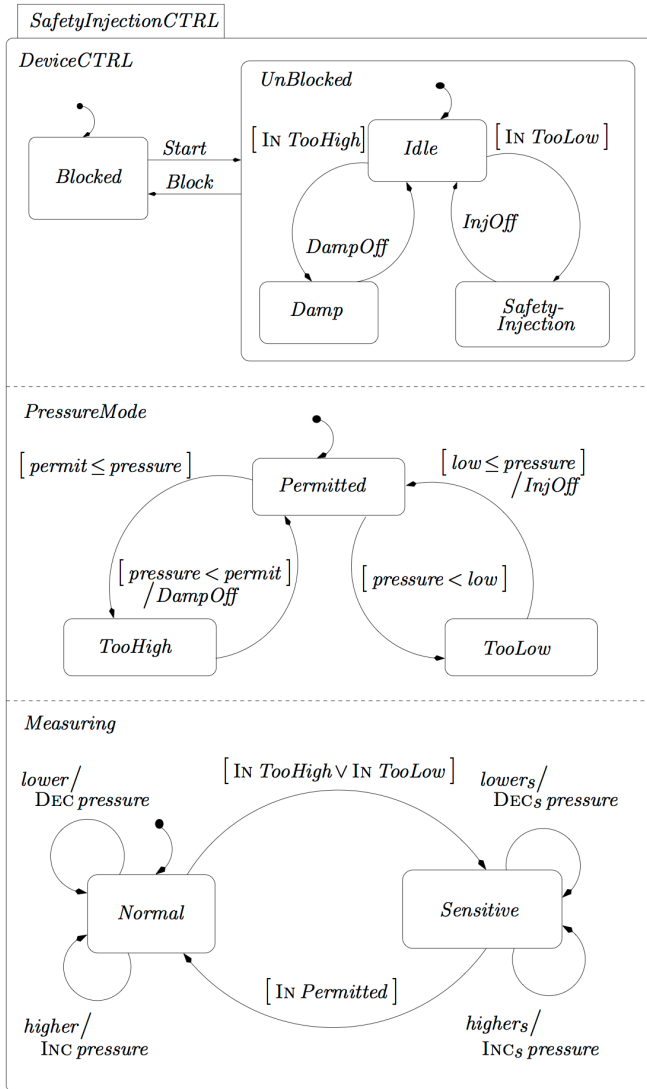


Fig. 3: Statecharts Specification of a *Safety Injection System*

DampOff occurs, State *Damp* must be left. Similarly, State *Safety-Injection* must be left, if Event *InjOff* occurs.

In the middle of Fig. 3, we model State *PressureMode*. Depending on the constants *permit* and *low*, we determine whether the value of the data variable *pressure* is in the regular segment (*Permitted*) or in a critical segment. Critical segments can be an overpressure (*TooHigh*) or a too low pressure (*TooLow*). Whenever a critical situation can be averted, the value of *pressure* returns to the regular segment. This effect comes along with generating Event *DampOff* or Event *InjOff* in order to re-close the appropriate valves.

Finally we describe State *Measuring*, which is depicted at the bottom of Fig. 3. In this state we determine the value of the data variable *pressure*. First, we consider the measuring procedure for the regular mode. This means that the data variable is in the regular segment and State *Normal* is activated. The pressure change is discovered by sensors of the environment. If the pressure has increased, Event *higher* is generated by an appropriate sensor. Consequently, the

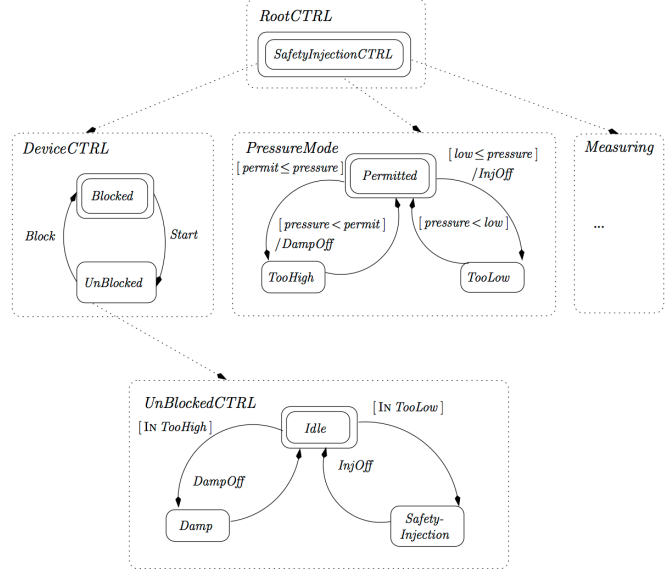


Fig. 4: Hierarchical Automaton of the *Safety Injection System*

data variable must be changed by the operator *INC*. Similar Event *lower* is generated for a decreased pressure and the variable must be changed by the operator *DEC*. Whenever *pressure* values outside of the regular segment are reached, the measuring becomes more sensitive. This is modeled by the state transition between State *Normal* and State *Sensitive*. In contrast to the measuring procedure of the regular mode, the pressure change is discovered by more sensitive sensors. In the model, this is reflected by the Event *higher_s* and the Event *lower_s*. Consequently the data variable must be changed by the operators *INC_s* and *DEC_s*.

II. FORMALISING STATECHARTS IN ISABELLE/HOL

A Statecharts specification can be adequately represented by a Hierarchical Automaton¹ which consists of a finite set of Sequential Automata. The Hierarchical Automaton of the safety injection system in Fig. 4 gives a basic impression of this representation. We have thus decomposed the Statecharts into five Sequential Automata, which are connected by arrows in order to represent the hierarchy of the specification. In this section, we describe a formalisation of Statecharts using Hierarchical Automata in Isabelle/HOL.

We begin by introducing the definition of Hierarchical Automata. To this end, we provide a *type* for the description of Hierarchical Automata by building the type abstraction over a set. In Definition 2.1, we represent the composition of Sequential Automata as a partial function by the type constructor \rightarrow . Sequential Automata are basically labelled transition systems; to keep the exposition concise we omit its definition here; for the full definitions see [8].

¹Note, we support a subset of Statechart's syntax only, e.g. an inter-level transition cannot be described by an ordinary HA and would need the expressive power of EHA (*Extended Hierarchical Automata*). However, Mikk et. al have shown[7], [5] that such an extension is straightforward and does not imply any fundamental restrictions, because this coding is structure-preserving as well.

Definition 2.1 (Hierarchical Automata (HA)): Let σ be a type of state identifiers, ϵ a type of event identifiers, and δ a type for the data space. Then a Hierarchical Automaton HA over $(\sigma, \epsilon, \delta)$ is represented by a quadruple (S_A, E, F_{Comp}, D) , where

- S_A is the set of sequential automata
- E is the set of events
- F_{Comp} is the composition function, and
- D is the initial assignment of the data space

of the Hierarchical Automaton HA. These components must fulfil the internal consistency condition `HierAutoCorrect` on Hierarchical Automata. The type $(\sigma, \epsilon, \delta)hierauto$ consists of all Hierarchical Automata over $(\sigma, \epsilon, \delta)$.

$$\begin{aligned}
 (\sigma, \epsilon, \delta)hierauto \equiv_r \{ & (S_A, E, F_{Comp}, D) \mid \\
 & (S_A :: ((\sigma, \epsilon, \delta)seqauto set)) \\
 & (E :: \epsilon set) \\
 & (F_{Comp} :: (\sigma \rightarrow ((\sigma, \epsilon, \delta)seqauto set))) \\
 & (D :: \delta data). \\
 & HierAutoCorrect S_A E F_{Comp} D \} \\
 & \text{justified by HierAutoNonEmpty}
 \end{aligned}$$

The above definition is an example of an Isabelle/HOL type definition. Type definitions are the basic building block for so-called *conservative extensions* of HOL. A new type is defined by a defining predicate (in the above case `HierAutoCorrect`) over an existing type. The new type is then given as a disjoint copy of all elements of the old type that fulfil the defining predicate. This predicate becomes a *well-formedness* condition for all elements of the new type. Two internally created bijections between the new type and the elements of the old type identified by the predicate – in the example `Abs_hierauto` and `Rep_hierauto` – enable a translation between old and new type. A type extension is conservative because new predicates and operators over the new type may only be defined based on the operators of the base type implicitly assuming the well-formedness predicate for elements of the domain of new functions.

For the above example, we omit the actual definition of `HierAutoCorrect` (see [8] for a complete presentation of all definitions). It essentially encodes that F_{Comp} is a well-formed composition with respect to S_A as defined in the following Definition 2.2.

Definition 2.2 (Well-Formed Composition): Let S_A be a set of Sequential Automata and F_{Comp} a composition function. The predicate `IsCompFun` describes the internal consistency for these components.

$$\text{IsCompFun} ::_c [(\sigma, \epsilon, \delta)seqauto set, \sigma \rightarrow ((\sigma, \epsilon, \delta)seqauto set)] \rightarrow bool$$

$$\begin{aligned}
 \text{IsCompFun } S_A F_{Comp} \equiv_c & \\
 \text{dom } F_{Comp} = (\bigcup A \in S_A. \text{States } A) \wedge & \\
 (\bigcup \text{ran } F_{Comp}) = (F - \{\text{Root } S_A F_{Comp}\}) \wedge & \\
 \text{RootExists } S_A F_{Comp} \wedge & \\
 \text{OneAncestor } S_A F_{Comp} \wedge & \\
 \text{NoCycles } S_A F_{Comp} &
 \end{aligned}$$

The predicate `RootExists` $S_A F_{Comp}$ guarantees that there exists an unique root automaton called `Root` $S_A F_{Comp}$. The predicate `OneAncestor` $S_A F_{Comp}$ reflects that each Sequen-

tial Automaton of S_A except the root automaton has exactly one ancestor state. The last predicate `NoCycles` ensures that the composition function does not contain any cycles.

$$\begin{aligned}
 \text{NoCycles } S_A F_{Comp} \equiv_c & \\
 \forall s : \mathbb{P} (\bigcup A : S_A. \text{States } A) . & \\
 s \neq \emptyset \implies & \\
 \exists s : S. s \cap (\bigcup A : \text{the}(F_{Comp} s). \text{States } A) = \emptyset &
 \end{aligned}$$

A. Optimisation

When applying the formalisation introduced in the previous section, we frequently need selection theorems for a Hierarchical Automaton HA to obtain its constituents, e.g. all its defining Sequential Automata. We must provide such selection theorems in Isabelle/HOL by proving them. For these proofs, we need a special theorem, that reflects the well-formedness property of Hierarchical Automata (essentially Definition 2.2). Deriving a well-formedness property is expensive. For example, the check that there are no cycles in the composition function involves checking a predicate for each non-empty subset of the state set of HA. We thus obtain $2^{\#(\text{States } HA)}$ different proof obligations. Clearly, this proof is inefficient even for small Hierarchical Automata.

To solve this problem, we propose a stepwise procedure to construct Hierarchical Automata from their defining Sequential Automata. Here, we need to exploit the tree-like structure of Hierarchical Automata. We define two constructors: one to build a pseudo Hierarchical Automaton from a given Sequential Automaton, and another to add a given Sequential Automaton at a specified state position to a Hierarchical Automaton extending the constituents of the Hierarchical Automaton. The idea is that this construction process guarantees wellformedness thus reducing the cost of proof. We start with the definition of a pseudo HA. The result of this construction does not contain a hierarchy and represents in fact a Sequential Automaton, which is wrapped up as a Hierarchical Automaton.

$$\text{PseudoHA } SA D \equiv_c \text{Abs_hierauto}(\{SA\}, \text{Events } SA, \text{EmptyMap } (\text{States } SA), D)$$

The above used operator `EmptyMap` defines a composition function, in which each state in the set of states of an SA is mapped onto the empty set. The operator ensures that a pseudo HA contains only simple states.

Using the following construction operator \boxplus , we extend a Hierarchical Automaton by a Sequential Automaton at a specified state position.

Definition 2.3 (Extension of a HA by a SA): Let HA be a Hierarchical Automaton, S a state and SA a Sequential Automaton. Then the extension of HA by SA in the state S is defined as follows.

$$\begin{aligned}
 _ \boxplus _ & ::_c [(\sigma, \epsilon, \delta)hierauto, \sigma * (\sigma, \epsilon, \delta)seqauto] \\
 & \rightarrow (\sigma, \epsilon, \delta)hierauto \\
 _ \boxplus _ & \equiv_c (\lambda HA (S, SA) . \\
 & \text{let } S'_A = \{SA\} \cup (SAs HA); \\
 & \quad E' = \text{Events } HA \cup \text{Events } SA; \\
 & \quad F'_{Comp} = \text{CompFun } HA \oplus (S, SA) \\
 & \quad D' = \text{InitData } HA \\
 & \text{in Abs_hierauto}(S'_A, E', F'_{Comp}, D'))
 \end{aligned}$$

In Definition 2.3, we use the operator \oplus to extend the composition function F_{Comp} by a Sequential Automaton SA on

```

SafetyInjectionSystem ::_c (string,string,ds) hierauto
SafetyInjectionSystem ≡_c
(PseudoHA RootCTRL (LiftInitData [V0,15]))
⊕ ("SafetyInjectionCTRL",DeviceCTRL)
⊕ ("SafetyInjectionCTRL",PressureMode)
⊕ ("SafetyInjectionCTRL",Measuring)
⊕ ("UnBlocked",UnBlockedCTRL)

```

Fig. 5: Stepwise construction for the model of the *Safety Injection System*

a state S . The extension is well-formed if and only if two conditions hold. First, S must be contained in the domain of F_{comp} , and second, S must not be a state of SA . These conditions ensure that the construction using the operator \oplus yields a cycle-free composition function.

In Fig. 5 we present the definition of the running example of the safety injection system using construction operators. The operator $InitDS$ represents an initial data space assignment. $Pressure$ is there assigned to 15.

The benefit of such a construction is that we can derive theorems automatically reflecting selection theorems of constructed Hierarchical Automata. To achieve such automation, we have implemented a tactic in Isabelle/HOL. The tactic decomposes a constructed Hierarchical Automaton into its components and derives for each component a selection theorem, e.g for the selection of Sequential Automata. These selection theorems are subsequently reused to derive one selection theorem of the whole Hierarchical Automaton. To decompose a constructed Hierarchical Automaton, we use theorems reflecting the well-formedness property of Hierarchical Automata under construction.

A pseudo HA is always well formed.

```
HierAutoCorrect {SA} Events SA EmptyMap (States SA) D
```

Accordingly, we have derived for the construction operator \boxplus a theorem of well-formedness. However, this property is not always satisfied. We have framed the conditions, which have to be proved.

Theorem 2.4 (Well-Formed \boxplus - Construction):

$$\begin{array}{c}
 \boxed{\text{States } SA \cap \text{States } HA = \emptyset} \quad \boxed{S \in \text{States } HA} \\
 S'_A = \{SA\} \cup (SAs \ HA) \quad E' = \text{Events} HA \cup \text{Events } SA \\
 F'_{comp} = \text{CompFun } HA \oplus (S, SA) \quad D' = \text{InitData } HA \\
 \hline
 \text{HierAutoCorrect } S'_A \ E' \ F'_{comp} \ D'
 \end{array}$$

The well-formedness theorems help to identify and construct the conditions used in the construction of elements of the type of Hierarchical Automata. To round off the smooth construction with the optimised constructor operators we only need the following set of theorems that helps to lift the selection operators to constructed HAs. To access the SA contained in a pseudo HA we use the following initial theorem.

$$SAs (PseudoHA \ SA \ D) = \{SA\}$$

For the general case, the previously identified well-formedness assumptions (see Theorem 2.4) help to select SA in a constructed HA.

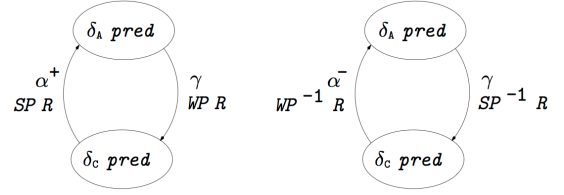


Fig. 6: Galois Connections based on predicate transformers

Theorem 2.5 (Selecting SA of a HA Construction):

$$\frac{\text{States } SA \cap \text{States } HA = \emptyset \quad S \in \text{States } HA}{SAs (HA \boxplus (S, SA)) = \{SA\} \cup SAs \ HA}$$

With the formalisation of these original construction operators we end here the presentation of the basic formalisation of Statecharts in Isabelle/HOL omitting the semantics. It follows – as the basic syntax – quite closely the work by Mikk [5]. Its Isabelle/HOL representation is detailed in [8]. We next consider the abstraction process of Statecharts as a preparation for model checking.

III. DATA ABSTRACTION

Our formalisation in Isabelle/HOL [8] contains a theory of the temporal logic CTL and a theory of the specification language Statecharts. Both theories are integrated semantically. In principle, this formalisation enables the derivation of CTL properties on a Statecharts specification. However, the drawback is that the user has to prove these properties in an interactive way. Our approach combines Isabelle/HOL with the model checker SMV to automate the reasoning. The core of this combination is a data abstraction theory for Statecharts, which we introduce in this section.

A. Property Preserving Abstraction

We investigate property preserving data abstraction for Statecharts to preserve CTL formulas. In the literature we find two major approaches for transition systems: over- and underapproximation.

In an overapproximation the abstract model can contain new behaviour, but old behaviour cannot be lost. That is, properties of the universal fragment of CTL (\forall CTL), that are valid on all paths of the overapproximated abstract model, must hold on the paths of the concrete model. By contrast in an underapproximation, new behaviour cannot be added, but old behaviour can be lost. Accordingly, properties of the existential fragment of CTL (\exists CTL) are preserved by underapproximated abstract models. In the work of Dams [32] two automata representing these different kinds of abstractions are generated. Depending on the property that has to be verified the appropriate model has to be chosen.

The theoretical basis for over- and underapproximations are Galois connections [33]. For complete lattices, C and A , a pair of monotone maps, $\alpha ::_c C \rightarrow A$ and $\gamma ::_c A \rightarrow C$ define a Galois connection, written *GaloisCorrect* $A \ C \ \alpha \ \gamma$, iff $\alpha \circ \gamma \leq_A id_A$ and $id_C \leq_C \gamma \circ \alpha$. Fig. 6 represents two instantiated Galois connections relating predicates on an abstract (δ_a) and a concrete (δ_c) data space. Since they are predicates, these elements are ordered by implication.

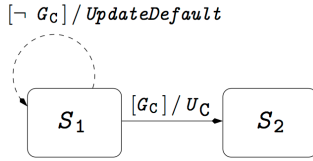


Fig. 7: Implicit behaviour of a SA

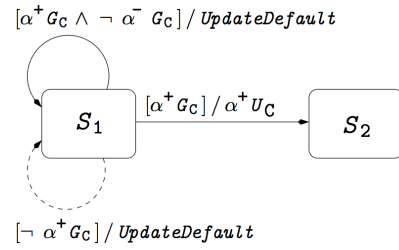


Fig. 8: Overapproximation of a SA

On the left side we define the Galois connection

$$\text{GaloisCorrect}(\delta_a \text{ pred})(\delta_c \text{ pred}) \alpha^+ \gamma$$

for overapproximation to weaken a concrete predicate pc by the abstraction α^+ as is reflected in the Galois property $pc \implies \gamma(\alpha^+ pc)$. Correspondingly on the right, the Galois connection

$$\text{GaloisCorrect}(\delta_c \text{ pred})(\delta_a \text{ pred}) \gamma \alpha^-$$

for underapproximation strengthens concrete predicates. This is reflected in the Galois property $\gamma(\alpha^- pc) \implies pc$.

Given an abstraction function $R ::_c \delta_c \rightarrow \delta_a$ mapping elements of the concrete data space into the abstract data space, we can generally define α^+ , α^- and γ . The following definitions of strongest post- and weakest precondition form two Galois connections for any such R (cf. Fig. 6).

$$\begin{aligned} SP R P &\equiv_c \lambda a. \exists c. (Rc) = a \wedge P c \\ WP R P &\equiv_c \lambda c. \forall a. (Rc) = a \implies P a \\ SP^{-1} R P &\equiv_c \lambda c. \exists a. (Rc) = a \wedge P a \\ WP^{-1} R P &\equiv_c \lambda a. \forall c. (Rc) = a \implies P c \end{aligned}$$

In our theory, the abstraction function is total. Hence, the weakest precondition coincides with the inverse strongest postcondition ($WP R = SP^{-1} R$). Furthermore, a Galois connection includes that α^+ can be expressed by α^- which is reflected by the property $SP R (\neg P) = \neg (WP^{-1} R P)$.

To avoid the detailed definitions of the abstraction functions, we use in the next section an abbreviation. Analogous to the depiction of Fig. 6, we abbreviate $SP R$ and $WP^{-1} R$ by α^+ and α^- omitting the parameter R .

B. Overapproximation of SA

Statecharts, HA, and accordingly SA belong to the family of synchronous languages. Synchronous languages build on a synchronous step semantics, which we have formalised for HA in Isabelle/HOL [8]. It includes a formalisation of SA. The semantics of SA is there a special case of the semantics of HA because an SA can be viewed as an HA without hierarchy (cf. the definition of a pseudo HA in Sec. II-A). Based on this decomposition, we present in this section the overapproximation of SA independently from HA. Reusing this theory, we introduce in the next section the overapproximation of HA.

One special property of synchronous languages is that in each semantical status – synchronised by a global clock – the system performs a defined calculating step. Semantical statuses of SA, where no transitions fire, perform a trivial calculating step, in which the data variables are assigned to the previous value. This effect can be interpreted as complementation by implicit transitions. On the left side of Fig. 8, this is depicted by a dashed self-transition, where the guard

$\neg G_C$ is constructed as the negated guard of the exiting transition. The action-part $UpdateDefault$ represents that the data variables are assigned to the previous value. Note that $UpdateDefault$ will be an unwanted effect in the case of writing data variables by a synchronous executed transition, which must be prioritised. Such synchronous executed transition could arise from an SA which is composed in parallel to the considered SA. A more detailed discussion, how the SAs inside a HA mutually affect each other, can be found in Sec. III-D.

As a rule, the guard of the implicit transition must be constructed as the conjunction of negated guards of all exiting transitions. Overapproximating an SA, we construct an identical structured SA. We adopt the control states and abstract the transitions. Abstracting transitions, we abstract guards and updates separately.

In general, it is impossible to construct an abstracted guard which exactly describes a concrete guard G_C . First, we propose to weaken G_C by α^+ using overapproximation. Building such weaker guards adds new behaviour to the model but deletes some implicit behaviour simultaneously caused by the special semantics of synchronous languages. The reason is that the guard of the implicit transition $\neg \alpha^+ G_C$ is automatically stronger. Therefore – secondly – we must add a suitable self transition, to adjust this unwanted effect. The guard of this self transition must be constructed by a conjunction of the overapproximated guard of G_C and the negated underapproximated guards of all exiting transitions. This procedure is illustrated on the right side of Fig. 8.

Abstracting the example, only one exiting transition has to be considered. First, we abstract the guard G_C by α^+ and introduce a self transition. The guard of this self transition is constructed by a conjunction of the overapproximated guard of the exiting transition $\alpha^+ G_C$ and the negation of the underapproximated guard of the exiting transition $\neg \alpha^- G_C$. The latter can be expressed by α^+ using the theorem of subsection III-A, so that finally we obtain the following guard for the self transition.

$$[\alpha^+ G_C \wedge \alpha^+ \neg G_C]$$

In Fig. 8, only one exiting transition exists for State S_1 . For n transitions, the guard of the self-transition would be constructed in the following way.

$$[\alpha^+ G_C^1 \wedge \neg \alpha^- G_C^1 \wedge \alpha^+ G_C^2 \wedge \neg \alpha^- G_C^2 \wedge \dots \wedge \alpha^+ G_C^n \wedge \neg \alpha^- G_C^n]$$

Building traditional overapproximation of updates [34] is compatible with SAs because a weaker update adds new behaviour to the system but old behaviour cannot be lost. Accordingly on the right side of Fig. 8, the update is overapproximated

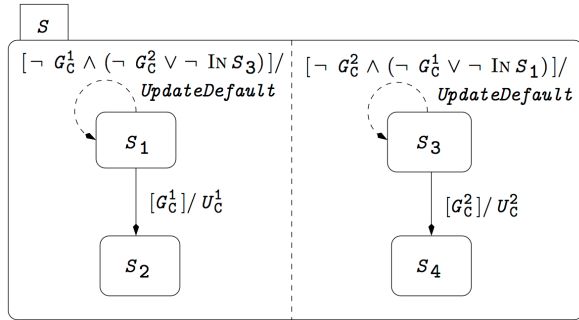


Fig. 9: Implicit behaviour of a HA

by α^+ . However, in general, abstracting updates results in non-constructive predicates that violate the action language of Statecharts. More precisely, for each U_C we obtain in general more than one abstract update by α^+ .

We must restrict to overapproximation because SAs do not allow a reduction by underapproximation. This is caused by the special semantics of synchronous languages, which can be interpreted as a complementation by implicit transitions. This complementation restricts the possibility for reducing behaviour fundamentally. Consider the example in Fig. 8 on the left side. If we propose to build a stronger guard of G_C by α^- , we obtain a weaker guard for the implicit transition. That is, we add new behaviour to the abstract model, which is unsound for underapproximation. Hence, for the example on the left side of Fig. 8 we cannot build an underapproximation, where the abstract model is again a SA. The only way out is to reduce nondeterministic branches to deterministic ones but this is not sufficient for a general procedure. Consequently, the result of an underapproximation cannot usually be expressed by a SA. Hence our abstraction technique is restricted to preserve properties of the universal fragment of CTL. If the overapproximation is too rough, we will obtain undesirable counterexamples, describing so called spurious behaviour. Because of the reason described above, we cannot refine the model directly. Instead in such cases we start from scratch and apply the well-known counterexample guided abstraction and refinement loop (CEGAR) [35] generating a new and more detailed overapproximation.

C. Semantical Characteristics of HAs

We are interested in developing a data abstraction technique which can be applied in a compositional manner. This means that we decompose a given HA into its defining SAs. After that, we abstract each SA independently. Finally, we compose the abstracted SAs to an abstracted HA.

In this section we describe such compositional abstraction techniques for HAs. Because of the rather complex semantics of HAs this is a challenge: in general, a compositional procedure does not yield a valid overapproximation for a given HA.

1) *Implicit behaviour of HAs:* Implicit behaviour occurs in synchronous modelling languages whenever a transition cannot fire at the beginning of a clock cycle. In this situation the statechart executes a trivial calculation step that restores the data state. In Sec. III-B the implicit behaviour has been

represented in the model by a dashed arrow (cf. Fig. 8). This special transition only fires if no other transition of the model is enabled. As is shown in Fig. 9, we can model the implicit behaviour of a HA explicitly in a similar fashion. Note, however, that with respect to compositional abstraction, the guard of the self-transition depends on context information that lies outside the SA in which the self-loop is defined.

Considering the SA on the left side of Fig. 9, we observe that the guard of the implicit self-transition of the control state S_1 holds, if and only if the guard G_C^1 of the transition exiting S_1 does not hold. In addition a predicate of the parallelly composed SA must hold. Either the control state S_3 is not active or the guard of the transition exiting S_3 is not valid. More generally, in all composed SAs of a HA there must not be any transition that is enabled. The modelling of implicit behaviour of HAs, shown in Fig. 9, becomes more complicated, because usually we have more than one local state in a SA. The concept of partial default update functions – presented in the next subsection – can avoid this effect, because they abstract from the dependencies between parallelly composed SAs (see Fig. 10).

2) *Partitions on Data Spaces and Partial Updates:* In general, the data space of a HA consists of a finite number of disjoint partitions. Update-functions can be defined in such a way that they do not write on all partitions. The semantics of HAs determines the values of partitions after transition execution also in cases in which a transition does not write on the partition.

On the left side of Fig. 10 we have modelled a data space consisting of two partitions D_C^1 and D_C^2 . The update-function of the transition between the states S_1 and S_2 assigns D_C^1 using an auxiliary function U_{PC}^1 and does not write on partition D_C^2 . Note, the second partition is assigned to *None*. This indicates in our formalisation that the partition is unwritten. That is, the update-function is partial. In one step of calculation of HA, transitions of several parallel SAs can be executed synchronously. If a transition does not write on a partition (e.g. D_C^2), the first question is whether there is another synchronously executed transition writing on this partition. If this is the case, the value of the synchronously executed transition is selected. In case of a concurring write of several transitions on one partition (so-called racing) the resulting conflict is resolved by introducing a non-determinism (interleaving semantics). In contrast, if there is no transition writing on a partition, the semantics assigns to this partition the value prior to execution of the transition. Note, that we provide a complete formal semantics of HAs including data spaces and partial update-functions in Isabelle/HOL [8].

Summarising the above, it is, in general, not possible to decide locally inside a SA, whether implicit behaviour for writing a partition on the data space will occur. Hence it is challenging to define a compositional abstraction technique, whereby a partial update-function can be abstracted independently from synchronously executed updates. In the next section we propose a procedure abstracting update-functions only based on local informations available inside a SA. Therefore, we demand special properties from the abstraction functions in order to construct overapproximations of HAs in a compositional manner.

D. Overapproximation of HA

We present in this section a data abstraction technique for HA, which is characterized by three properties. First, it is a structure-preserving abstraction. This means that the result of the abstraction is represented by a HA, whose structure is identical to the input-HA. Second, our abstraction is an overapproximation preserving properties of the universal fragment of CTL. Third, the technique can be applied in a compositional way.

1) *Partial Default Update-Functions*: First, we address the modelling of implicit behaviour inside a SA. Consider again the example of Fig. 10, which is depicted on the left side.

Applying the technique of Sec. III-B, we obtain the self-transition on the right side of Fig. 10. Note that in contrast to the dashed self-transition on the left side of Fig. 10, the guard of the abstracted self-transition is constructed only using local informations of the SA. Hence, the guard is weaker than a precise approximation which could be constructed using context informations of SAs, that are composed in parallel. A more precisely approximated guard for our example is the following predicate.

$$[\alpha^+ G_C^1 \wedge \neg \alpha^- G_C^1 \wedge (\neg (\alpha^+ G_C^2 \wedge \neg \alpha^- G_C^2) \vee \neg \text{IN } S_3)]$$

Consequently, locally constructed self-transitions will be more often executed than self-transitions, which are labeled with precise approximated guards. Nevertheless, this effect is invisible in our semantics because we have used as action a so-called partial default update-function (*PUpdateDefault*). This update-function has a slightly different effect in comparison to *UpdateDefault*. The partial default update-function represents that data partitions are only *potentially* assigned to the previous value. In the case of writing a data partition by synchronously executed transitions, the semantics of *PUpdateDefault* does not have a writing effect on this data partition. Consequently, in this case executing the self-transition is invisible in the semantical states of the abstracted HA.

2) *Data Structure Preserving Abstraction Functions*: To tackle the problem of Sec. III-C2 we propose to use an abstraction function that preserves the structure of the data space. Therefore, we demand for each concrete data partition a unique counterpart in the abstract data space. Furthermore each concrete partition has to be mapped into the domain of its abstract counterpart using a given abstraction function. Note, that this mapping must be done independently from other partitions of the data space. To ensure this requirement, we force the user of our abstraction technique to prove some special properties of the abstraction function.

Again we consider the example of Fig. 10 on the left side. There we have used a partial update-function writing the first partition of the data space only. To construct a precise overapproximation of this update-function inside a SA, we demand that the binary structure of the data space must be preserved by the abstraction function. This means that the concrete data partitions D_C^1 and D_C^2 will be mapped into the abstract data partitions D_A^1 and D_A^2 independently, which is illustrated in the middle of Fig. 10. Assuming that we are interested in using a predicate abstraction, D_A^1 and D_A^2 would be characterized by boolean variables. In this case, each boolean variable corresponds exclusively to D_A^1 or D_A^2 . Furthermore,

each boolean variable reflects the validity of a predicate on the corresponding concrete data partition.

Assuming that both data partitions are declared as integers, we can define an abstraction function for a predicate abstraction in the following way.

$$R [D_C^1, D_C^2] \equiv_{df} [D_C^1 \leq D_C^2, D_C^2 \leq 2]$$

In this example, each abstract data partition is represented by a single boolean variable. For example the partition D_A^1 is represented by a boolean variable, which is evaluated by the predicate $D_C^1 \leq D_C^2$. Accordingly, D_A^2 is represented by a variable, which is evaluated by $D_C^2 \leq 2$. It is obvious that the abstraction function does not map the data partition D_C^1 into D_A^1 independently from D_C^2 . Consequently, it is not possible to calculate a precise overapproximation of the update-function U_{PC}^1 locally. The reason is that we calculate the abstract update-function by simulating U_{PC}^1 . Therefore, we must map the result of U_{PC}^1 into the abstract data space. However, we have not much information about D_C^2 after executing U_{PC}^1 because this partition is locally unwritten. This means that the value of D_C^2 can be the previous value of D_C^2 or any other value, which is written by a synchronous executed transition. The only way out would be to construct the simulation using type properties of D_C^2 resulting in an unprecise approximation. Hence we have defined a special type of wellformed abstraction functions in order to avoid these unprecise approximations.

Definition 3.1 (Abstraction Function): Let δ_c be a type of the concrete data space and δ_a a type of the abstract data space. Then an abstraction function R over (δ_c, δ_a) is represented by a triple (L, D_C, D_A) , where

- L is the list of functions for the abstraction of data partitions,
- D_C is the concrete data space, and
- D_A is the abstract data space

of the abstraction function R . These components must fulfil the internal consistency condition *AbsCorrect* on abstraction functions. The type $(\delta_c, \delta_a) \text{ abs}$ consists of all abstraction functions over (δ_c, δ_a) .

$$\begin{aligned} (\delta_c, \delta_a) \text{ abs} \equiv_{\tau} \{ & (L, D_C, D_A) \mid \\ & (L :: (\delta_c \rightarrow \delta_a) \text{ list}) \\ & (D_C :: \delta_c \text{ dataspace}) \\ & (D_A :: \delta_a \text{ dataspace}). \\ & \text{AbsCorrect } L \ D_C \ D_A \} \\ & \text{justified by AbsNonEmpty} \end{aligned}$$

The predicate *AbsCorrect* $L \ D_C \ D_A$ guarantees that the number of concrete data partitions, the number of abstract data partitions, and the number of elements in the function list are identical. Furthermore, the predicate requires for each function of the list that the range of this function is contained in the domain of its corresponding abstract data partition.

$$\begin{aligned} \text{AbsCorrect } L \ D_C \ D_A \equiv_c & \\ \#D_C = \#D_A \wedge \#D_C = \#L \wedge & \\ \forall i < \#D_C. \text{ran}(L!i) \subseteq \text{dom } D_A i & \end{aligned}$$

Note that the operator $!$ selects the abstraction function of the i -th partition from the list L . Furthermore, the operator $\#$ gives the number of partitions for a data space corresponding to the number of abstraction functions for a list.

Based on this definition of wellformed abstraction functions, we have defined in our Isabelle/HOL-formalisation an

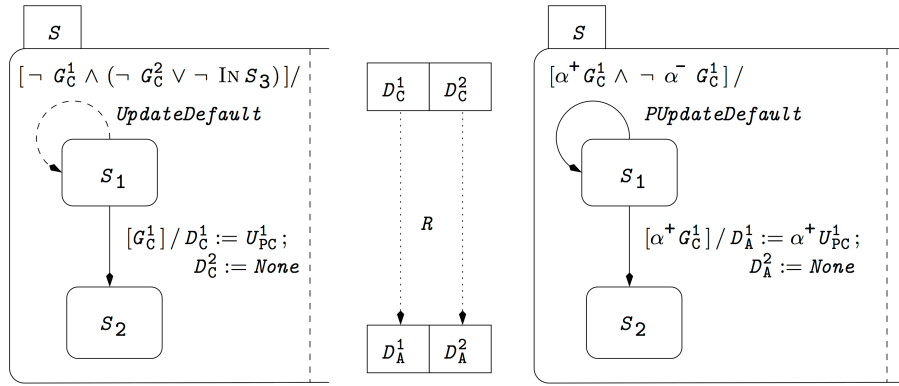


Fig. 10: Structure-preserving abstraction using partial update-functions

operator $AbsBy_{HA}^+$ constructing an overapproximation from a given HA and a given abstraction function. This operator implements the ideas of Sec. III-B and Sec. III-D in a compact manner. Basically, we decompose the HA in its defining SA and build the overapproximation of each SA independently. In order to construct the overapproximation of a SA, we have to abstract guards and update-functions of each transition of the SA. Additionally, we have to introduce self-transitions in the abstracted SA if the abstracted guard of a transition becomes weaker than the original one.

Furthermore, we have defined an operator $AbsBy_{CTL}^-$ constructing the underapproximation for a given CTL-formula and a given abstraction function. In comparison to defining abstracted hierarchical automata, $AbsBy_{CTL}^-$ is constructed directly and straightforwardly. First, the operator traverses a formula in an inductive manner getting access to all atomic propositions that are defined on the infinite data space. Second, the operator constructs underapproximations from these atomic propositions using the operator α^- (cf. Sec. III-A).

IV. MODEL CHECKING STATECHARTS

Based on the abstraction theory of Sec. III we have designed and implemented two tactics to verify Statecharts more efficiently. Fig. 11 gives an overview of the framework describing the architecture at an abstract level². On the left side, the theorem prover is depicted including all developed Isabelle/HOL-theories. In principle, proof obligations on Statecharts can be derived inside Isabelle/HOL based on these theories. However, proofs are not fully automated in the prover. They are often complex and exhausting processes involving many interactive steps. To overcome this, we have developed two tactics external to Isabelle/HOL that are invoked via the oracle interface – a specific Isabelle/HOL interface for plugging in additional support tools. The first tactic can be used to check CTL-properties for Statecharts that are defined on finite data spaces. This tactic uses the model checker SMV [36], [37] checking the properties automatically. The second tactic can be used to abstract Statecharts that are defined on infinite data spaces. This tactic implements an au-

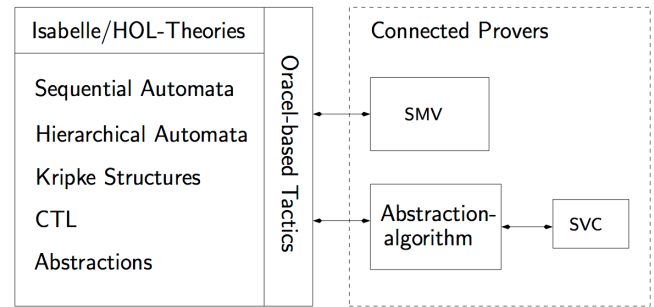


Fig. 11: Framework architecture: Isabelle-theories and connected provers

tomatic predicate abstraction and invokes SVC³[39] evaluating propositional properties including arithmetic. Note, the second tactic is limited to formulas of the universal fragment of CTL because it includes a property-preserving abstraction algorithm supporting this subclass only.

Applying the first tactic, a user of Isabelle/HOL is able to check whether a temporal formula F is satisfied by the behaviour description of a given HA representing a finite Statecharts specification.

$$HA \models_{HA} F$$

Therefore, the tactic analyses the internal term structure of the proof state in Isabelle/HOL and translates the collected information into the input language of SMV [28]. Of course, it may happen that SMV will not be able to verify the proof obligation successfully. Consequently in this case, the tactic fails. Otherwise the tactic returns true.

If a user is interested in verifying an infinite Statecharts specification, he has to apply the second tactic. In a first step, the proof obligation is reduced to the following proof state assuming that a property-preserving abstraction is used.

$$(HA AbsBy_{HA}^+ R) \models_{HA} (F AbsBy_{CTL}^- R)$$

Abstracted counterparts to HA and F are defined using the operators $AbsBy_{HA}^+$ and $AbsBy_{CTL}^-$ for a given abstraction

²Note, the whole abstraction theory including tactics is not part of [8]. However, interested readers are welcome to request the sources by email to the authors.

³The current version is ported to Isabelle 2013-2 and uses CVC3 [38], because SVC will be not longer supported by all platforms.

function R (cf. Sec. III-D). If we want to prove the proof state in Isabelle/HOL, we must exploit these definitions. In contrast to this, the tactic ignores the definitions and uses an algorithm external to Isabelle/HOL that constructs the abstraction more efficiently. Note, the tactic is applicable only if the following assumptions are satisfied.

- 1) R respects the structure of the data space satisfying the wellformedness property of Definition 3.1,
- 2) R defines a predicate abstraction interpreting the infinite data space as a finite set of atomic propositions on it, whereby each of them is represented as a boolean variable inside the abstract data spaces, and
- 3) F is a formula of \forall CTL.

For the design of the abstraction algorithm, we reuse existing work for abstracting ordinary transition systems [34] and adapt this approach to hierarchical state systems. Accordingly, we also use adjunction theorems of Galois connections defining α^+ and α^- different to our Isabelle/HOL-definitions of Sec. III-A.

$$\alpha^+ P_C \equiv_{df} \bigwedge \{ P_A \mid P_C \implies_P \gamma P_A \}$$

$$\alpha^- P_C \equiv_{df} \bigvee \{ P_A \mid \gamma P_A \implies_P P_C \}$$

The identifier P_C and P_A represent predicates on the concrete and abstract data space. An implementation of the overapproximation α^+ is sufficient, because α^- can be expressed using α^+ (cf. Sec. III-D). To calculate the overapproximation of a predicate P_C , we have to implement two steps. First, we check the proof obligation $P_C \implies_P \gamma P_A$ for all P_A of the abstract predicate type using SVC. Second, we conjugate all predicates, where the check was successful.

To implement this procedure, we need an algorithm calculating the predicate transformer γ for a given predicate efficiently. Thanks to the guaranteed properties of predicate abstraction, we are able to define γ as a substitution. For a given abstract predicate we replace a contained boolean variable by its corresponding atomic proposition, which is defined on the concrete data space only. If we do so for all contained variables, we obtain a concrete predicate as a result.

We refer to the Safety Injection System (cf. Example I-C) for an illustration of the substitution. There we have defined a single integer variable *pressure*, which is used in both guards and actions of the Statechart (cf. Fig. 3). Consequently we need to abstract these predicates algorithmically. Assuming a user of the tactic likes to interpret *pressure* by the atomic propositions $pressure < 10$ and $pressure < 20$ we introduce two boolean variables B_1 and B_2 for the abstract data space representing, whether the propositions are satisfied or not.

For example, if we like to determine the overapproximation of the guard $permit \leq pressure$ we must check for all elements of the abstract predicate type, whether they ensure the guard or not. For example, we must check for the abstract predicate $B_1 \vee_P B_2$ the following statement⁴.

$$20 \leq pressure \implies_P \gamma (B_1 \vee_P B_2)$$

First, we replace B_1 and B_2 by their corresponding predicates.

$$(B_1 \vee_P B_2) [pressure < 20/B_1, pressure < 10/B_2]$$

$$\Leftrightarrow (pressure < 20 \vee_P pressure < 10)$$

Afterwards, we evaluate the statement above to false. The

⁴For the sake of simplicity we assume, that 20 is assigned to the constant *permit*.

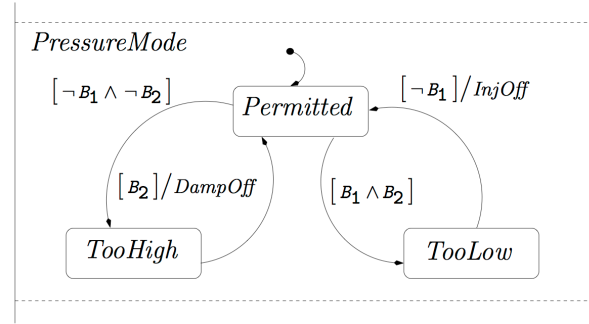


Fig. 12: Abstracted specification of the SA *PressureMode*

overapproximation of the guard is then built as the conjunction of all possible predicates that pass the test (cf. adjunction theorem defining α^+).

Note, the abstract predicate type represents atomic propositions only. Consequently, we can use a finite set of formulas based on the disjunctive normal form to represent the whole predicate space. Nevertheless, we do not check all predicates of this set, because the complexity is too high. Assuming k represents the number of boolean variables used to represent the propositions inside the abstract data space, we obtain 2^{2^k} proof obligations.

To overcome this problem, we use a qualified strategy selecting predicates, which was proposed by Saïdi and Shankar [34]. They proved that it is sufficient for calculating an overapproximation, to do the check for all disjunctions of literals, whereby a literal is a boolean variable or the negation of it. So the complexity can be reduced to at most $3k - 1$ proof obligations. Fig. 13 shows the selection for a set of atomic propositions, which are constructed using two boolean variables.

To sum up, this selection strategy helps to calculate the overapproximation of a predicate more efficiently. For example, to overapproximate a guard G , we build the conjunction of all possible disjunctions of literals that pass the test for G . Afterwards, we simplify the result and convert it into disjunctive normal form.

So far, we have introduced how propositional predicates defined on an infinite data space can be abstracted using a predicate abstraction. For the abstraction of a whole HA, we replicate the idea of Sec. III-D2 of decomposing the HA in its defining SA. Subsequently we abstract each SA independently. In contrast to defining $AbSBY_{HA}^+$, we calculate overapproximations in both guards and action predicates algorithmically.

Fig. 12 shows the abstracted SA *PressureMode* of the Safety Injection System. Inside the original SA *PressureMode* (cf. Fig. 3), the data variable *pressure* is used for guards only. Hence, we transfer actions into the abstract model without changing them. Furthermore, all guards can be precisely expressed using the boolean variables B_1 and B_2 . Accordingly, we are not introducing self-transitions to represent implicit behaviour. Such behaviour must be modeled explicitly only if the abstracted guard is weaker than the original counterpart.

To calculate the abstraction of the SA *Measuring*, we need to abstract all actions of the self-transitions (cf. Fig. 3). The only part of an action, that involves data variables, is

the update-function. Consequently, we have to abstract the update-functions *INCpressure* and *DECpressure*. Therefore, we interpret an update-function as a binary predicate defined over the pre and post state. Following common notational conventions, we write the post state of (boolean) variable B_1 as B_1' . Unfortunately in this particular example, we are not able to determine a precise overapproximation of the update-functions because the preceded guards give no information on the data variable *pressure*. Consequently, incrementing or decrementing *pressure* maps into the whole abstract data space. The only effect, which we omit here, is $B_1' \wedge \neg B_2'$ because this configuration is obviously not satisfiable. Accordingly, the abstraction of an update-function results in the following predicate.

$$(B_1' \wedge B_2') \vee (\neg B_1' \wedge B_2') \vee (\neg B_1' \wedge \neg B_2')$$

The predicate is a disjunction with three disjoint parts. We represent each part as an abstract update-functions. Note, that the notation of Statecharts allows one update-function per transition only. Consequently, we have to duplicate the self-transitions in the example obtaining three abstracted self-transitions for each (cf. Fig. 14).

V. CONCLUSIONS

We have presented here an approach to model checking of Statecharts with data that combines a full formalisation of the original Statmate semantics of Statecharts in Isabelle/HOL with abstraction techniques, also formalised in Isabelle/HOL, to finally check the abstracted Statecharts in the model checker SMV and additionally use an integration with the SVC validity checker to solve proof obligations resulting from the abstraction process. The main issue of this paper is to give an overall impression of this project that is the core of Steffen Helke's PhD thesis [18].

We finalise this paper with a few concluding remarks concerning the presented parts. The formalisation of the syntax and semantics of Statecharts in Isabelle/HOL has only been reported on partially in this paper; the semantics has been left out. We only wanted to give a gist of the level of explicitness we have used to represent Statecharts in Higher Order Logic. We used type definitions for SA and HA thereby making wellformedness conditions implicit. More importantly, we used explicit polymorphic HOL types to represent the data contained in a Statechart. This makes the representation very concise and also very efficient. Since the data types of our object (the Statecharts) are (generic data) types of the logic HOL, we can exploit a lot of the existing proof infrastructure for proving about Statecharts. On the other hand, explicit formalisations about types become quite tricky. We have, however, illustrated in this work that it is just about possible to formalise notions of partitions of data space in this model.

At the same time, the above described – slightly “shallow” – embedding of data containing Statecharts, is ideally suited for the design of automated tactics. We have here, thus, presented the abstraction techniques we employ to use the type information efficiently to produce proof obligations rising from abstracting Statecharts. Although fairly shallow, the formalisation of Statecharts is deep enough to enable meta theoretical proofs. We have omitted those as they are not in

the centre of interest here.

Finally, we showed how we integrate, on the practical side, model checkers and validity checkers to round off the verification process.

In our current research we try to further extend the practicality of model checking Statecharts. Two current projects that show the emphasis of our current and future work are a front-end tool for lay-outing Statecharts before they are fed into the verification process and another project build on MDD (*Model Driven Development*) to integrate the tool chain for Statecharts.

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Normal form (2^{2^k})

$B_1 \wedge B_2$	$B_1 \vee (\neg B_1 \wedge B_2)$
$\neg B_1 \wedge B_2$	$B_1 \vee (\neg B_1 \wedge \neg B_2)$
$\neg B_1 \wedge \neg B_2$	$B_2 \vee (\neg B_1 \wedge \neg B_2)$
$B_1 \wedge \neg B_2$	$B_2 \vee (B_1 \wedge \neg B_2)$
$B_1 \vee B_2$	$(B_1 \wedge \neg B_2) \vee (\neg B_1 \wedge B_2)$
$\neg B_1 \vee B_2$	$(B_1 \wedge B_2) \vee (\neg B_1 \wedge \neg B_2)$
$\neg B_1 \vee \neg B_2$	$(B_1 \wedge \neg B_2) \vee (\neg B_1 \wedge \neg B_2)$
$B_1 \vee \neg B_2$	$(\neg B_1 \wedge B_2) \vee (\neg B_1 \wedge \neg B_2)$

Approximation ($3^k - 1$)

$B_1 \vee B_2$	B_1
$\neg B_1 \vee B_2$	B_2
$\neg B_1 \vee \neg B_2$	$\neg B_1$
$B_1 \vee \neg B_2$	$\neg B_2$

Fig. 13: Selection strategy for $k = 2$

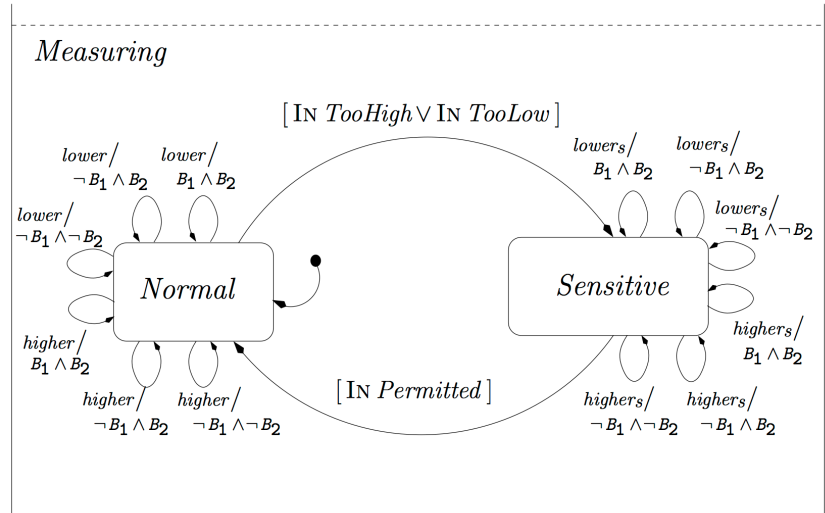


Fig. 14: Abstracted specification of the SA Measuring

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Weighted Unsupervised Learning for 3D Object Detection

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Abstract—This paper introduces a novel weighted unsupervised learning for object detection using an RGB-D camera. This technique is feasible for detecting the moving objects in the noisy environments that are captured by an RGB-D camera. The main contribution of this paper is a real-time algorithm for detecting each object using weighted clustering as a separate cluster. In a preprocessing step, the algorithm calculates the pose 3D position X, Y, Z and RGB color of each data point and then it calculates each data point's normal vector using the point's neighbor. After preprocessing, our algorithm calculates k-weights for each data point; each weight indicates membership. Resulting in clustered objects of the scene.

Keywords—Weighted Unsupervised Learning, Object Detection, RGB-D camera, Kinect

I. INTRODUCTION

Object detection for unlabeled and unsegmented data points [1] is widely studied. In general, visualization and machine learning are the main issues, which are reviewed in existing studies. Machine learning is classified into two categories; supervised and unsupervised learning. In the first case, many researches have addressed the object detection with supervised methods [2], [3], [4], [5], [6], [7], [8]. Kevin Lai's and many other researchers work on the weighted supervised learning for object detection [9], [10], [11], [12] using a hierarchical, multi views, and sparse distance learning. That method can be useful for known objects intended to be detected. Therefore, his algorithm is used for object detection of a specific item that is stored in a database in a preprocessing step such as an apple, egg, etc. Therefore, when using supervised learning for object detection, researchers focus on the accuracy of object detection and detecting known objects. On the other hand, time is very critical in real-time object detection techniques.

The second category addresses the object detection problem with unsupervised learning method [2], [13], [14], [15], [16], [17], [18], [19], [20], [21] using techniques of unsupervised learning such as K-means and spectral clustering. Most of these techniques are not sufficient for real-time applications since they do not address time complexity and memory consumption. With regard to accuracy of object detection using an RGB-D camera, an efficient method for RGB-D camera is weighted clustering since capturing by this kind of camera has more noise introduced by moving objects; thus, labels need to be updated in a few iterations that span less than a second. If

we want to compare the clustering between computer graphics and other domains; data points are not changing during running time in most fields such as data mining, but data points in real-time object detection in the field of computer graphics, machine vision, and robotics are continuously changing; frame by frame and second by second. As regards to Liefeng Bo [14] who uses a dictionary for his work, this technique is very efficient for synchronized video, and also this method is fast enough for video processing with around 2 frames per second (FPS). But, in real-time applications, it needs to be faster than 2 FPS. Weighted Unsupervised Learning such as weighted k-means [22] or many other methods [23], [24], [25], [26], [29], [30] are implemented in different domains. Fuzzy object detection and weighted clustering is addressed and used for moving objects [15], [27], [28]. In 2010, Maddalena et al. [15] had worked on fuzzy logics and learning. That work used only pixel-by-pixel as their features, which can be very efficient for image processing. Thus, those methods never use other sensors such as depth information as a specific feature for 3D, whereas Maddalena's method has many limitations for indoor capturing for 3D object detection. Some other researches use Vicon system as object detection and object controlling for real-time applications, but this device cannot capture color and surface of the objects [31], [32], [33], [34], [35]; thus, Vicon cannot be implemented as a colorful application; however, it can be an efficient method for rigid object detection that could calculate object position as the only feature. Therefore, an unsupervised and RGB-D camera method that addresses the accuracy, time-complexity, and memory consumption and colorful surface capturing simultaneously is unprecedented.

Visualization is an important step in object detection for evaluating the results. Graphical Processing Unit (GPU) based application is a powerful method in the computer graphic domain, but hence not applicable in mobility applications. In general, GPU-based application needs powerful hardware; therefore, for mobility application, researchers focus more on Central Processing Unit (CPU) based applications.

This paper is an extension of the authors' prior work in [13] where we used RGB-D camera and studied the Boolean version of clustering. The color space from RGB was improved to Hunter-Lab color space [36], [37] where the Hunter-Lab color space gave smoother result of clustering in real-time application. In this paper, we use the RGB color space. We improved the clustering part by adding k weights to each

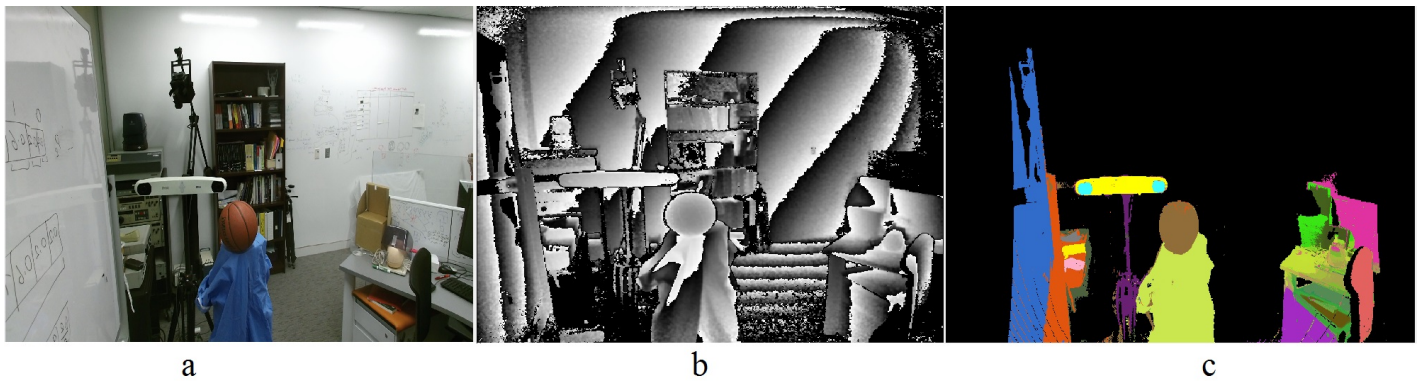


Fig. 1: a) Kinect color frame (RGB) with resolution of 1920 X 1080; b) Kinect depth frame with resolution of 512 X 424; c) Proposed method object detection using $k= 15$ clusters, and after 15 iterations.

data point. This improvement affects frame rate, accuracy, and memory consumption in the real-time application. We propose a real-time object detection algorithm using k weighted clusters with memory consumption that is useful for mobility application. The maximum memory needed for this algorithm is 650 MB for 50 clusters, less than one GB for 100 clusters, and we use multi-thread processing to improve frame rate.

In short, new contributions and unique features of the proposed method in this paper are as follows. 1. Weighted unsupervised learning is presented, which reduces noise for moving and small objects, has better time complexity, lower memory consumption, and higher accuracy than previous methods, 2. CPU-based implementation is offered to make our method capable of mobility usage, and 3. A segmentation technique is proposed to detect a user defined object.

II. PIPELINE AND METHODS

The pipeline of weighed unsupervised object detection algorithm presented in this paper is illustrated in Figure 2 and composed of 8 steps as follows:

- 1) Capturing RGB color and depth information;
- 2) Mapping RGB and depth information;
- 3) Applying back-projection for generating cloud points in the 3D world coordinate system;
- 4) Calculating data points normal vectors based on each point neighbors;
- 5) Segmenting and removing the background to limit the area where we want to detect the objects;
- 6) Calculating distance between each point and the k -centers of the clusters;
- 7) Updating the k -weights of each cloud point; and
- 8) Assigning color to each data point by using the point's k -weights.

This paper is organized as follows: preprocessing is presented in section II-A followed by clustering step in section II-B. Section II-C talks about visualization. Finally, numerical and experimental results are presented in section III. Followed by discussion in section IV and finally the conclusion and future work is in section V.

A. Preprocessing

Preprocessing is used for generating the 3D cloud points from input data that is captured by an RGB-D camera and needed for clustering steps; thus, in preprocessing steps, the algorithm generates cloud points by pose-the 3D as XYZ, color as RGB, and normal as n_x, n_y, n_z . In this research paper, we use Kinect for Windows V2 as an RGB-D camera. In short, preprocessing steps are: get input, perform frames mapping, back projection, and normal calculation.

1) *Get Input*: The Kinect camera was designed as a hands-free game controller. It has two input sensors, which include an RGB camera with a resolution of 1920 X 1080 pixels, and a depth sensor with the resolution of 512 X 424 pixels. Field of View (FOV) is 84.1 X 53.8 for RGB color space and 70.6 X 60 degree for depth sensor information. The resulting average is about 22 X 20 pixels per degree for RGB and 7 X 7 pixels per degree of depth data [40], [41]. Kinect can capture depth information of objects displaced up to 4.5-5 meters from the camera, but we can limit it manually between a and b meters for indoor object detection [38], [39].

2) *Frames Mapping*: The two inputs' frames of Kinect have dissimilar resolutions. The mapping is needed for matching color space and depth information in the same space. Our approach is like other studies [42], [43] for generating colored cloud points. After mapping, we have aligned frames with information about the position x, y, z ; and color R, G, and B for each data point. These data points will be ready for the next step, back-projection [42], [43].

3) *Back Projection*: In this step we convert the cloud points into 3D world coordinate system. The Kinect input from mapping step includes x, y , and z parameters. We follow equations 10 through 14 that are demonstrated in the appendix to back-project the points into the world coordinate System. In summary, to generate an accurate 3D position of each data point in the color frame, the 2D position is back-projected using the depth data from the depth frame as z and the Kinect camera intrinsic parameters to obtain the correct 3D position of each cloud point in the real world coordinate system.

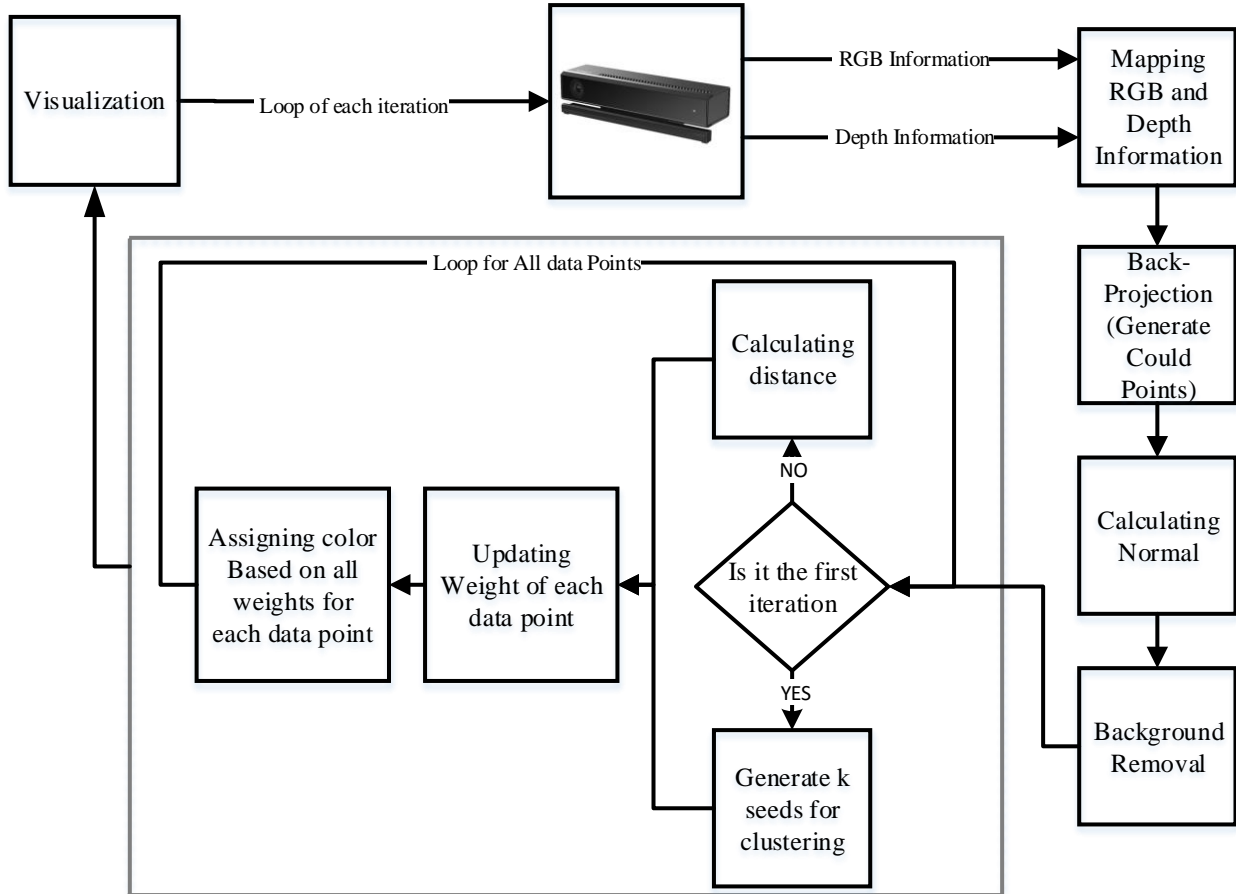


Fig. 2: Pipeline of 3D Object detection using RGB-D camera has two main parts: 1) Preprocessing including Mapping, Back-Projection, Normal Generating, Background removal and 2) Clustering including assigned initial weight, distance calculation, update weight and assign color, and finally visualization to illustrate the results.

4) *Normal Calculation*: In this step, normal vector of each cloud point is calculated as new feature indicated by n_x, n_y, n_z . The normal of each data point is calculated using its neighbors.

$$p_i = (x \ y \ z \ r \ g \ b \ n_x \ n_y \ n_z)^T. \quad (1)$$

B. Clustering step

The aim of this paper is object detection using weighted unsupervised learning. In our approach, we use k-means clustering algorithm with k-weights for each data point, where k is the number of clusters that is defined by use. The clustering step is divided into two main parts; initial seeding, and updating the weights. First part in the first iteration is initialization of k-seeds for the k-means algorithm, and initialization of k-weights for each data point. We use k-means++ [44], [49] algorithm to obtain the initial seeds of the k-means clustering, while the k-weights for each data point are initialized to zero. The second part is updating the k-weights [22]. Weights are going to be updated at each of the following iterations. At each iteration, each data point will have k values which indicate a membership

for one of the k clusters. At the end of each iteration, each point will belong to the cluster that has the highest weight. The algorithm details are given in equation 1, where C_i is the color of each cluster that is assigned at the beginning as unique color for all clusters. The μ_i is k weights of each point, and will continually be changing by each iteration and getting new values. Clustering step includes the labeling, updating the k weights for each data point, and finally visualization is utilized to illustrate the results.

1) *Labeling*: In the adopted weighted unsupervised learning, each data point has a k-values. At beginning of the running time, all weights (μ_p) values are equal to zero for all data points. After the first iteration, the algorithm starts to update the k weights of each data point using the formula in equation 2. According to equation 2, δ_i is the previous weight of the cluster i resulting from the previous frame, which will be updated using scale of weight Ψ to update all memberships.

$$\delta_i = \delta_i + \Psi \text{ where } 0 < \Psi \leq 1. \quad (2)$$

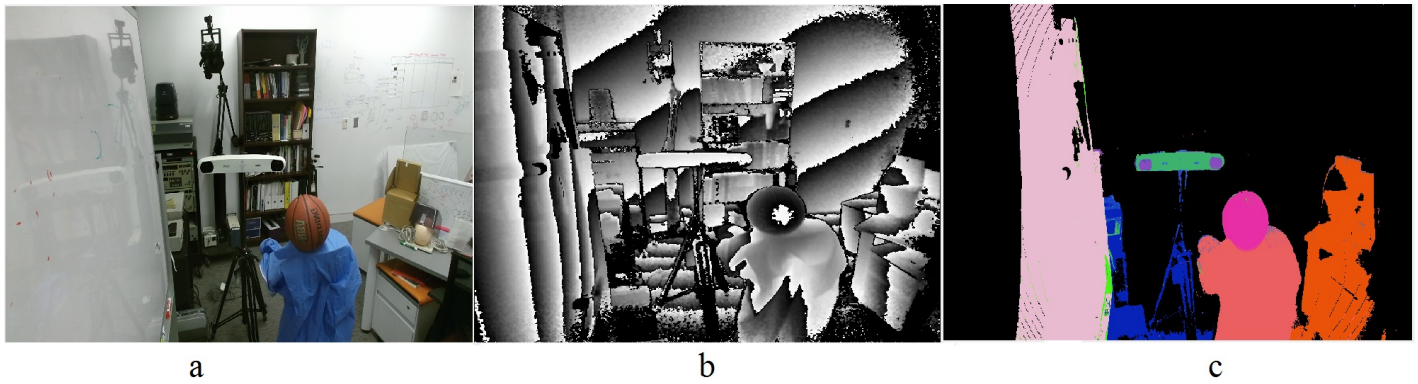


Fig. 3: a) Kinect color frame (RGB) with resolution of 1920 X 1080; b) Kinect depth frame with resolution of 512 X 424; c) Proposed method object detection using k= 7 clusters, and after 10 iterations. Memory consumption is 320 MB and frame rate is 8.1 ± 0.2 FPS.

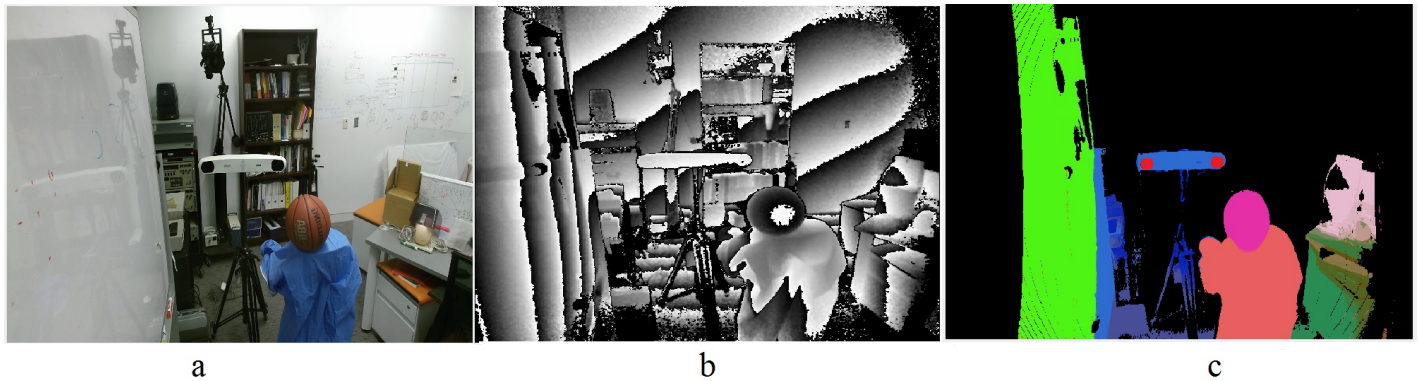


Fig. 4: a) Kinect color frame (RGB) with resolution of 1920 X 1080; b) Kinect depth frame with resolution of 512 X 424; c) Proposed method object detection using k=10 clusters, and after 10 iterations. Memory consumption is 340 MB and frame rate is 6.5 ± 0.2 FPS.

For each iteration, we update the weights of a data point by equations 2 and 3 where δ_i is weigh before normalization, and τ is the number of iterations. The increasing rate of τ depends on frame rate that is addressed in section III and Figure 7. Therefore, iteration number is increasing around fps_t per second.

$$\tau = \sum_{s=1}^t fps_t \quad (3)$$

Equation 4 presents a distance function of the clustering step, Euclidean distance between data points is used as similarity measure in many clustering algorithms including k-means. To consider the color difference of these points while clustering the data point of each frame, the RGB value is incorporated in the Euclidean distance between any two point's v_i and v_c as addressed in the following equation 4, where the scales α and δ insure that geometric distance and the color distance between two points are in the same order of magnitude. Experimentally, the best value of α is between

0.002 and 0.1. The scale of position is denoted by δ that is calculated from α using equation 5. By this equation, we define our new hybrid similarity measure as a function f of two terms; one in the Euclidean distance, $dist$, between two points v_i and v_c which are data point and centroid information, respectively.

$$dist(v_i, v_c) = \sqrt{\left(\delta^2 \left(\begin{matrix} (X_i - X_c)^2 + \\ (Y_i - Y_c)^2 + \\ (Z_i - Z_c)^2 \end{matrix} \right) + \alpha^2 \left(\begin{matrix} (R_i - R_c)^2 + \\ (G_i - G_c)^2 + \\ (B_i - B_c)^2 \end{matrix} \right) \right)} \quad (4)$$

$$\delta + \alpha \leq 1 \text{ where } 0 < \alpha < 1. \quad (5)$$

After calculating the distance between a point and a cluster centroid using their position and color as two different features, we need to incorporate the effect of normal vectors difference between them; thus, for the Euclidean distance, $dist$, between

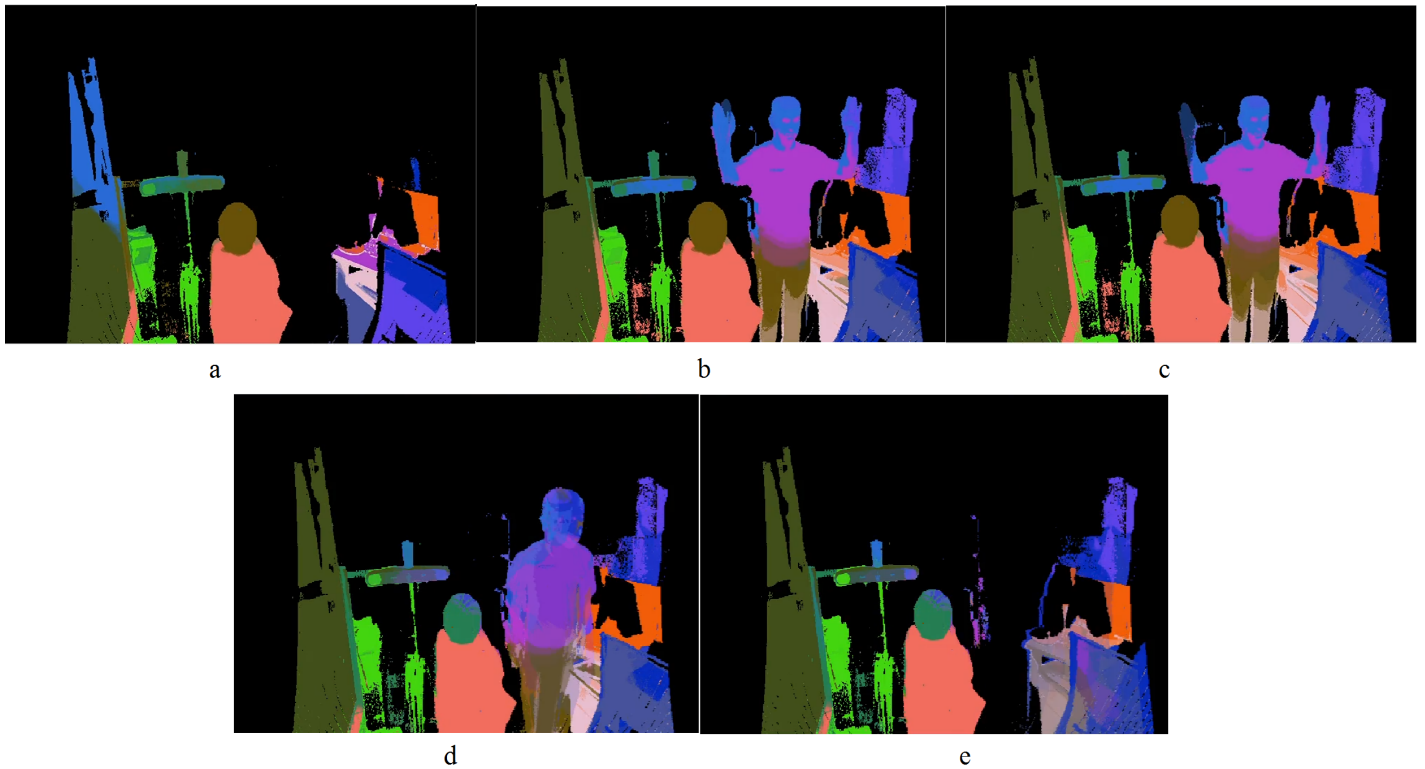


Fig. 5: a) Shows clustered output after 10 iterations and using k=13 clusters; b) Shows a moving object entering the scene. After several iterations the algorithm succeeded in distinguishing the moving object as new object; c) Shows moving hand, where the algorithm successfully detected the moving object; d) Shows a moving object going out of the scene and how the weight of that object was reduced only after one iteration, and finally; e) Shows clustered output after several iterations as detected objects.

two points v_i and v_c which denoted each data point and centroid respectively. The angle between normal of the centroid and normal of each data point is denoted by θ . γ indicates the scale of normal for distance function. The best value of scale of normal, γ , in our experiment was found to be between 0.0001 and 0.01. The final similarity measure between the data points and the centroids that examine the similarity between their positions, colors and normal vectors is given by the following equation 6.

$$f(v_i, v_c) = dist(v_i, v_c) + \gamma(1 - \cos(\theta(n_i - n_c))). \quad (6)$$

2) *Update weight*: the algorithm updates the weights according to equation 2 and normalizes them for each data point according to equations 7 and 8. Then, the algorithm assigns each data point the label of the cluster with the highest weight. With regard to equations 7 and 8, (μ_p^i) denotes the weight of cluster i of data point p and τ is the number of iterations which are calculated by the equation 3. The summation of k weights of each data point after normalization could be less than or equal to one.

$$\mu_p = \frac{\sum_{i=0}^{\tau} (\delta_i)}{\|\sum_{i=0}^{\tau} (\delta_i)\|}, \quad (7)$$

where

$$\sum_{i=0}^k \mu_p^i \leq 1. \quad (8)$$

C. Visualization

After calculating all k weights for all data points at each iteration, we illustrate the results as an object detection or segmentation by assigning each cluster a unique color C_i . Equation 9 indicates assigning one of the k colors to each data point based on its weight μ . According to equation 8, the summation of all weights for data point is less than or equal to one, where μ is the weight of each cluster in a data point. That means, if we have a data point with k different weighted labels, the color of it is assigned by following equation:

$$C_p = \sum_{i=0}^k (C_i * \mu_i) \quad (9)$$

As regards to Algorithm 1, it has one main loop that contains the iteration of our system, the second loop contains clustering step that calculate and update weights. For each data points, the algorithm calculates its distance with respect to all centroids using RGB as color space, XYZ as points point position and n_x, n_y, n_z as normal vector. After that algorithm updates the k weights of each data points after each iteration. Initially in the first iteration, k seeds are obtained by k-means++ and all weights labels are initialized equal to zero. For color assignment, all k weight of each point are used to give the point a label.



Fig. 6: Results of segmenting scene objects using proposed algorithm; *a*) Segmentation of small duck; *b*) Segmentation and detection of piece of red paper; *c*) Object detection of a box; *d*) Shows handy bag; *e*) Segmentation of box, the border of the box has lower weight and it will be completed after several iteration; *f*) Representation of moving object, segmentation of a person; *g*) Segmentation of basketball.

Algorithm 1:

```
while main // This is main loop that is starting at the beginning and each iteration
    representing one frame
do
    while all of data points // this loop represents the number of data points we have in
        each frame
    do
        if Is it the first iteration // for the first iteration only we need to initialize the
            seeds
        then
            Assign centroid by using K-means++ // calculating the all centroid using K-means++
            Assign first weighted using regular labels
            // calculating the first label for each data point
        else
            while all of clusters // this loop start from 1 to k
            do

$$dist(v_i, v_c) = \sqrt{\left( \delta^2 \begin{pmatrix} (X_i - X_c)^2 + \\ (Y_i - Y_c)^2 + \\ (Z_i - Z_c)^2 \end{pmatrix} + \alpha^2 \begin{pmatrix} (R_i - R_c)^2 + \\ (G_i - G_c)^2 + \\ (B_i - B_c)^2 \end{pmatrix} \right)}$$

                /* calculating distance between each data point and centroid using
                    position and color of each data point */

$$f(v_i, v_c) = dist(v_i, v_c) + \gamma(1 - \cos(\theta(n_i - n_c)))$$

                /* incorporate the differences between angle of normal vectors of each
                    data point and centroid */
                if  $f(v_i) > f(V_{i+1})$  // condition of distance between previous frame and current
                    frame
                then
                    update labeled weights  $\delta_i = \delta_i + \Psi$  where  $0 < \Psi \leq 1$ 
                    /* updating the weight of data point by scale of weights (one in
                        our experiment) */

$$\mu_p = \frac{\sum_{i=0}^{\tau} (\delta_i)}{\|\sum_{i=0}^{\tau} (\delta_i)\|}$$


$$\sum_{i=0}^k \mu_p^i \leq 1$$

                    /* normalize all weights for each data point where the summation of
                        all weights should be less or equal than one */
            while all of data points // assigning color to each data point for visualization
            do

$$C_p = \sum_{i=0}^k (c_i * \mu_i)$$


```

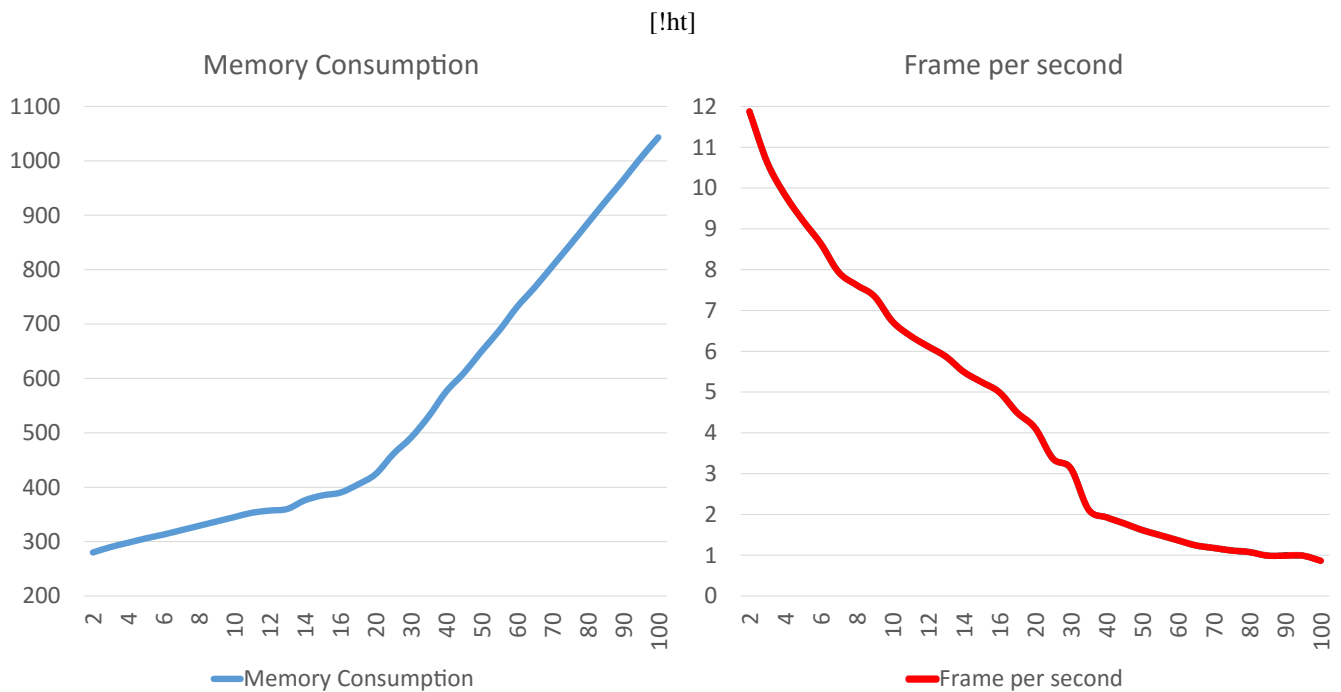


Fig. 7: Left: Chart indicates the memory consumption for different number of clusters where x-axis represents the number of clusters, k, value and y-axis show the memory consumption in MB. The larger the k value the more memory needed to process all scene points. Right: The frame rate is given in the right chart where x-axis is number of clusters, k, and y-axis is frame per second. The larger the k value the less frames per second evaluated.

III. RESULTS

We test our algorithm using different number of clusters, k, and evaluate memory consumption and frame rate. Figure 7 indicates the frame rate experiments with k between 2 and 100 along with the memory consumption of each experiment. The frame rate and memory consumption are tightly affected by the number of clusters. When we have large number of clusters, the algorithm needs more memory, and frame rate will be reduced. Figure 3 shows the result of weighted supervised learning with k = 7 clusters after several iterations; the memory consumption for seven clusters is 322 MB, and frame rate is 7 ± 0.2 frame per second. The figure 4 shows the result of k=10 clusters where memory consumption is 344 MB, and frame rate is 6.5 ± 0.2 frame per second. The algorithm implementation consumes multi-thread programming in Visual Studio 2015, to implement parallel processing for the preprocessing and clustering parts. All of the loops use the parallel computational model introduced by Microsoft API [45] assign including for each data point back-projection, normal calculation, mapping, and assigning color along with calculating the weights of each data point [46]. The used hardware running our algorithm plays another factor for evaluating the system. The used CPU is capable of multi-thread programming and parallel processing, which is dual Xeon E5 family 2.29 GHz speed. It has 12 core and 24 logical processors. We do not use GPU in this application, but the GPU of the system is Nvidia Quadro K5000 with 4 GB GDDR5 speed [47] with 32 GB memory and having a speed of 1333 MHz. In addition, we use Universal Serial Bus (USB) version 3.0 for Kinect connection. Our input camera is Kinect V2 for Windows [39].

IV. DISCUSSION

With regard to our results, this algorithm provides a unique output that can be useful for researchers in computer graphics, computer vision, robotics, and other related fields. The weighted unsupervised learning for object detection proposed in this paper shows its capabilities to the smooth the output in noisy environments in real-time producing scene objects. The results shown in Figure 7 indicate that this model can be real-time and has the capability for mobility applications with low memory consumption without the need for an expensive GPU. Using three features in clustering step, position, color and normal, gives us the capability to group similar data points to objects with accurate results. Our results show that the use of the weight to indicate the membership for each data point to a cluster and then for object detection is sufficient for detecting moving objects in noise environment, which was captured by an RGB-D camera.

V. CONCLUSION AND FUTURE WORK

In this paper, we proposed a weighted unsupervised learning for object detection framework using a single RGB-D camera. In the proposed method, we use CPU-based programming and multi-thread processing. The results are provided in real-time. In preprocessing step, we generate 3D data points, and after preprocessing, clustering step is applied by initialization of seeds and updating the weights of each data point. Finally assigning colors is used to illustrate object detection results. Another distinct contribution of this paper is segmentation of a particular object, particularly for moving

objects that can be updated by each frame. The frame rate of this work is between 1 and 12 frame per second, and memory consumption of this algorithm is between 280 MB and 1 GB for different values of clusters. This algorithm can be applied in any object detection and segmentation application in the fields of computer graphics, robotics, vision, or for surveillance and object controlling. Future directions of this work can be summarized in extending this method to GPU based programming, increasing the performance and frame rate simultaneously.

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VI. APPENDIX

Converting the projective 3D position to real world with respect to camera is done by following equations 10, 11, 12, 13, and 14, where field of view is denoted by fov , W_x , W_y and W_z is world coordinate system of X, Y, and Z respectively, and P_x , P_y , and P_z shows the projection parameters.

$$scale_x = 2 * \tan\left(\frac{fov_x}{2}\right) \quad (10)$$

$$Scale_y = 2 * \tan\left(\frac{fov_y}{2}\right) \quad (11)$$

$$W_x = P_z * Scale_x * \left(\frac{P_x}{R_w} - 0.5\right) \quad (12)$$

$$W_y = P_z * Scale_y * \left(\frac{P_y}{R_w} - 0.5\right) \quad (13)$$

$$W_z = P_z \quad (14)$$

fov_x for Kinect 1 is equal to 1.014468 and fov_y is equal to 0.7898094 [48], and in many researches, people use these numbers that reduced the field of view, but as new resolution of Kinect V2 we use 1.22173047 for fov_x and 1.0471975511 by changing degree to radian $Radian = \frac{Degree * \pi}{180}$. The weight of each data point can be equal or bigger than zero, $\mu_i > 0$ where $i \in [0, k - 1]$ and k is the number of clusters. The distance which is calculated in equation 4 and 6 and shown as $f(v_i, v_c)$, can be zero if and only if centroid and data point i are equal to each other, $i = c$. The scale of color and position as shown δ and α , both are positive number and cannot be zero, and as we mention to that, summation of them is less than one. All results are calculated by sampling of one, so if we increase the sampling to more than one our resolution, and memory consumption will decrease and frame rate will be increased.

A Novel Approach for On-road Vehicle Detection and Tracking

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Abstract—On the basis of a necessary development of the road safety, vision-based vehicle detection techniques have gained an important amount of attention. This work presents a novel vehicle detection and tracking approach, and structured based on a vehicle detection process starting from, images or video data acquired from sensors installed on board of the vehicle, to vehicle detection and tracking. The features of the vehicle are extracted by the proposed GIST image processing algorithm, and recognized by the state-of-art Support Vectors Machine classifier. The tracking process was performed based on edge features matching approach. The Kalman filter was used to correct the measurements. Extensive experiments were carried out on real image data validate that it is promising to employ the proposed approach for on road vehicle detection and tracking.

Keywords—Vehicle detection; Vehicle tracking; GIST; SVM; Edge features; Kalman filter

I. INTRODUCTION

Advanced Driver Assistance Systems (ADAS) play an important role in enhancing car safety and driving comfort. One of the most important difficulties that ADAS face is the understanding of the environment and guidance of the vehicles in real outdoor scenes. These systems aim to alert a driver about driving environments, possible collision with other vehicles, or take control of the vehicle to enable collision avoidance and mitigation. Vehicle accident statistics disclose that the main threats drivers are facing are from other vehicles. Consequently, robust and reliable vehicle detection system is a required critical and important task not only for ADAS, but also for other real-world applications including urban scene understanding, automated driving, self-guided vehicles, etc. The most common approach to vehicle detection is using active sensors such as lidar, millimeter wave radars, or lasers [1]. Prototype vehicles employing active sensors have shown promising results. However, active sensors have several drawbacks, such as low spatial resolution, slow scanning speed, and are rather expensive. Moreover, when there is a large number of vehicles moving simultaneously in the same direction, these sensors may interfere with each other, which can cause many problems. On the other hand, passive sensors, such as optical cameras, offer a more affordable solution and can be used to track more effectively cars entering a curve or moving from one side of the road to another. Moreover, visual information can be very important in a number of related applications,

such as lane detection, traffic sign recognition, or object identification (e.g., pedestrians, obstacles), without requiring any modifications to road infrastructures.

In general, vehicle detection using optical sensors is a very challenging task due to several factors. For example, vehicles come into the view with different speeds and may vary in shape, size, and color Fig. 1(a). Air pollution and weather conditions (e.g., rain, snow, fog, shadows, and clouds) may affect the visibility of vehicles Fig. 1 (b), In addition, its appearance depends on its pose and can be affected by obstacles, such as other vehicles, pedestrians, etc. Fig. 1 (c), Outdoor lighting conditions varying from day to night may also affect the visibility on the road scenes Fig. 1(d). Moreover, real-time constraints make this task even more challenging.

Almost every visual vehicle detection system follows two basic steps: Hypothesis Generation (HG), which hypothesizes the locations in images, where vehicles might be present. Hypothesis Verification (HV), which verifies the hypothesis.

In this paper, we present a new vehicle detection and tracking approach using vision-based techniques. For the detection, first we refer to hypothesis generation and verification using the sliding window and GIST as a features descriptor along with a well trained linear SVM. For the tracking, we refer to an edge based features matching method. The rest of the paper is organized as follows. Section 2 presents an overview of past work on vehicle detection and tracking. Section 3 details the proposed approach to vehicle detection and tracking. Experimental results are illustrated in Section 4. Section 5 concludes the paper.

II. RELATED WORK

Many different approaches to vehicle detection and tracking have been proposed and it is difficult to compare between these approaches since they are based on different data. Regarding the detection problem, different approaches have been proposed. In the older studies, e.g. [2], [3], as well as in many recent ones, e.g. [4], [5], authors refer to monocular vision or stereo-vision based techniques.

For the monocular vision based techniques, the use of some features to detect vehicles is widely employed in the literature. For example, Histogram of oriented gradient (HOG) features [6] have been used in a number of studies [7]. In [8], the symmetry of the HOG features extracted in a given image

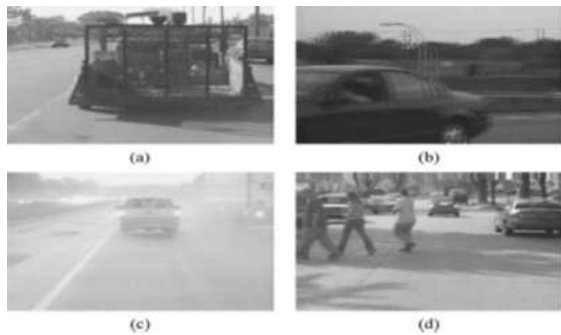


Fig. 1: Examples for difficulties facing the vehicle detection task.

patch, along with the HOG features themselves, was used for vehicle detection. Haar-like features have also been used for vehicle detection in a number of studies [9]–[14]. Haarlike features are popular for two main reasons. First, Haar-like features are well suited to the detection of horizontal, vertical, and symmetric structures. Second, by using the integral image, feature extraction is very fast. While studies that use either HOG or Haar-like features comprise a large portion of recent vehicle detection works, other general image features have been used. In [5], a combination of SURF and edge features are used to detect vehicles, with vehicle parts identified by keypoint detection. In [15], vehicles are detected as a combination of parts, using SIFT features and hidden Conditional Random Field classification. In [5], a combination of speeded-up robust features and edges is used to detect vehicles in the blind spot. In [16] Gabor and Haar features were used for vehicle detection. These features are feed to the classifiers to perform the vehicle detection process. Support vector machines (SVMs) have been widely used with HOG features for vehicle detection [7], [9]. The HOG-SVM formulation was extended to detect and calculate vehicle orientation using multiplicative kernels in [17]. In [9], [18], artificial neural networks were used to classify extracted features for vehicle detection. The combination of Haar-like feature extraction and Adaboost classification has been used to detect rear faces of vehicles in [11], [19], [20]. In [21], Waldboost was used to train the vehicle detector. Motion-based approaches, which require a sequence of images in order to recognize vehicles, are also employed to detect vehicles. In general, these methods has been less common than features-based methods. In [2], [22], adaptive background modelling was used, with vehicles detected based on motion that differentiated them from the background. Optical flow [23], a fundamental machine vision tool, has been used for monocular vehicle detection [24]. In [25], a combination of optical flow and symmetry tracking was used for vehicle detection. In [26], interest points that persisted over long periods were detected as vehicles traveling parallel to the ego vehicle. Ego-motion estimation using optical flow, and integrated detection of vehicles was implemented in [3]. In [27], optical flow was used to detect overtaking vehicles in the blind spot.

On the other hand, in the stereo-vision based techniques, multi-view geometry allows for direct measurement of 3D information, which provides for understanding of scene, motion characteristics, and physical measurements. The ability to track points in 3D, and distinguish moving from static objects, affects the direction of many stereo-vision studies. While monocular vehicle detection often relies on appearance features and machine learning, stereo vehicle detection often relies on motion features, tracking, and filtering. In [28], a histogram of depths, computed from stereo matching, was used to segment out potential vehicles. Clustering was used for object detection in [29]. In [4], clustering was implemented using a modified version of iterative closest point, using polar coordinates to segment objects. In [30], the occupancy grid is populated using motion cues, with particles representing the cells, their probabilities the occupancy, and their velocities estimated for object segmentation and detection.

In the tracking process, many approaches have been used. For example, in [31], the concept of 6D-vision, the tracking of interest points in 3D using Kalman filtering, along with ego-motion compensation, is used to identify moving and static objects in the scene. The authors in [32] have proposed a novel real time traffic supervision approach, which employs optical movement to detect and track vehicles. It uses two new techniques: color contour based matching and gradient based matching. Occupancy grids are widely used in the stereo-vision literature for tracking. In [29], scene tracking and recursive Bayesian filtering are used to populate the occupancy grid each frame, while objects are detected via clustering. In [33], tracked 3D points, using 6D vision, are grouped into an intermediate representation consisting of vertical columns of constant disparity, termed stixels. Stixels are initially formed by computing the free space in the scene, and using the fact that structures of near-constant disparity stand upon the ground plane. The use of the stixel representation considerably reduces the computation expense over tracking all the 6D vision points individually. The tracked stixels are classified as vehicles using probabilistic reasoning and fitting to a cuboid geometric model.

In this work, the vehicle detection process is performed in only one step. Sliding window is processed by a features descriptor and given to a classifier to decide whether they contain vehicles or not. Then, the regions are considered as true vehicles based on a threshold. For vehicle tracking, edge features are used. A matching edge process is used to find corresponding edges between consecutive frames. These correspondences are used to track the vehicle in the next frame.

III. PROPOSED METHOD

The aim of this work is detection and tracking of the on-road vehicles. The realization of this system was motivated by the crucial need of enhancing the people safety while driving on a road, and providing more comfort when they face other vehicles.

A flow chart of the proposed approach is shown in Fig. 2. The first step of the proposed approach is vehicle detection. Sliding window is processed by a features descriptor and given to a classifier to decide whether they contain vehicles or not. The

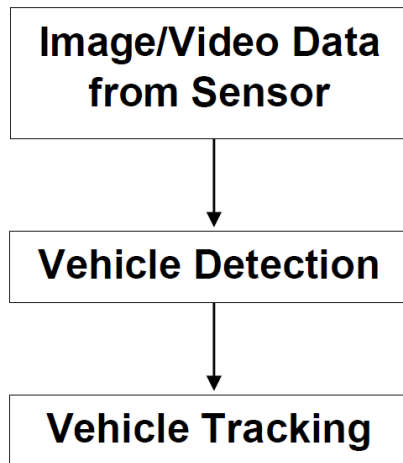


Fig. 2: Flow chart of the proposed approach.

positive windows are classified as true vehicle or false alarms based on a threshold. The second step is vehicle tracking which is based on edge features. Edge curves matching process is applied to find corresponding edges between two consecutive frames. These correspondences are used to find the tracked vehicles in the next frame. Kalman filter is used then to correct the measurements or to predict the new positions if there is no measurements.

A. Vehicle Detection

On-board vehicle detection systems have high computational requirements as they need to process the acquired images at real-time or close to real-time to save time for driver reaction. Searching the whole image to locate potential vehicle locations is prohibitive for real-time applications. The majority of methods reported in the literature follow two basic steps: 1) HG where the locations of possible vehicles in an image are hypothesized and 2) HV where tests are performed to verify the presence of vehicles in an image. Various HG approaches have been proposed in the literature [34]–[36], which can be classified into one of the following three categories:

- 1) knowledge-based: Knowledge-based methods employ a priori knowledge to hypothesize vehicle locations in an image.
- 2) stereo-based: These kind of approaches take advantage of the Inverse Perspective Mapping (IPM) to estimate the locations of vehicles and obstacles in images.
- 3) motion based : Motion-based methods detect vehicles and obstacles using optical flow.

The hypothesized locations from the HG step form the input to the HV step, where tests are performed to verify the correctness of the hypothesis.

In this work, sliding window is used to detect vehicles. The use of the concept sliding window spans a broad variety of domains, such as information technology where it has been widely used in signal processing for analysis of frequent items in packet streams [37]. In informatics, it has been applied to detect a variety of object such as faces, pedestrians, traffic

signs and vehicles etc. Sliding windows have also been applied within the field of natural language processing for collocation detection [38]. We applied the concept of the sliding window to detect relevant information from the on-road images captured by a camera mounted on a car.

The sliding_window function requires three arguments. The first is the image that we are going to loop over. The second argument is the stepSize.

The stepSize indicates how many pixels we are going to 'skip' in both the (x,y) direction. Normally, we would not want to loop over each and every pixel of the image (i.e. stepSize=1) as this would be computationally prohibitive if we were applying an image classifier at each window.

Instead, the stepSize is determined on a per-dataset basis and is tuned to give optimal performance based on your dataset of images. In practice, it's common to use a stepSize of 4 to 8 pixels. Remember, the smaller your step size is, the more windows you will need to examine. The last argument windowSize defines the width and height (in terms of pixels) of the window we are going to extract from the image. Regarding this work we have performed several experiments on the on-road image scenes using different parameters for the sliding window.

stepSize = 4,8,12 in both directions, and we keep skipping the same number of pixels in (x,y) axes for each experiments.

To detect vehicles in multi scale, the window size we have chosen varies between 20 x 20 and 100 x 100.

For more precision we only consider regions where more than 10 detections were found as vehicle. This constraint will reduce the number of false alarms.

We validate the windows based on an analysis of a set of features in the original domain. We propose a set of perceptual dimensions (naturalness, openness, roughness, expansion, ruggedness) that represent the dominant spatial structure of a scene. These dimensions may be reliably estimated using spectral and coarsely localized information (GIST Descriptor) according to [39].

First introduced in [39]. GIST is a low dimensional representation of the scene, which does not require any form of segmentation according to [40].

Intuitively, GIST summarizes the gradient information (scales and orientations) for different parts of an image, which provides a rough description (the gist) of the scene.

The algorithm presents the advantage of being biologically plausible and of having low computational complexity, sharing its low-level features with a model for visual attention that may operate concurrently on a vision system.

Given an input image, a GIST descriptor is computed by:

- Convolve the image with 32 Gabor filters at 4 scales, 8 orientations, producing 32 feature maps of the same size of the input image.
- Divide each feature map into 16 regions (by a 4 x 4 grid), and then average the feature values within each region.
- Concatenate the 16 averaged values of all 32 feature

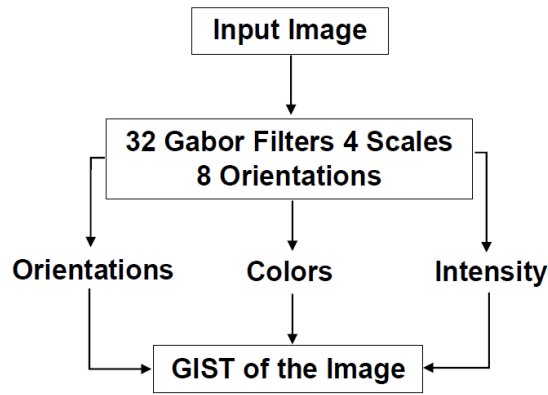


Fig. 3: Framework of GIST.

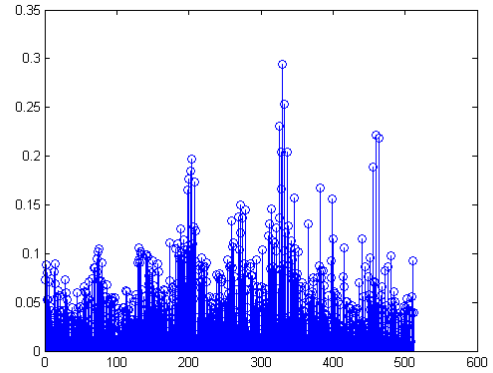


Fig. 5: The GIST histogram of the positive image shown in Fig 4.



Fig. 4: Positive image (left) with its gist descriptor (right).

maps, resulting in a $16 \times 32 = 512$ GIST descriptor.

The Fig. 3 presents a framework of the GIST descriptor and Fig. 4 shows an image from the database with its GIST descriptor.

The GIST descriptor features vector containing 512 values (see Fig. 5) is introduced to a state of the art classifier. Support Vector Machine with linear kernel in a version implemented in Matlab software, and trained by Levenberg Marquardt algorithm.

A new database created in the [41] containing 4000 positive vehicle images and 4000 negative vehicle images is used to train and test the classifiers.

The database consists of images of resolution 64×64 acquired from a vehicle-mounted forward-looking camera. Each image provides a view of the rear of a single vehicle.

A cross-validation procedure is used to test the method. Specifically, 75% of the images are randomly selected for the training set and the remaining 25% are used for the testing set. This process is repeated three times and the average is computed.

B. Vehicle Tracking

In this section, we describe the proposed method for vehicle tracking summarized in Fig. 6. The algorithm is based on edge

features. To extract edges, we had used canny edge operator [42] for the reason that it yields continuous edge curves which are vital to the proposed method and for its detection precision. The algorithm of association proposed in [43] is used to find relationships between consecutive frames. Edge curve matching process is applied based on these relationships to find correspondences between edge curves of two consecutive images. The matched edge curves in an image k of the edge curves of any detected vehicle in the image $k-1$, are used to measure the position of that vehicle in the image k . Kalman filter is used to correct the current position based on the new measurement and the past states.

1) *Edge curves matching process:* Here, we describe the algorithm that matches edge curves between two consecutive images I_{k-1} and I_k . We begin by describing the association algorithm used to find relationship between consecutive images. More details can be found in [43]. Let us consider two edge points P_{k-1}^L and Q_{k-1}^L belonging to the edge $C_{k-1}^{L,i}$ in I_{k-1}^L and their corresponding ones P_k^L and Q_k^L belonging to the edge $C_k^{L,j}$ in I_k^L (see Fig. 7). The associate point to the point Q_k^L is defined as the point belonging the edge $C_{k-1}^{L,j}$ with the same y-coordinate as of Q_k^L (e.g. as shown in Fig. 7 the associate of the point Q_k^L is the point P_{k-1}^L) [43], [44].

Based on these associate edge points computed used the association algorithm described above, we match the edge curves of each two consecutive images. Let $ASS(C_{k-1}^i)$ be the set of associate edge points of the edge curve C_{k-1}^{i-1} . We find the matched edge curve C_k^j of the edge curve C_{k-1}^i by looking for the edge curve which contains the maximum number of edge points in $ASS(C_{k-1}^i)$.

2) *Kalman filter:* In advanced driver assistance systems (ADAS) applications, the fps is high (the time interval between consecutive frames is too small), which allows to assume that the vehicle velocity is constant. In the tracking process, we use the position and the velocity to describe characteristics of

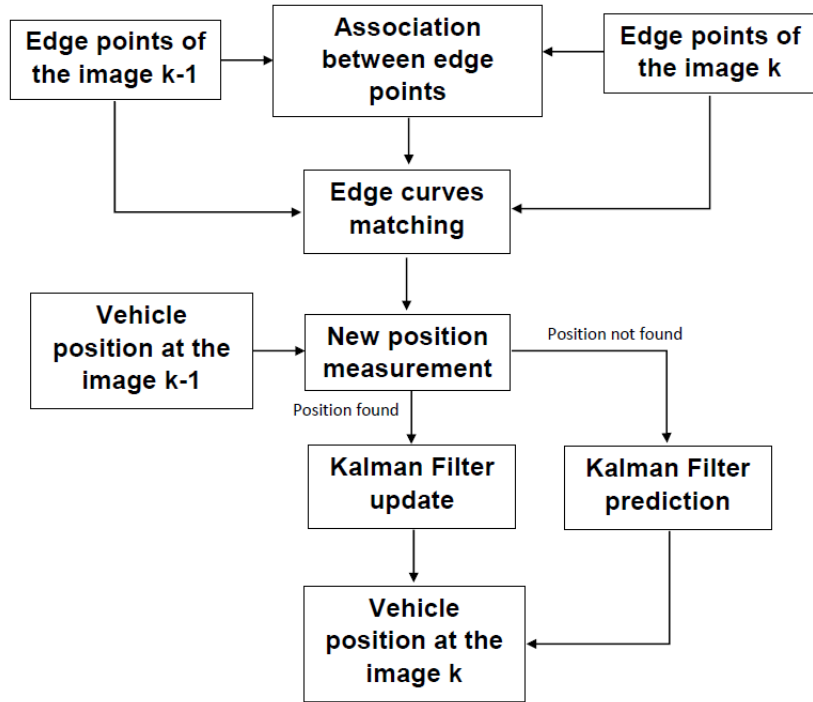


Fig. 6: Main steps of the proposed tracking algorithm.

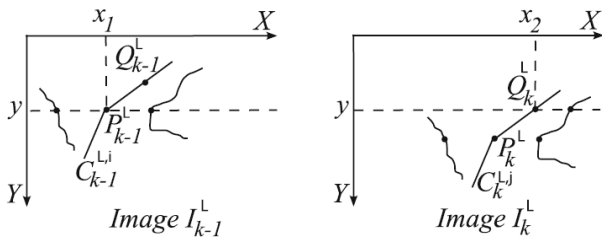


Fig. 7: I_{k-1}^L and I_k^L two consecutive left images. The point P_{k-1}^L in image I_{k-1}^L is the associate of the point Q_k^L in the image I_k^L .

vehicles motion. The system state is defined by the vector:

$$X_k = (x, y, v_x, v_y) \quad (1)$$

The description of the state and the measurement at the step k are defined by the following equations:

$$X_k = F \times X_{k-1} + W_k \quad (2)$$

$$V_k = H_k \times X_k + V_k \quad (3)$$

Which implies that the structure of F is something like:

$$F = \begin{pmatrix} 1 & 0 & dt & 0 \\ 0 & 1 & 0 & dt \\ 0 & 0 & 1 & 0 \\ 0 & 0 & 0 & 1 \end{pmatrix}$$

We measure only the position of the vehicle $z=(x,y)$, which implies that the structure of H is something like:

$$H = \begin{pmatrix} 1 & 0 \\ 0 & 1 \\ 0 & 0 \\ 0 & 0 \end{pmatrix}$$

Kalman tracking process is divided into two steps, prediction and correction. In the step of prediction, the process computes a predicted system state \hat{X} and an error covariance \hat{P} . In the step of correction, the Kalman filter computes the Kalman gain based on error covariance \hat{P} . Then it corrects the state using Kalman gain and the measurement value. In this work, we had used the kalman filter implemented in OPENCV.

3) *Position measurement and update:* The measurement of the new position of any vehicle is performed by looking for the edge curves of that vehicle that have corresponding edge curves in the next image. Let $MEC(V_i)$ be the set of the matched edge curves in the next image of the edge curves of the vehicle V_i . The center of the box containing all the matched edge curves in the set $MEC(V_i)$ is considered as the new position of the vehicle V_i in the next image. The new position is given to the Kalman filter discussed above to update the state. If an observation is unavailable for some reasons (i.e. occlusion), the update may be skipped and multiple prediction steps are performed.

TABLE I: Classification accuracy rates of GIST compared with CR-HOG using SVM in %.

	CR-HOG	GIST
Acc	94.98	96.0

IV. EXPERIMENTAL RESULTS

In this section, we present the results obtained by the new method. We note that the hardware used in our experiments is a HP Pavilion dv3 Intel(R) Core(TM) Duo CPU 2.10GH running under Windows 7.

A. Vehicle Detection

In the following we present results from the classification applied. The database consists of 4000 vehicle images and 4000 non vehicle images of resolution 64×64 acquired from a vehicle-mounted forward-looking camera. Each image provides a view of the rear of a single vehicle. And in order to confirm the performance of the proposed scheme, we compare it with the approach proposed in [41].

The original images are manually labeled to provide the ground truth. The image containing vehicle is labeled as positive sample; otherwise, it is labeled as a negative sample. In order to prevent over-fitting of the classification results, we exploited three-fold cross-validation for all the classification experiments.

An example of each type of images from the database that was used for the experiment is shown in Fig. 8. The accuracy of correctly classified rate of samples is provided in Table I. Table I shows the classification results of the SVM in the original color space. From these results we can conclude that the proposed approach outperforms the [41] scheme and shows a superior performance, with an improvement of 1.02% in term of accuracy.

The results obtained can be explained by:

CR-HOG feature extractor is based on the HOGs that evaluate local histograms of image gradient orientations in a dense grid. The underlying idea is that the local appearance and shape of the objects can often be well characterized by the distribution of the local edge directions, even if the corresponding edge positions are not accurately known. The CR-HOG is a new configuration of HOG composed of concentric rectangular cells. In addition to the distribution of the local edge directions, the GIST descriptor represents the scale information. Consequently, more variability is found in the gradient orientation map. Thus better discrimination ability in the features extractor. The results are improved to an overall accuracy of **97.5%** when using the MLP neural network as a classifier instead of the linear SVM.

This can be explained by the few advantages that MLP over other classifiers [45] such as better generalization ability, robust performance and less training data.

B. Vehicle Tracking

In order to assess its performances, the new method has been tested on an image sequence acquired by a sensor mounted



Fig. 8: Samples from the database.

aboard a moving car, the velocity of the car was 90km per hour and the sensor provides 10 images per second. The images size is 377×286 pixels.

Fig. 9 shows the results obtained by the proposed method on the frames #4132, #4136 and #4140 of the sequence described above. it shows the vehicles detected and tracked over time by the proposed method. We had successfully detected the two vehicles in the scene but we have also some false alarms. The two detected vehicles had been successfully tracked over time. The method treats 13 images per second and the sensor used in our experiments provides 10 images per second which makes the new method a real time method.

V. CONCLUSION

In this work, a novel vehicle detection and tracking method was proposed. This method is robust in the context of real traffic scenes. As regards vehicle detection, the GIST descriptor has been utilized based on the extraction of gradient features (scales and orientations) for different parts of an image. The descriptor has proven to have good discriminating properties using a reduced number of features in a simple linear kernel SVM. Thus resulting in a rough description of the scene and ideally suited for real time applications. For vehicle tracking, a new method based on edge curves matching between consecutive frames was proposed. The matched edge curves between consecutive frames are utilized to find the new vehicle's position in the next frame, then Kalman filter was applied to correct the measurements. In addition to its cost effectiveness, the over all system exhibits a good functionality on real world scenarios. For further studies, more complicated scenes and environmental factors must be considered to make the approach more robust. We will combine the GIST descriptor with an other descriptor such as HOG to improve the detection process. We will also use the feature detector SURF together with the matching process used in this paper to improve the tracking process.

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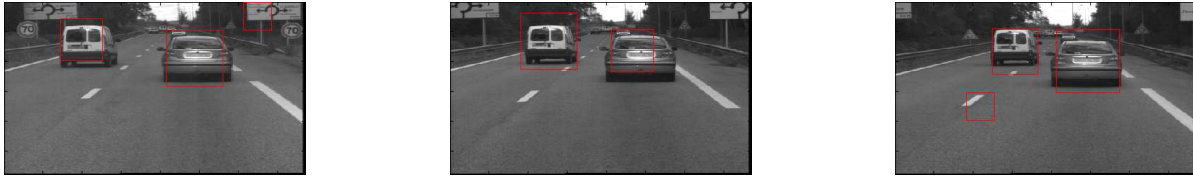


Fig. 9: The results obtained by the proposed method on the frames #4132, #4136 and #4140 of the sequence described above

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A Robust Hash Function Using Cross-Coupled Chaotic Maps with Absolute-Valued Sinusoidal Nonlinearity

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Abstract—This paper presents a compact and effective chaos-based keyed hash function implemented by a cross-coupled topology of chaotic maps, which employs absolute-value of sinusoidal nonlinearity, and offers robust chaotic regions over broad parameter spaces with high degree of randomness through chaoticity measurements using the Lyapunov exponent. Hash function operations involve an initial stage when the chaotic map accepts initial conditions and a hashing stage that accepts input messages and generates the alterable-length hash values. Hashing performances are evaluated in terms of original message condition changes, statistical analyses, and collision analyses. The results of hashing performances show that the mean changed probabilities are very close to 50%, and the mean number of bit changes is also close to a half of hash value lengths. The collision tests reveal the mean absolute difference of each character values for the hash values of 128, 160 and 256 bits are close to the ideal value of 85.43. The proposed keyed hash function enhances the collision resistance, comparing to MD5 and SHA1, and the other complicated chaos-based approaches. An implementation of hash function Android application is demonstrated.

Keywords—Hash Function; Cross-Coupled Chaotic Map; Sinusoidal Nonlinearity; Information security; Authentication

I. INTRODUCTION

The advancement in communication technologies have led to a great demand in reliable and robust information security, involving data confidentiality, verification of data integrity, authentication and non-repudiation of origin [1]. A hash function, which encodes an arbitrary length input message into a hash value with a fixed length, has played an important role in advanced information security, particularly including in cryptography and secure protocol methods. As for security purposes, the desirable performances of hash functions include high possibility of collision resistance and high security against preimage and second-preimage attacks.

The typical MD4, MD5, and SHA-1 hashing algorithms have extensively been realized in software industries for integrity verification of electronically transmitted files as well as security in protocols. Such typical hash functions are designed based on logical operations or multi-round iterations and therefore the hashing process efficiency depends upon inherent

ciphers which necessarily require complicated computation processes. Moreover, it has been notified recently through the collision frequency analysis that those typical hash functions contain several undiscovered flaws [2]. In order to overcome such flaws, the multiple-block-length hash functions have been suggested [3-5]. Nonetheless, the implementation of such multiple-block-length hash functions is complicated in terms of security and computation processes.

As a ubiquitous aspect in nature, chaos is a deterministic nonlinear dynamical system that possesses distinctive properties, mainly involving pseudo-randomness and sensitivity to initial conditions and control parameters. With such properties, chaos-based hash algorithms have consequently been of much interest as an alternative to those of typical hash functions. Several chaos-based hash function algorithms have been proposed recently [6-9]. Despite the fact that these algorithms have offered satisfied statistical performances in terms of statistical performance and collision resistance, the difficulty in small key space, flexibility, low performance, and weak security functions are obstacles that elevate an attempt in designing efficient and secure hash functions. Furthermore, structural topologies of existing algorithms are somewhat complex as evident from multiple maps, multi-stage connections, or multiple feedback loops, leading to complicated signal processing and extensive iteration time.

As for compact and effective chaos-based hash function implementations, this paper presents a new alternative in both chaotic map and hash function topology. The proposed chaotic map employs absolute-value of sinusoidal nonlinearity and offers robust chaotic regions over broad parameter spaces with high degree of randomness through chaoticity measurements using the Lyapunov exponent. The proposed hash function is implemented by a cross-coupled topology. Hash function operations involve an initial stage when the chaotic map accepts initial conditions and input messages in ASCII format, and a hashing stage that accepts input messages and generates the alterable-length hash values. Hashing performances are evaluated in terms of original message condition changes, statistical analyses, and collision analyses. The proposed keyed hash function enhances the collision resistance, comparing

to MD5 and SHA1, and is comparable to other complicated chaos-based approaches.

II. PROPOSED CHAOTIC MAP USING ABSOLUTE SINUSOIDAL NONLINEARITY

A sinusoidal function typically contains an inherent infinite and complex nonlinearity described by an infinite Maclaurin series as follows;

$$\sin(x) = x - \frac{x^3}{3!} + \frac{x^5}{5!} - \dots \quad (1)$$

Such a sinusoidal function in (1) has therefore been utilized as a potential inherent nonlinearity in various chaotic maps. Unlike a single-modal chaotic map, i.e. a logistic map, or another family of multi-modal chaotic maps based on polynomial functions which comprises several mathematical terms, the sinusoidal function potentially provides complex chaotic time series with unique dynamical characteristics.

Table 1 summarizes related chaotic maps implemented based on sinusoidal functions. As for an attempt of using sinusoidal function in a particular application on a hash function design where the input is an ASCII code with values in the region of 32 to 126, the typical sine map in [10] limits the system parameter in the range of (0,4). Although the Iterative Chaotic Map with Infinite Collapses in [11] potentially offers an infinite parameter space, the output time series swing over the values (-1,1) and it consequently may not be suitable for value normalization in parameter space where the previous value is zero. Moreover, other chaotic maps in [12-16] require complicated mathematical models with more-than-one parameter spaces, resulting in the complex process of parameter optimization. This paper therefore presents a one-dimensional sinusoidal chaotic map in combination with absolute-value nonlinearity given by

$$x_{n+1} = \alpha |\sin(\omega x_n + \phi)| \quad (2)$$

where parameters ω and ϕ are frequency and phase shift, respectively. The absolute-value nonlinearity is suggested in order to limit the output values in the range of zero to one.

Fig. 1 shows the detailed block diagram of the proposed absolute sinusoidal chaotic map described in (1). It is seen from Fig.1 that the output is delayed and fed back through the absolute sinusoidal nonlinearity for each iteration for generating chaotic signals. In particular, the two parameters that significantly set dynamic behaviors are frequency and phase shift. In order to primarily analyze the complex dynamics of the proposed chaotic map, the bifurcation diagram is employed as a tool for a qualitative measure. The bifurcation diagram shows a period doubling that accompanies the onset of chaos, and also represents the sudden appearance of a qualitatively different solution for a nonlinear system as some parameter is varied. On the other hand, the positive Lyapunov exponent (λ) is realized as a tool for a quantitative measure. The Lyapunov exponent characterizes the rate of separation of infinitesimally close trajectories, and can be described as

$$\lambda = \lim_{t \rightarrow \infty} \lim_{\Delta x_o \rightarrow 0} \frac{1}{t} \ln \frac{|\Delta x(t)|}{|\Delta x_o|} \quad (3)$$

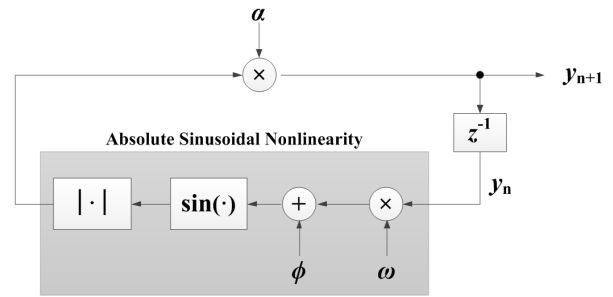


Fig. 1. A detailed block diagram of the proposed chaotic map using absolute sinusoidal nonlinearity described in (1).

where Δx_o is an initial separation of the two trajectories in phase space. Typically, the cases where $\lambda < 0$ and $\lambda = 0$ indicate that the orbit attracts to a stable fixed point or stable periodic orbit and a neutral fixed point, respectively. In the particular case where $\lambda > 0$, the orbit is unstable and the system exhibits chaotic orbits.

In order to investigate chaotic dynamics of the proposed chaotic map, simulations have been performed in MATLAB where initial conditions were set to 0.1 for all cases. Fig.1 shows bifurcating structures of the proposed chaotic map, which are obtained from the positive Lyapunov exponent, i.e. the dark-color region represents chaotic behaviors where $\lambda > 0$ while the white region represents non-chaotic behaviors where $\lambda = 0$ or $\lambda < 0$. Primary investigations on effects of values of parameters α and ω on chaotic dynamics were particularly performed in the region $[0, 10]$ while the phase shift is in the region $[0, \pi]$. Noted that the chaotic behaviors for the phase shift in the region $[\pi, 2\pi]$ completely resemble the dynamics in those of region $[0, \pi]$. In Figs.2 (a) and (b), the values of the gain α were respectively fixed at 1 and 10, and the parameters ω and ϕ were scanned. In Figs.2 (c) and (d), the values of phase shift ϕ were respectively fixed at 0 Radian and π Radian, and the parameters α and ω were scanned. Finally, the value of frequency ω was fixed at 10 rad/s while the parameters α and ϕ were scanned in Figs.2 (e) and (f). It can be concluded from Fig.1 that the proposed chaotic map has a unique pattern of bifurcation structure. The frequency and phase shift are two parameters that significantly set such bifurcation patterns while the gain potentially provide an increase in chaos region. In addition, the proposed chaotic map offers relatively robust chaos over most of the entire parameter spaces, and hence high-complexity operations of the hash function. The selection of parameter values of chaotic map should be in that of dark-color regions in order to guarantee chaotic outputs under most of iteration processes. It should be emphasized in Figs.2 (a) and (b) that the phase shift can be set at zero since chaotic dynamics are still apparent. Upon setting the fixed phase shift at zero Radian, Figs.3 (a) and (b) illustrates the bifurcating diagram and Lyapunov spectrum of the proposed chaotic map for the cases $\alpha=1$ and $\alpha=10$, respectively. It can be considered from Fig.3 (a) that the maximum values of the outputs fall within the range $[0,1]$ while the maximum values of Fig.3 (b) increases correspondingly to an amplifying gain of $\alpha=10$ in each iteration. Nonetheless, there has no significant difference in terms of chaoticity measured by the positive Lyapunov exponent. As a result, the frequency is a major parameter that

TABLE I. SUMMARY OF RELATED CHAOTIC MAPS IMPLEMENTED BASED ON SINUSOIDAL FUNCTIONS.

References	Types of Chaotic Maps	Mathematical Models
[10]	Sine Map	$x_{n+1} = \frac{a}{4} \sin(\pi x_n)$
[11]	Iterative Chaotic Map with Infinite Collapses	$x_{n+1} = \sin\left(\frac{a}{x_n}\right)$
[12]	Circle Map	$x_{n+1} = x_n + \Omega - \frac{K}{2\pi} \sin(2\pi x_n)$
[13]	Climbing Sine Map	$x_{n+1} = M_a(x_n + a \sin(2\pi x_n))$
[14]	Sine Iterative Map	$x_{n+1} = \sin^2(a \arcsin(\sqrt{x_n}))$
[15]	Moir Grating Map	$x_{n+1} = 0.5 + 0.5 \text{sign}(\sin(\frac{2\pi}{\lambda} x_n))$
[16]	Sine Square Map	$x_{n+1} = A \sin^2(x_n - B)$

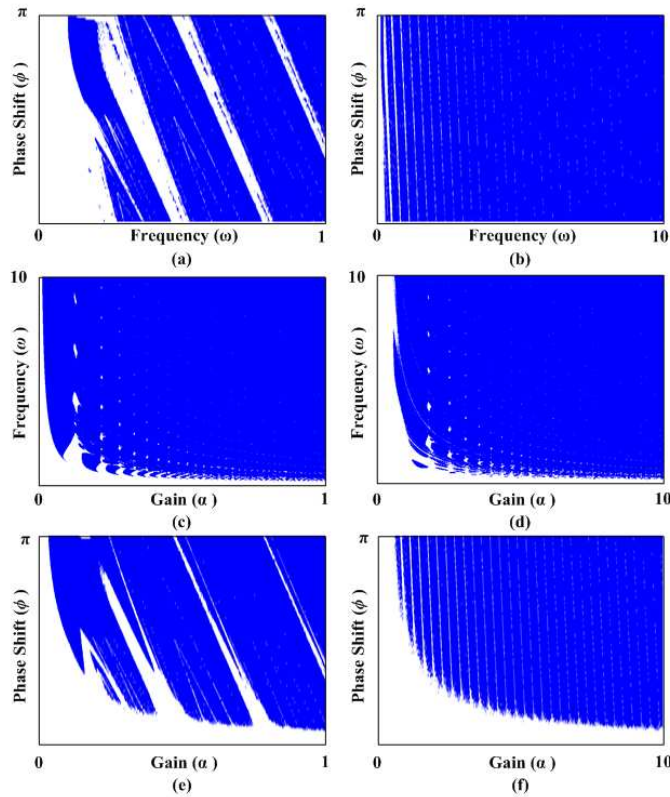


Fig. 2. Bifurcation structures of the proposed chaotic map using absolute sinusoidal nonlinearity.

determines chaotic dynamics and the gain can be arbitrarily set to any values based on required conditions with no changes in complexity.

As for a particular example, the proposed chaotic map is illustrated for its chaotic dynamics with parameter values $\alpha=1$, $\omega=10$, and $\phi=0$, i.e. $x_{n+1}=\sin(10x_n)$. Fig.4 (a) shows an apparently chaotic time-domain waveform. Fig. 4(b) shows the Cobweb plots between x_n and x_{n+1} , indicating that the iterations are mapped over the absolute-value nonlinearity. Figs.4 (c) and (d) show the frequency spectrum using periodogram and autocorrelation plots, respectively. It can be seen that the values are distributed with flat spectrum feature and the values are random reflected by the low autocorrelation of less than approximately 0.01. It can be concluded that the map

potentially offers robust and effective randomness for use in hashing algorithm.

III. PROPOSED KEYED HASH FUNCTION

The whole structure of the proposed hashing scheme is depicted in Fig. 5. Assuming that a 128-bit hash value is required, the procedures for generating hash values are described as follows:

(1) The secret keys of the algorithm include the selected initial conditions $y_{0,1}$ and $y_{0,2}$.

(2) The original message M is padded such that its length is a multiple of 2.

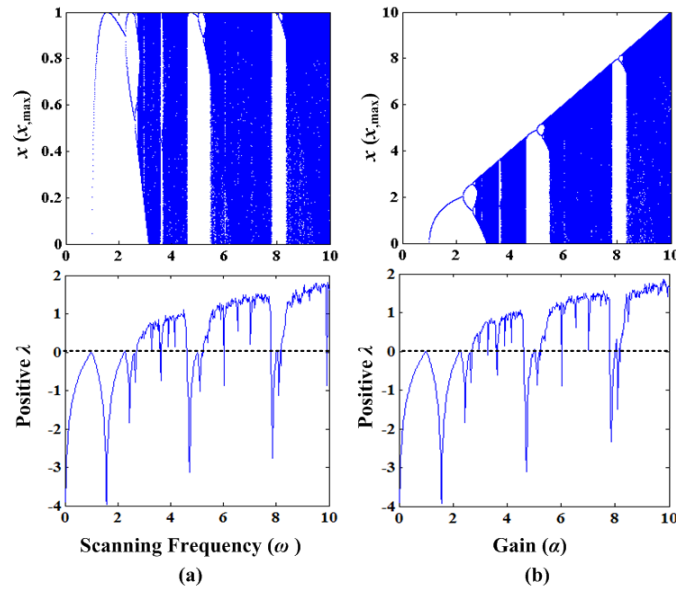


Fig. 3. Bifurcation diagram and Lyapunov spectrum of the proposed chaotic map using absolute sinusoidal nonlinearity.

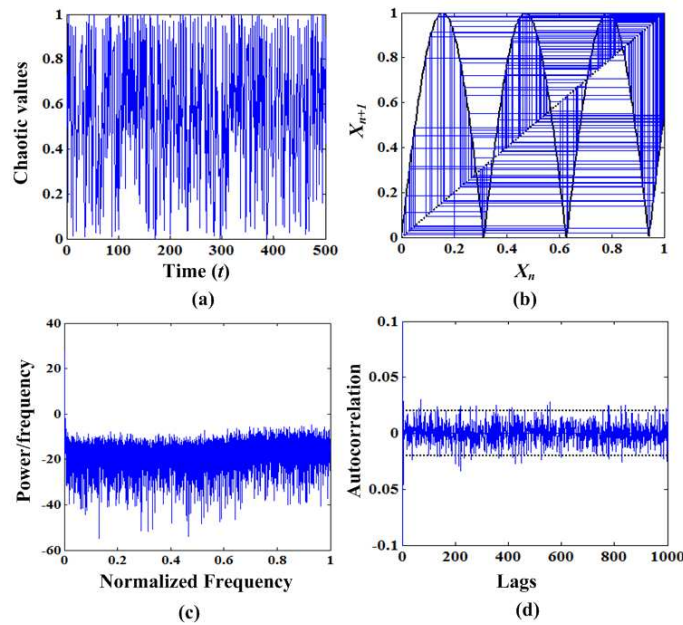


Fig. 4. Chaotic dynamics of the proposed chaotic map where $\alpha=1$, $\omega=10$, and $\phi=0$; (a) Time-domain waveforms, (b) Cobweb plots, (c) Frequency spectrum using periodogram and (d) Autocorrelation plots.

(3) The padded message is divided into 2 sub-blocks of length S , $\omega_{i,1}$ and $\omega_{i,2}$, where $i = 1, \dots, S$.

(4) For the input stage, S iterations are needed for the absolute sine map with the intention of generating the intermediate output. The first iterations after initial stage are $y_{i,1}(t) = \alpha_1 |\sin(\omega_{i,1} \cdot (y_{i-1,1} + y_{i-1,2}) + \phi_1)|$, and $y_{i,2}(t) = \alpha_2 |\sin(\omega_{i,2} \cdot (y_{i-1,1} + y_{i-1,2}) + \phi_2)|$, for $i = 1, \dots, S$.

(5) The last two output values $y_{S,1}(t)$ and $y_{S,2}(t)$ are mapped into decimal integer values d_1 and d_2 with interval $[0, 2^{64}]$

(6) The decimal integer d_1 and d_2 are converted into 64-bit binary numbers b_1 and b_2 .

(7) Finally, b_1 and b_2 are cascaded to form a 128 bit hash value H .

IV. PERFORMANCE ANALYSIS

A. Uniform Distribution of Hash Value

The uniform distribution of hexadecimal hash value is the crucial property of hashing scheme. In Fig. 6(a) the ASCII characters of the original message are localized in a small

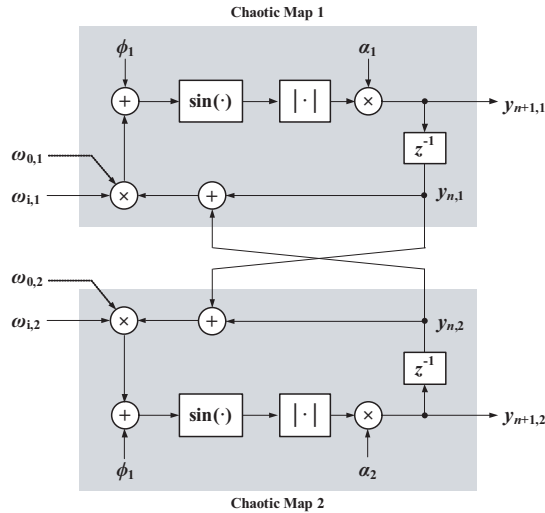


Fig. 5. Block diagram of the cross-coupled topology of the proposed keyed hash function.

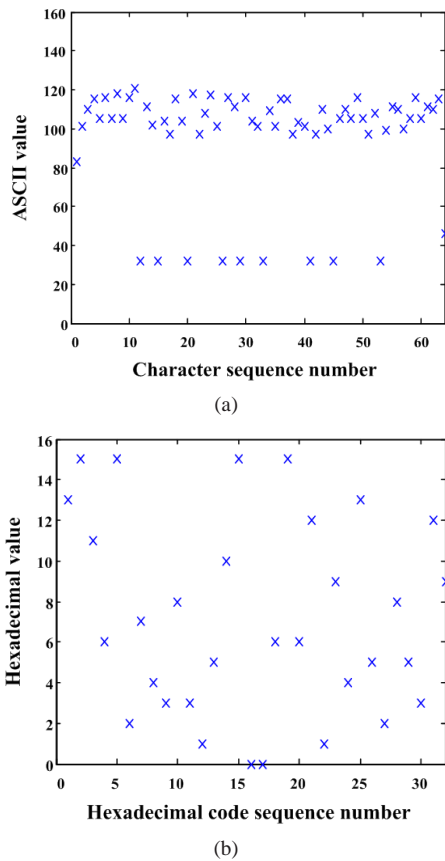


Fig. 6. The distribution of values : (a) original message in ASCII: (b) hash values in hexadecimal format.

interval between 97 and 122. On the contrary, the hash values of the hashing scheme are uniformly spread over the possible range of hash values as illustrated in Fig. 6(b). This signifies that no information of the original message remains after the hashing process.



Fig. 7. An original grayscale Lena image.

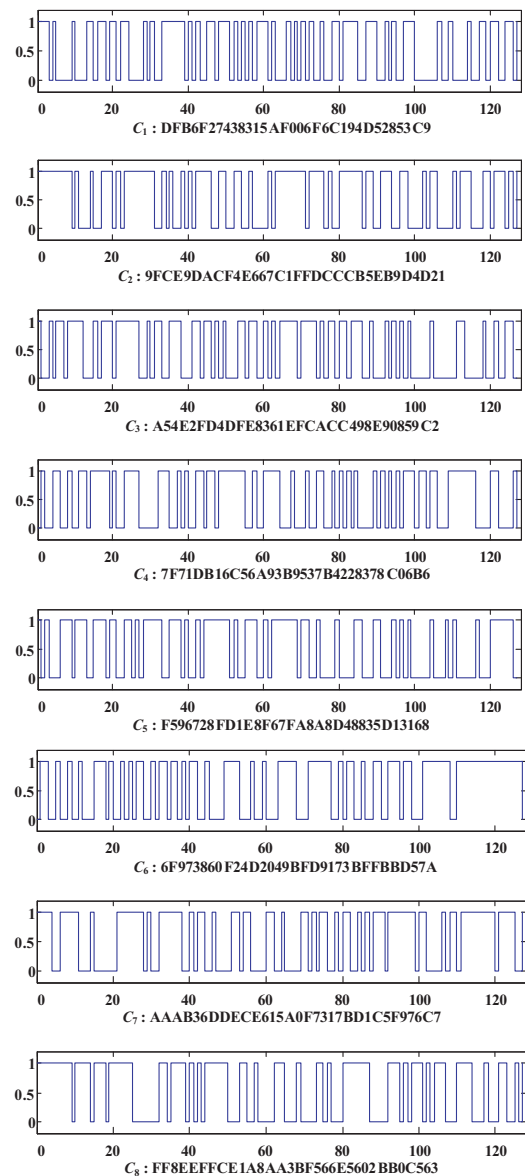


Fig. 8. Corresponding binary sequences of the cases C_1 , C_2 , C_3 , C_4 , C_5 , C_6 , C_7 and C_8 .

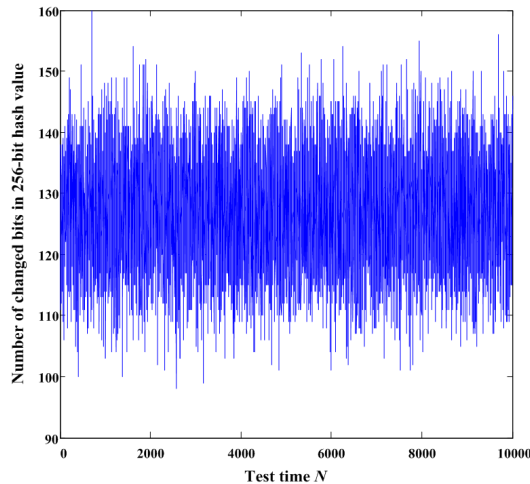


Fig. 9. Distribution of the number of bits changed.

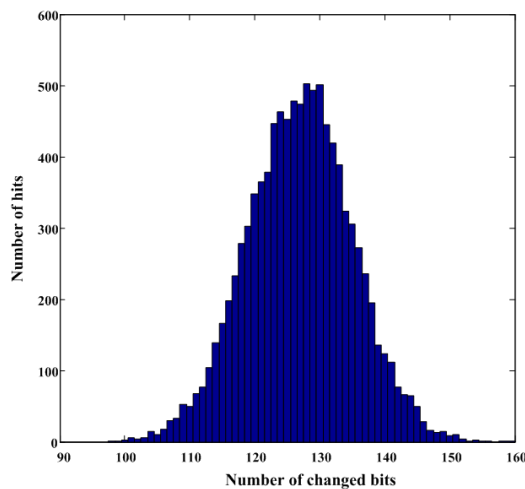


Fig. 10. Histogram of the distribution of number of bit changes for =256 and =10000.

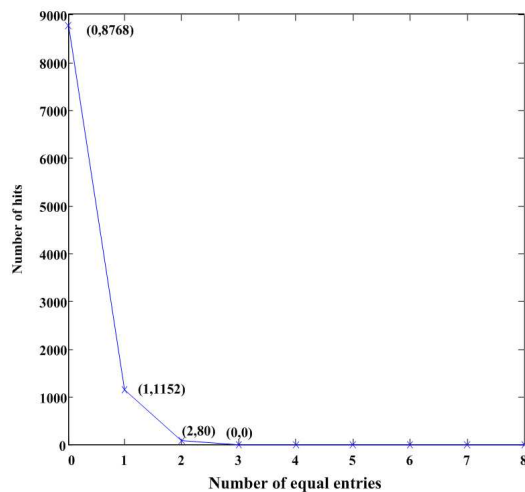


Fig. 11. Distribution of the number of locations where the ASCII characters are identical for =256 and =10000.

B. Sensitivity to Small Changes in Message and Initial Conditions

This subsection depicts the high sensitivity of the proposed hashing function to small changes in original messages and initial conditions. In order to explore this issue, the following tests have particularly been verified.

C_1 : The original message is: “Sensitivity of hash value to the message and initial conditions.”.

C_2 : Substitute the initial condition $y_{0,1} = 2233.4567$ with $y_{0,1} = 2233.4568$. The two 128-bit hash values for C_1 and C_2 differ in 65 positions.

C_3 : Substitute the first character of the original message by “A”. The two 128-bit hash values for C_1 and C_3 differ in 63 positions.

C_4 : Substitute the character “i” in the word “initial” by “e” to become “enitial”. The two 128-bit hash values for C_1 and C_4 differ in 66 positions.

C_5 : Substitute the last character of the original message “.” by “;”. The two 128-bit hash values for C_1 and C_5 differ in 62 positions.

C_6 : Lena original image (256 x 256 pixels) shown in Fig. 7.

C_7 : Substitute the gray value of the pixel located at the upper left corner by “0”. The two 128-bit hash values for C_6 and C_7 differ in 63 positions.

C_8 : Substitute the gray value of the pixel located at the lower right corner by “1”. The two 128-bit hash values for C_6 and C_8 differ in 63 positions.

The corresponding binary sequences of $C_1, C_2, C_3, C_4, C_5, C_6, C_7$ and C_8 are illustrated in Fig. 8. The result reveals that any small change in the original message and initial condition can result in a 50% chance of changing for each bit of hash value.

C. Confusion and Diffusion

Strong confusion and diffusion properties of the hashing scheme are essential to make it durable to most attacks. The purpose of diffusion is to disperse the hash values randomly over the possible range with the intention of hiding statistical properties of the original message. Confusion employs the transformation to make the relationship between the original message and hash value as complicated as possible.

The n -bit hash value of a random message of size $L = 50n$ is produced and displayed. Diffusion and confusion test procedure proceeds as follows: the n -bit hash value of a random message of size $L = 50n$ is computed. One bit of the random message is randomly selected and switched, and the n -bit hash value of the altered message is computed. Then two hash values are compared and the number of bit changes is quantified. This test is repeated N times for $N = 256, 512, 1024, 2048$ and 10000 for hash values of size n , where $n = 128, 160$ and 256 . The statistical measures, as illustrated in Table 2, are employed for statistical computations where B_i is the number of bit changes in the i -th test. The results achieved

in tests for $n = 128, 160$ and 256 and $N = 256, 512, 1024, 2048$ and 10000 are demonstrated in Tables 3-5.

Distribution of the number of bits changed for various number of test times N is depicted in Fig. 9. Fig. 10 illustrates the histogram distribution of the number of bit changes for $n=256$ and $N=10000$. From the results in Tables 3-5, it can be observed that \bar{B} and P are nearly the ideal values of $n/2$ and 50% respectively. All values of ΔB and ΔP are very small, which signifies that diffusion and confusion capability of the proposed hashing scheme is very strong and stable.

D. Collision Analysis

1) *Collision Test*: Hash collision occurs when two distinct input messages generate the identical hash values. The following collision test has been done with the purpose of measuring the collision resistance of the proposed hashing scheme. The n -bit hash value of a random message of size $L = 50n$ is produced and displayed in ASCII format. Then one bit in the generated message is randomly selected and flipped. The new hash value is also produced and displayed in ASCII format. The ASCII characters of these two hash values are then compared. The number of hits, which is the number of ASCII characters with the same value at the same position, is quantified. The absolute difference of these two hash values is expressed as

$$d = \sum_{i=1}^{n/8} |dec(m_i) - dec(m'_i)| \quad (4)$$

where m_i and m'_i are the i^{th} ASCII character of the original and modified hash values, respectively, and $dec()$ converts m_i and m'_i to the corresponding decimal numbers. This procedure is repeated 10000 times. Table 6 depicts the minimum, maximum and mean values of d . It can be seen that the average absolute difference of each character of the proposed hashing scheme is close to the ideal value of 85.43 [20]. This means that the proposed hashing scheme has the stronger collision resistance than the well-known approaches such as MD5 and SHA-1, and the other chaos-based approaches in [17,18,19,21,22]. The distribution of the number of hits is demonstrated in Fig. 11. It can be seen that the maximum number of equal character is only 2. The results signify that the probability of collision is extremely low.

2) *Resistance to Birthday Attack*: A birthday attack is a kind of cryptographic attack based on mathematical behind the birthday problem in probability theory. The name is obtained from the surprising result that in a room of 23 people, there is a probability of 50% that at least two people have the same birthday. The hashing scheme should be robust against birthday attack, which makes it difficult to find two distinct messages that have the same hash value. The difficulty of the birthday attack depends on the size of the hash value. For a secure hashing scheme with n -bit hash value, the difficulty of the attack is $2^{n/2}$. Therefore, the value of n is needed to be large enough to make a birthday attack computationally infeasible. For instance, if the size of the hash value is set to 256, the difficulty of the attack would be 2^{128} . This keeps the system robust against this type of attack.

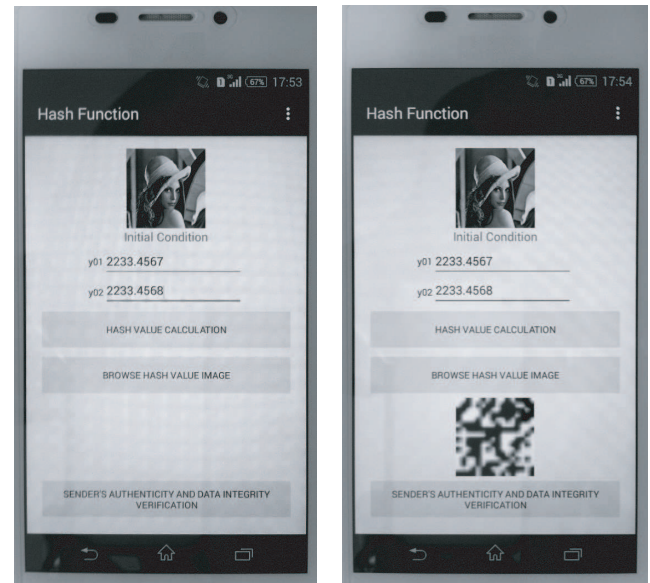


Fig. 12. Android application user interface : (a) image input for hash value calculation (b) displaying calculated hash value image in PNG format.

E. Speed Analysis and Hash Values Obtained from Different Environments

The running speed of the proposed hashing scheme on a 3.20 GHz Intel(R) Xenon(R) computer with 16 GB of RAM running Windows 7 (64 bit) is approximately 10.27 Megabits per second. The proposed hashing scheme is iterated using double precision floating-point arithmetic. The IEEE-754 floating-point standard was adopted in the early 1980s. If two computers with different platforms (operating systems and hardware) employ the IEEE-754 floating-point standard, two hash values produced by both computers must be identical [23]. In order to affirm this issue, the hash values of the origin message illustrated in section 4 are created in computers with different CPU, operation systems and amounts of memory. The results of the hash function at different environments are demonstrated in Table 7. As can be seen, it reveals that if the IEEE-754 floating-point standard is employed, the hash values obtained from different computers must be the same.

V. IMPLEMENTATION OF PROPOSED KEYED HASH FUNCTION ON ANDROID DEVICE

The implementations of the proposed keyed hash function on an Android device are presented in order to verify sender's authenticity and the integrity of transmitted data. Fig. 12 demonstrates Android application user interface for hash value calculation. The procedures for hash value calculation are described as follows:

- (1) Input the initial conditions ($y_{0,1}$ and $y_{0,2}$).
- (2) Choose and import the image file.
- (3) Click on "HASH VALUE CALCULATION" button.
- (4) The calculated hash value is displayed and saved as PNG file.

TABLE II. STATISTICAL ANALYSIS FORMULAS.

No.	Statistical measures	Formulas
(1)	Minimum number of bit changes	$B_{\min} = \min(\{B_i\}_{i=1}^N)$
(2)	Maximum number of bit changes	$B_{\max} = \max(\{B_i\}_{i=1}^N)$
(3)	Mean number of bit changes	$\bar{B} = \frac{1}{N} \sum_{i=1}^N B_i$
(4)	Mean changed probability	$P = \frac{\bar{B}}{n} \times 100\%$
(5)	Standard deviation of the number of bit changes	$\Delta B = \sqrt{\frac{1}{N-1} \sum_{i=1}^N (B_i - \bar{B})^2}$
(6)	Standard deviation	$\Delta P = \sqrt{\frac{1}{N-1} \sum_{i=1}^N \left(\frac{B_i}{n} - P\right)^2} \times 100\%$

TABLE III. THE RESULTS OBTAINED THROUGH STATISTICAL MEASURES FOR A 128-BIT HASH VALUE.

Measures	$N = 256$	$N = 512$	$N = 1024$	$N = 2048$	$N = 10000$
B_{\min}	46	47	43	41	43
B_{\max}	82	80	80	81	83
\bar{B}	63.32	63.17	63.24	63.53	63.21
$P(\%)$	49.47	49.35	49.41	49.31	49.63
ΔB	5.59	5.78	5.55	5.64	5.65
$\Delta P(\%)$	4.36	4.51	4.34	4.41	4.41

TABLE IV. THE RESULTS OBTAINED THROUGH STATISTICAL MEASURES FOR A 160-BIT HASH VALUE.

Measures	$N = 256$	$N = 512$	$N = 1024$	$N = 2048$	$N = 10000$
B_{\min}	63	62	60	60	58
B_{\max}	98	100	102	99	101
\bar{B}	79.20	79.43	79.17	79.13	79.33
$P(\%)$	49.50	49.64	49.48	49.45	49.58
ΔB	6.11	5.99	6.18	6.08	6.02
$\Delta P(\%)$	3.82	3.74	3.86	3.80	3.76

TABLE V. THE RESULTS OBTAINED THROUGH STATISTICAL MEASURES FOR A 256-BIT HASH VALUE.

Measures	$N = 256$	$N = 512$	$N = 1024$	$N = 2048$	$N = 10000$
B_{\min}	104	103	106	103	94
B_{\max}	153	152	154	150	154
\bar{B}	127.87	127.20	127.49	127.20	127.46
$P(\%)$	49.95	49.68	49.80	49.68	49.79
ΔB	7.81	7.87	8.40	7.84	7.92
$\Delta P(\%)$	3.05	3.07	3.28	3.06	3.09

The initial conditions are shared between sender and receiver and the image file is transmitted to the receiver along with the calculated hash value image file. Android application user interface for sender's authenticity and data integrity verifications can be illustrated in Fig.13. The sender's authenticity and data integrity verification processes are described as follows:

- (1) Input the initial conditions ($y_{0,1}$ and $y_{0,2}$).
- (2) Choose and import received image and hash value image files.
- (3) Click on "SENDER'S AUTHENTICITY AND DATA INTEGRITY VERIFICATION" button.

(4) The receiver-calculated hash value is compared with the received hash value.

(5) The results of sender's authenticity and data integrity verifications are displayed on the screen.

These results confirm that the proposed keyed hash function can be used to verify the sender's authenticity and the integrity of transmitted data on the Android device.

VI. CONCLUSION

The new compact and robust chaos-based keyed hash function has been presented. The proposed chaotic map exploits absolute-value of sinusoidal nonlinearity for generating

TABLE VI. COMPARISON OF ABSOLUTE DIFFERENCE, WHERE = 10000.

Absolute difference (<i>d</i>)	Min.	Max.	Mean	Mean/Character
MD5(128bit)	590	2074	1304	81.5
SHA-1(160bit)	795	2730	1603	80.15
Kanso's scheme[17](128bit)	737	2320	1494	93.375
Wang's scheme [18] (128bit)	689	2295	1526	95.375
Ren's scheme [19] (128bit)	599	2455	1439	89.9375
Wang's scheme [20] (128bit)	655	2064	1367	85.4375
Xiao's scheme [21] (128bit)	658	2156	1431	89.44
Xiao's scheme [22] (128bit)	605	1952	1227	76.69
The proposed scheme (128bit)	544	2400	1348	84.25
The proposed scheme (160bit)	809	2782	1687	84.35
The proposed scheme (256bit)	1402	3954	2716	84.87

TABLE VII. HASH VALUES OBTAINED FROM DIFFERENT ENVIRONMENTS.

CPU	OS	Memory	Hash values
Intel Core 2 Duo E7400 2.80 GHz	Windows XP	2 GB	DFB6F27438315AF006F6C194D52853C9
Intel Core i3-2100 3.10 GHz	Windows 7	4 GB	DFB6F27438315AF006F6C194D52853C9
Intel Core i5 2.90 GHz	Mac OS	16 GB	DFB6F27438315AF006F6C194D52853C9

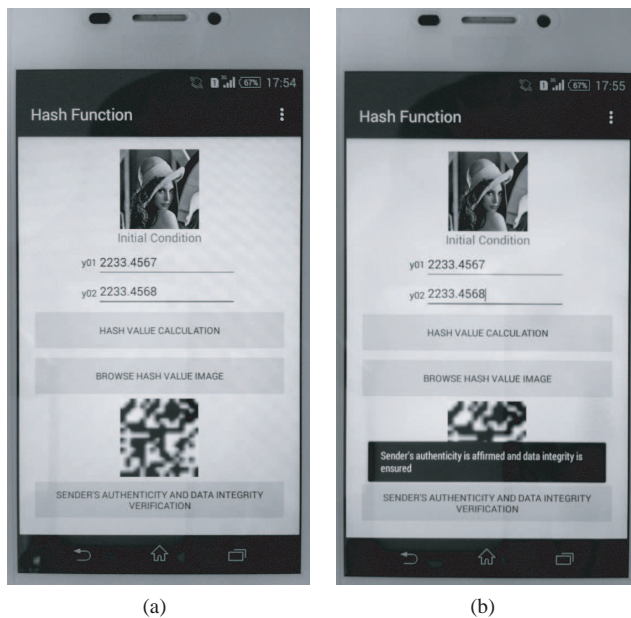


Fig. 13. Android application user interface : (a) image input for verifying sender's authenticity and data integrity (b) displaying results of sender's authenticity data integrity verifications.

highly random iterated values in the diffusion process of ASCII input messages. Chaotic aspects have been investigated through bifurcation structures of Lyapunov exponent as well as Cobweb plots, autocorrelation, and signal characteristics in both time and frequency domains. The proposed hashing structure is relatively simple using only two chaotic maps in the cross-coupled topology that enhances randomness quality for statistical performances. The designed hashing algorithms involve the initial stage when the cross-coupled maps accept initial conditions utilized as secret keys, and the iterative

hashing stage that accepts input messages and generates the alterable-length hash values. With such a compact hash function structure, simulation results have revealed several desirable features in terms of statistical performances, involving the mean changed probabilities that are very close to 50%, and the mean number of bit changes that is also close to a half of hash value lengths. In addition, the mean absolute difference of each character values for the hash values of 128, 160 and 256 bits are close to the ideal value of 85.43. This indicates that the proposed hash function has superior performance over well-known algorithms such as MD5 and SHA1, and the other complex structures of chaos-based approaches in [17,18,19,21,22]. A new implementation of hash function Android application has been demonstrated. As a result, the proposed hash function has offered a potential alternative to cryptography and secures protocol methods.

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An Efficient Method for Distributing Animated Slides of Web Presentations

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Abstract—Attention control of audience is required for successful presentations, therefore giving a presentation with immediate reaction, called reactive presentation, to unexpected changes in the context given by the audience is important. Examples of functions for the reactive presentation are shape animation effects on slides and slide transition effects.

Understanding the functions that realize the reactive presentation on the Web can be useful. In this work, we present an effective method for synchronizing shape animation effects on the Web, such as moving the objects and changing the size and color of the shape objects. The main idea is to make a video of animated slides, called Web Slide Media, including the page information of slides as movie chapter information for synchronization. Moreover, we explain a method to reduce the file size of the Web slide media by removing all shape animation effects and slide transition effects from a Web slide media item, called Sparse Web Slide Media.

We demonstrate that the performance of the system is enough for practical use and the file size of the Sparse Web Slide Media is smaller than the file size of the Web Slide Media.

Keywords—Collaborative tools; communication aids; information sharing; Web services

I. INTRODUCTION

Presenters require instant manipulation of their presentation material to deal with unexpected contexts, such as the state of their audience, unexpected questions, and the knowledge of the audience. Attention controls that use animations and other effects are effective for dealing with the unexpected contexts. The conventional PowerPoint software, however, does not support instant manipulation of slide objects (i.e., objects on a slide) in the presentation mode. We implemented methods for controlling the attention of the audience members and for adding visual information to the presentation in PowerPoint in our prior research [1], [2]. We implemented a system that provides for the manipulation of slide objects so that users give a *reactive presentation*. The reactive presentation refers to giving a presentation with immediate reaction to unexpected changes in the context given by the audience. Although the methods for controlling the attention of the audience members were based on functions for manipulating objects on slides in real time during a presentation, the methods did not support giving a remote presentation. In this paper, we describe a reactive presentation system that synchronizes the slides between the presenter and the audience on the Web independently of the distance between them.

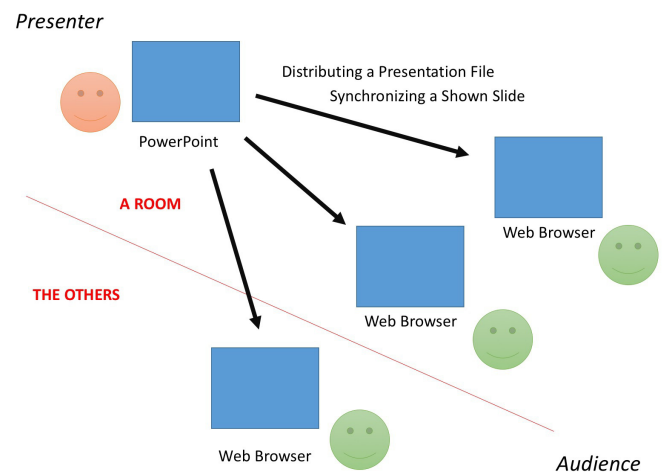


Fig. 1: Web Presentation.

This paper is organized as follows. In Section II, we describe features of the Web presentation support. In Section III, we describe a Web slide media item that is shown on a Web browser. In Section IV, we describe the system architecture of the Web presentation support systems and essential requirements for reactive presentations. In Section V, we describe the implementation of the system. In Section VI, we describe experimental results for evaluating communication traffic. In Section VII, we discuss the reason for traffic is increasing in the experiment and the ease of configuring the system settings. We conclude this paper in Section VIII.

II. WEB PRESENTATION SUPPORT

Collaborative presentation support enables us to have a close conversation between a presenter and audience members. Figure 1 shows that a presenter synchronizes the shown slide with the audience members and collects comments from audience members during a presentation. The audience members usually ask questions after the presentation, however, they may want to interrupt the presentation to ask questions.

Web presentation has been widely studied recently. The Web media slides and Silhouette Web Presentation system are useful for supporting presentations on the Web. HTML5 is used for Web presentation [3], [4]. There is a JavaScript library

for publishing multimedia Web contents using HTML5 [5]. Some studies on E-learning have used PowerPoint [6], [7].

In Figure 1, a presenter reuses his/her presentation file for knowledge sharing. Such service allows users to share presentations, documents, videos, and webinars. The service's goal is to share knowledge online.

The number of Web services to share presentation slides is increasing rapidly. For example, Office 365¹ is a cloud service provided by Microsoft. iCloud² is a cloud service provided by Apple. These cloud services provide, respectively, an Office PowerPoint and a Keynote³ on the Web and allow a user to edit a presentation and save the presentation on the cloud. Of course, a user begins a presentation slide show on a Web browser. SlideShare⁴ is a Web service to share slides.

Presentation applications such as PowerPoint and Keynote provide an abundance of wonderful animations and effects for presentations. However, adding and editing these animations and effects, such as the movements and changing the sizes and colors of the slide objects, are possible only if the applications enable an editing mode. Nevertheless, the slide show mode is better for maintaining the attention of the audience members at the time of the presentation because there is no salient cueing, with the exception of the slide objects, in the slide show mode. The problem is that existing presentation applications (e.g., PowerPoint and Keynote) do not support editing slide objects in the presentation mode. We cannot apply new animations and effects to slide objects when we need to immediately get the attention of the audience members in the presentation mode. For presentation applications, we apply an abundance of animations and effects to objects on slides. It is also beneficial for us to use the rich number of animations and effects in unexpected situations during presentations.

We assume that presenters and audience members should not be specialists of ICT technologies. We design a new presentation support system under the following constraints: 1) the presentation slide should be composed by using existing presentation software, 2) audience members only use Web browsers during presentations, and 3) presentation slides should be shared and reusable on the Web.

A. Reactive Presentation

Although studies on presentation support systems have been conducted for some time, studies about reactive presentation have never been conducted in this previous research. Our focus is to develop a presentation support system that enables us to react to unexpected context changes. We developed a new approach to properly react to requests of audience members during presentations.

We propose a system for manipulating objects on slides in real time during a presentation to give a reactive presentation. Manipulating slide objects, such as moving, zooming, and adding some effects, the presenter guides and maintains the attention of the audience members and communicates information in visuospatial terms by applying the animations and effects of slide objects reactively.

The presenter controls the attention of the audience members by adding animations and effects to deal with a certain context and reflecting on something in the presentation. When the audience members ask the presenter to explain the topic based on some unexpected conditions that are difficult to prepare for beforehand, the presenter only explains them by speaking or pointing them out.

1) *Attention Control*: Attention control in presentation is used to guide and maintain the attention of the audience members to crucial parts of slides, and it is effective for dealing with the changing context in a presentation. Attention guiding by means of cueing reduces the extraneous cognitive load of the audience [8].

2) *Animations for Attention Control*: Focusing on exogenous attention [9] by using animations and effects is useful for properly guiding the attention of the audience members. Visually guiding and maintaining their attention by using animations and effects reduces their cognitive load [10].

Animations and effects effectively communicate non-verbal information to the audience [11] by expressing various movements, colors, and features for slide objects.

3) *Slide Object Manipulation for Instant Attention Control*: It is necessary for reactive presentation to allow users to manipulate slide objects in real time in a slide show. A user should be able to manipulate slide objects during a slide show without changing into the edit mode to avoid losing the attention of the audience members. A person's eye direction and attention are almost entirely consistent [12], [13].

III. WEB SLIDE MEDIA

There are two requirements for inspecting a presentation file on the Web: The file allows users to skip slides and to play animations on slides. To cover the requirements, a presentation file should be Web-friendly content. We solve a visual appearance problem using HTML5 specifications that is supported by many modern Web browsers. To prevent the broken appearance of slides, we convert the presentation file to a video file. Converting to a video file uses PowerPoint functions.

A Web slide media item consists of an original presentation video and chapter information generated by our system. An HTML5-supported Web browser plays a Web slide media without browser plugins. A Web slide media item expresses animations and stops the playback with its own chapter information when the playback position reaches each of slide transition ends and animation ends. To solve an operability problem of an animated slide, playback animations, and previewing any slides randomly, we add chapter information to the animated slide.

A. Chapter Information

In this research, we focus on users creating a presentation file with PowerPoint. PowerPoint also has a video output function, but the output file has no chapter information. The presentation software Keynote has a function to save a presentation file as a video file. The function also adds chapter information to the output video file. To add chapter information

¹<http://office.microsoft.com/en-us/home-and-student/>

²<https://www.icloud.com/>

³<http://www.apple.com/iwork-for-icloud/>

⁴<http://www.slideshare.net/>

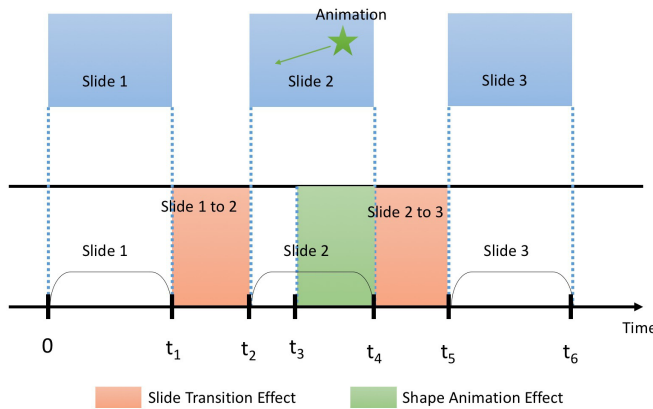


Fig. 2: Chapter Information.

to a video file, a player system detects slide transition and animation ends using the chapter informations.

Chapter information *chaps* consists of tuples of a timestamp, a slide number, and an animation number. The *chaps* consists of multiple pieces of chapter information, $chap_i$, where i indicates the chapter number. $chap_i$ consists of the tuple (t, s, a) , where t is the start time of an animation, s is the slide number at t and a is the animation number.

Using chapter information, a slide transition on a Web slide media item plays the video from a start time t . Figure 2 shows a relationship between slides of a presentation file and a time sequence of Web slide media. In Figure 2, the upper rectangles are three slides. Every slide has a slide transition, and Slide 2 has an animation. The bottom indicates the time series of a Web slide media item. Where the time of the head of the video is zero, the span from 0 to t_1 indicates Slide 1 on the video timeline. The span from t_1 to t_2 is a slide transition from Slide 1 to Slide 2. The span from t_2 to t_4 represents Slide 2. The span from t_2 to t_3 represents prior to the beginning of an animation, and the span from t_3 to t_4 is the animation. The span from t_4 to t_5 is a slide transition from Slide 2 to Slide 3, and the span from t_5 to t_6 represents Slide 3. It is possible to control slide playback on the video sequence by the beginning and end times of a slide, an animation, and a slide transition. In this case, to preview Slide 2 is an alternative on playback from t_1 through t_2 on the video sequence.

B. Sparse Web Slide Media

In comparison to the original video, generally, the size of the Web slide media tends to become larger. The file size of a Web slide media item created from a rich presentation file is larger than one without animations. The fidelity of animations of a Web slide media item and the file size have a trade-off relationship. In this research, our system enables users to adjust the priority of the relationship. To reduce the file size, users may adjust the resolution of a Web slide media item and the frame rates to lower. Also, removing video frames of slide transitions and animations produces smaller file size.

We called this type of Web slide media Sparse Web Slide Media. Specifically, a sparse Web slide media item consists of frames at t_1, \dots, t_6 of the Web slide media. In cases when it is necessary to preview a presentation file or reduce a file size, sparse Web slide media is a useful way.

IV. SYSTEM ARCHITECTURE AND REQUIREMENTS

We describe the design of our collaborative Web presentation support system, called “Silhouette Web Presentation System.” The slides for the Silhouette Web Presentation System are called “Web media slides.” Web media slides contain the image contents. In this section, we show our three design goals. Then we explain the three functions that we implemented to achieve a collaborative Web presentation.

A. Design Goals

1) *Comment Aggregation*: Collecting a comment in real time for a presentation and making a relation between the comment and a shown slide in the presentation prevents this information from becoming disorganized. All comments on a slide should be stored in it automatically to avoid losing valuable opinions.

2) *Reactive Presentation*: Both presenters and audience members require attention control functions to communicate with each other. It is difficult for audience members to tell the presenter slides they want to watch, because they do not have a proper way to indicate the slides in their mind.

3) *Easy to Use*: It is not necessary for a presenter to upload a presentation file. Audience members only access a Silhouette audience system on the Web browser and log in with a specific token without downloading the presentation file. A presenter uses an existing presentation software. An audience system is executable for many environments.

B. Functions

1) *Automatic Comment Crawler*: This function collects comments from the Web and saves the comments into the current slide. The audience members make comments anytime by using Twitter, email, and so forth. The collected comments are displayed on the left side of the presenter’s screen. These comments are saved into the shown slide at that time in the presentation. This function provides the audience members with the ability to ask questions and make comments during the presentation and provides the presenter with the ability to log the comments after the presentation.

2) *Slide Synchronization*: This function is to synchronize shown slides between a presenter and audience members. A tablet device in the proposed system has two modes: One is a free mode, and the other is a viewer mode. A displayed slide on a device that is set to the viewer mode is synchronized with that on the presenter’s device. A device that is set to the free mode is not synchronized. A user using the device can change the displayed slide. Then, the same slide is displayed on the presenter’s device and the viewer mode devices anytime. The presenter’s device sends synchronous messages to the other devices via a Silhouette hall. The devices that received the synchronous messages redraw their displayed slide based on these messages.

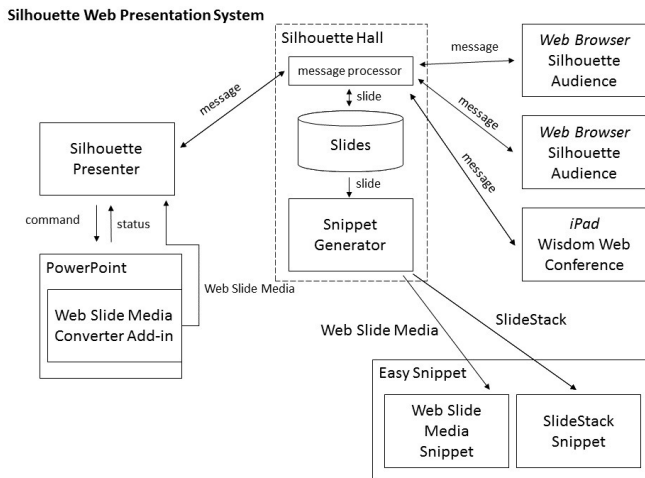


Fig. 3: System Architecture of the Silhouette Web Presentation System.

3) *Web Presentation Snippet*: To share a presentation slide file, the system has a feature to make a Web snippet and append it to existing Web pages. It is important to note that existing animation effects in the slides are also useful to control the attention of audience members. The slides are converted to Web slide media by the add-in of PowerPoint, WW Media, we developed. A Web slide media item expresses shape animations and slide transitions of the presentation slide file.

C. Architecture

We developed a reactive presentation system called the Silhouette Web Presenter System. An outline of our current system's architecture is shown in Figure 3. The system consists of three parts: a Silhouette presenter, a Silhouette audience, and a Silhouette hall. A presenter uses the system, and the slides are synchronized to audience members. Additionally, the presenter can reuse the presentation as a Web snippet and append it to any existing Web pages. Web slide media in Figure 3 are HTML5 contents generated from presentation slides, which are used for Web presentations.

V. IMPLEMENTATION

In this section, we describe the implementation of the Silhouette Web Presenter System. The Silhouette Web Presenter System consists of three subsystems: a server-side system and two client-side systems. The server-side system, called a Silhouette hall, relays control messages and presentation slides. The first client-side system, called a Silhouette presenter, is used by a presenter. The second client-side system, called a Silhouette audience, is used by an audience member of the presenter.

We given an overview of usage of the Silhouette Web Presenter System. The system identifies a synchronization group by a specific synchronization token. The presenter opens a presentation file in PowerPoint and starts the Silhouette presenter. The audience members start each Silhouette audience in their Web browsers. The presenter and audience members set the same synchronization token to their system.

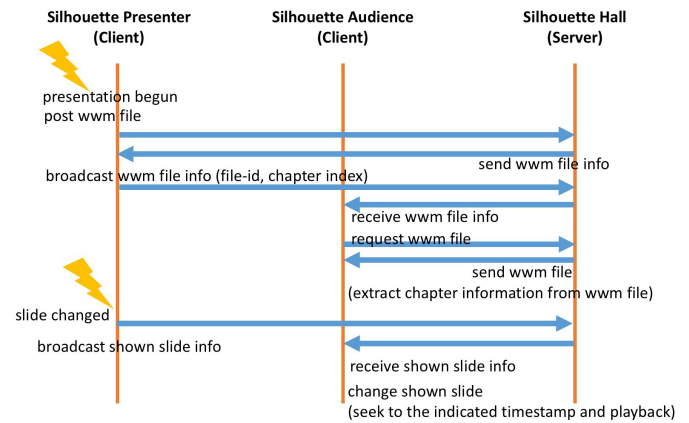


Fig. 4: Distributing Web Slide Media and Synchronizing Current Slide.

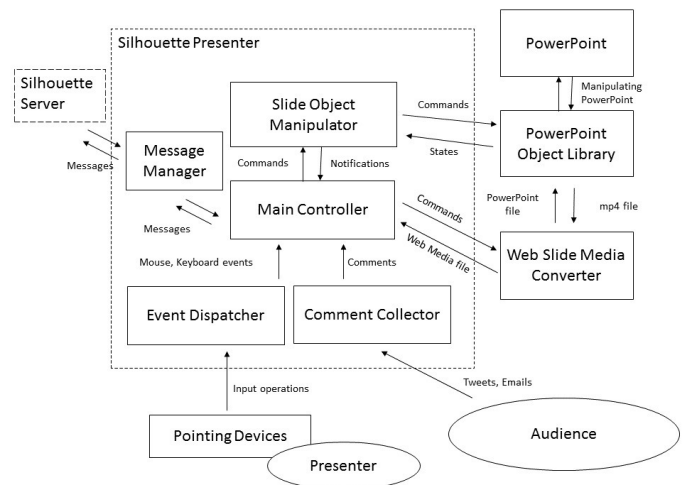


Fig. 5: System Diagram of the Silhouette Presenter.

A. Silhouette Hall

A Silhouette hall provides two services. The first is a file storage service over HTTP. The second is a synchronization message relay service over WebSocket. We implemented a Silhouette hall in Node.js⁵.

Figure 4 shows sequences of distributing a Web slide media item and synchronizing the current shown slide. First, a Silhouette presenter system converts the presentation file into a Web slide media item before the beginning of a slide show on PowerPoint, and uploads the Web slide media to the Silhouette hall. The Silhouette hall responds to uploading the animated slide. The Silhouette presenter broadcasts a synchronization message to Silhouette audiences via the Silhouette hall.

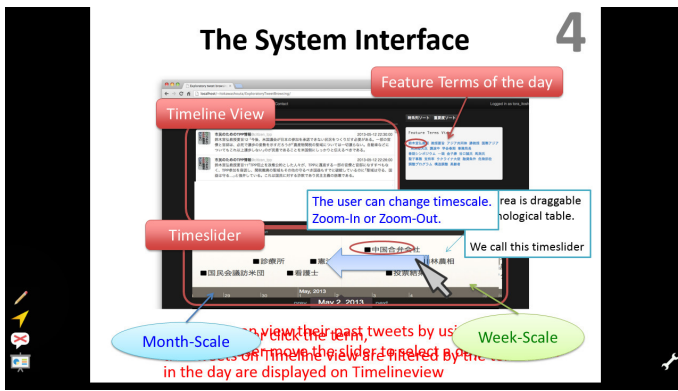


Fig. 6: Silhouette Presenter UI.

B. Silhouette Presenter

The Silhouette presenter is based on Microsoft PowerPoint 2013. We developed the system on .NET Framework 4.0 with Visual Studio 2013 C#. As for controlling PowerPoint, we use PowerPoint Object Library⁵. We describe the structure and functions of our system in this subsection.

Figure 5 shows a system diagram. The system is roughly divided into three parts. The first is the sensor part that detects the user inputs using a mouse or a keyboard. The second is the software part that processes some of the information obtained from the sensor part. The third is the screen part that reflects the results of the processing from the software part. The software part has six components. These components are for dispatching events, manipulating the slide objects, collecting the comments of the audience members, and managing the synchronization messages, the main controller, and PowerPoint. The screen part has two layers. The foreground layer displays the control menu of the system, and the background layer displays the PowerPoint slide show.

The Comment Collector is launched on a presenter's netbook device and collects comments about a presentation. The Comment Collector has three main submodules: a tweet collector for collecting tweets from Twitter, an email collector for collecting emails using IMAP, and a Web-form comment collector for collecting messages posted from a Web form.

The Message Manager sends and receives messages via WebSocket. When the presenter changes a shown slide in the slide show, the message manager creates a change message containing the slide index and sends it to the Silhouette hall. The Silhouette hall relays the messages to all the other Silhouette audiences, that is, Web browsers.

Figure 6 shows a Silhouette presenter's UI. The bottom-left icons are four function buttons. The top button, whose icon looks like a pen, is a pen function button. Push the button to draw freehand lines via the PowerPoint function. The second, whose icon looks like an arrowhead, is a direct mode function button. Push the button to through the user's mouse events to PowerPoint directly. The third button, whose icon looks like a balloon, is a comment function button. Push the button to show

⁵<http://nodejs.org/>

⁶<http://msdn.microsoft.com/en-us/library/office/ee861525.aspx>

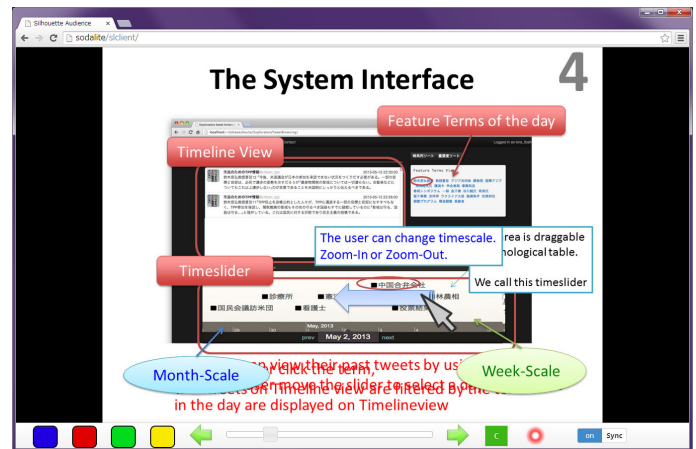


Fig. 7: Synchronization of animated slide with Silhouette Presenter in Silhouette Audience.

or hide the collected comments. The bottom button whose icon looks like a screen, is a quit function button. Push the button to quit the Silhouette presenter. The bottom-right button, whose icon looks like a wrench, is a setting function button. Push the button to show a setting dialog for the Silhouette presenter. The user can change the server address and the synchronization token in the dialog.

C. Silhouette Audience

We implemented a system called Silhouette audience that plays Web slide media. Silhouette audience can play Web slide media in a Web browser that supports HTML5. Figure 7 shows the user interface of Silhouette audience. The screen of Silhouette audience shows a presentation slide generated from a Web slide media item. To show the previous or next slide of the presentation, the user clicks the left-arrow button or right-arrow button, respectively. A slider between the left-arrow button and right-arrow button indicates the current slide position. The user changes the current shown slide by sliding a circle handle on the slider. The four colored buttons, blue, red, green, and yellow placed on the left beside the right-arrow button are used by a questionnaire feature of the Silhouette presenter. A switch placed on the right beside the left-arrow button indicates synchronization mode, whether the shown slide is synchronized with the Silhouette presenter or not.

D. Web Slide Media Maker Add-in for Presentation Software

In this section, we describe a system that builds a Web slide media item from a presentation file. The system targets a PowerPoint presentation file (PPTX) as the presentation file. The system is developed as a PowerPoint add-in. The developing environments are .NET Framework 4.0⁷ and PowerPoint Object Library (PPOL)⁸ 15.0. PPOL provides PowerPoint automation functions. Ffmpeg⁹ is a multimedia file converter. Figure 3 shows the system architecture of the add-in. The add-in builds a Web slide media item from a PPTX file. We describe the process of making a Web slide media item.

⁷[https://msdn.microsoft.com/en-us/library/vstudio/w0x726c2\(v=vs.100\).aspx](https://msdn.microsoft.com/en-us/library/vstudio/w0x726c2(v=vs.100).aspx)

⁸<https://msdn.microsoft.com/en-us/library/bb772069.aspx>

⁹<https://www.ffmpeg.org/>

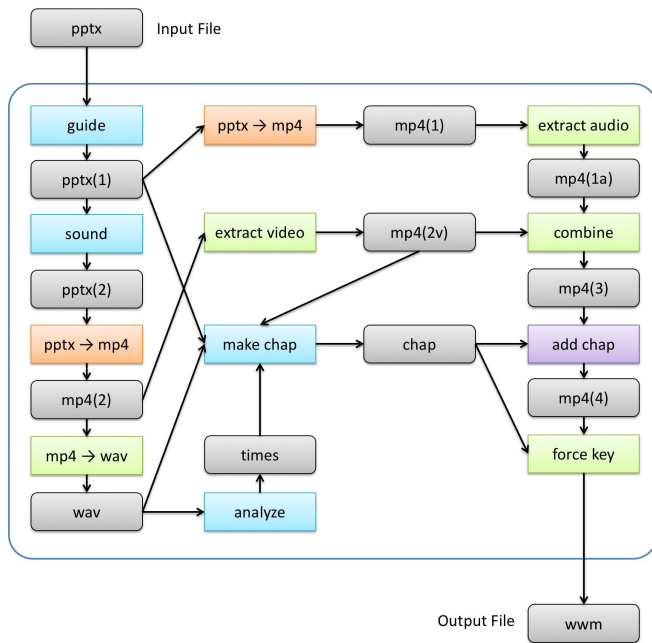


Fig. 8: Add-In Process and Data Flow of Making Web Slide Media

E. Making Web Slide Media

In this section, we describe an add-in process flow of making a Web slide media item. Figure 8 shows the process flow. Users input a pptx file into the add-in and get output as Web slide media *wwm*. A rectangle is a process. A corner-rounded rectangle is data. An arrow is a data flow. The data that points to a process is an input of the process. The data that is pointed to by a process is an output of the process. A process that has multiple inputs may start when all input data gets to be available. A blue-colored, corner-rounded rectangle is intermediate data. We describe file formats of intermediate files generated by making a Web slide media item. *pptx(1)* and *pptx(2)* are PowerPoint presentation files. *mp4(1)*, *mp4(1a)*, *mp4(2)*, *mp4(2v)*, *mp4(3)*, and *mp4(4)* are mp4 files. *wav* is an audio format wave file. *times* is time series. *chap* is chapter informations. We describe a process flow in the figure.

The process “*guide*” has an input as a pptx file and an output as a pptx file. The process inserts a sound-effect animation before each slide transition and animation activated by user-click events on the input pptx file and outputs a pptx file as *pptx(2)*. Inserting sound-effect animations uses PowerPoint functions via PPOL.

The process “*sound*” has an input as a pptx file and an output as a pptx file. The process decompresses the input pptx file using an archiver software 7za¹⁰. Next, the process modifies an XML file that contains animation information to append sound effect control information into the sound-effect that is inserted by the process *guide*. Finally, the process compresses the modified files and outputs a pptx file as *pptx(3)*.

The process “*pptx → mp4*” has an input as a pptx file and

an output as an mp4 file. The process converts the input pptx file to a mp4 file using PowerPoint functions via PPOL.

The process “*mp4 → wav*” has an input as an mp4 file and an output as a wave file. The process extracts an audio stream from the input file using ffmpeg functions and outputs a wave file. The ffmpeg executing command line is below. A string “*path/to/a.mp4*” is the input file name, and a string “*path/to/b.wav*” is the output file name. The command line means that it converts the input file to the output file where the output audio encoding is PCM little endian 16-bit encoded, the sampling rates is 44,100Hz, and the number of audio channels is 2.

```
ffmpeg -i "path/to/a.mp4" -f wav -acodec pcm_s16le -ar 44100 -ac 2 "path/to/b.wav"
```

The process “*extract video*” has an input as an mp4 file, and an output as an mp4 file. The process extracts a video stream from the input file using ffmpeg functions and outputs an mp4 file that has only a video stream and no audio stream. The ffmpeg executing command line is below. A string “*path/to/a.mp4*” is the input file name, and a string “*path/to/b.mp4*” is the output file name.

```
ffmpeg -i "path/to/a.mp4" -f mp4 -vcodec copy -an "path/to/b.mp4"
```

The process “*analyze*” has an input as a wave file and an output as a time series. The process analyzes audio wave forms from the input file and gets times when the wave forms rise up.

The process “*make chap*” has four inputs, a pptx file, an mp4 file, a wav file, and a time series $\langle t_0, t_1, \dots, t_n \rangle$, and outputs chapter information. The process gets an animation information array $\langle a_0, a_1, \dots, a_n \rangle$ from the input pptx file. Next, the process gets a video duration T_v and an audio duration T_a from the input mp4 file and the input wav file, respectively. Then, the process converts the time series $\langle t_0, t_1, \dots, t_n \rangle$ into an adjusted time series $\langle t'_0, t'_1, \dots, t'_n \rangle$ using the equation below. Finally, the process merges the animation information array $\langle a_0, a_1, \dots, a_n \rangle$ and the adjusted time series $\langle t'_0, t'_1, \dots, t'_n \rangle$ by n into chapter information.

$$t'_i = \frac{T_v}{T_a} t_i$$

The process “*extract audio*” has an input as an mp4 file and an output as an mp4 file. The process extracts an audio stream from the input file using ffmpeg and outputs an mp4 file that has only an audio stream. The ffmpeg executing command line is below. A string “*path/to/a.mp4*” is the input file name, and a string “*path/to/b.mp4*” is the output file name.

```
ffmpeg -i "path/to/a.mp4" -f mp4 -vn -acodec copy "path/to/b.mp4"
```

The process “*combine*” has two inputs as mp4 files and an output as an mp4 file. The process combines the two mp4 files into an mp4 file using ffmpeg. One of the input mp4 files has only an audio stream, and the other has only a video stream. The ffmpeg executing command line is below. A string “*path/to/a.mp4*” is the input file name that has only a video stream, a string “*path/to/b.mp4*” is the input file name that has only an audio stream, and a string “*path/to/c.mp4*” is the output file name.

```
ffmpeg -i "path/to/a.mp4" -i "path/to/b.mp4"
```

¹⁰<http://www.7-zip.org/>

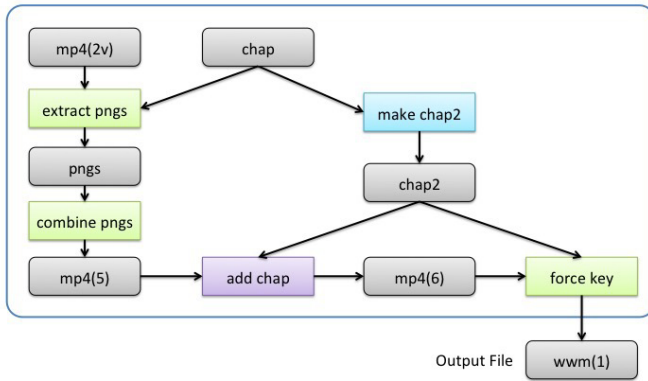


Fig. 9: Add-In Process and Data Flow of Making Sparse Web Slide Media

```
-f mp4 -vcodec copy -acodec copy "path/to/c.mp4"
```

The process “*add chap*” has two inputs, an mp4 file and chapter information, and an output as mp4 file. The process appends the chapter information to the mp4 file using MP4Box and outputs an mp4 file. The MP4Box executing command line is below. A string “*path/to/a.mp4*” is the input mp4 file name, and a string “*path/to/b.txt*” is the chapter information file name.

```
MP4Box "path/to/a.mp4" -chap "path/to/b.txt"
```

The process “*force key*” has two inputs, an mp4 file and chapter information and an output as an mp4 file. The process converts the input mp4 file with key frames that are pointed out by times included in the chapter information using ffmpeg and outputs an mp4 file. The ffmpeg executing command line is below. A string “*path/to/a.mp4*” is the input mp4 file name, and a string *<times>* is an array of a timestamp of the key frames. A timestamp of the key frames is formed by “0:00:00.000.”

```
ffmpeg -i "path/to/a.mp4" -force_key_frames
<times> -f mp4 -vcodec libx264 -r 30
-coder 1 -pix_fmt yuv420p -acodec
libvo_aacenc -ar 44100 -ac 2 "path/to/b.mp4"
```

F. Making Sparse Web Slide Media

In this section, we describe an add-in process flow of making sparse Web slide media. Figure 9 shows the flow. The process outputs a sparse animated slide using the intermediate files *mp4(2v)* and *chap* gotten from making the Web slide media. *pngs* is a set of multiple png files. *mp4(5)* and *mp4(6)* are mp4 files. *wave* is an audio wave file. *chap2* is chapter information. We describe processes and data flows in the figure.

The process “*extract pngs*” has two inputs, an mp4 file and chapter information, and outputs as multiple png files. The process extracts png files that are pointed out by the chapter information from the input mp4 file using ffmpeg. The ffmpeg executing command line is below. A string “*path/to/a.mp4*” is the input mp4 file name, a string *<time>* is an extracting timestamp, and a string “*path/to/b.png*” is the output png file name. The timestamp of the extracting frames is formed by “0:00:00.000.”

```
ffmpeg -i "path/to/a.mp4" -ss <time>
-vframes 1 -f image2 -c:v png "path/to/b.png"
```

The process “*combine pngs*” has inputs as png files and an outputs as an mp4 file. The process combines the input png files into an mp4 file using ffmpeg. The ffmpeg executing command line is below. A string “*path/to/a/%03d.png*” is the input png file name, and a string “*path/to/b.png*” is the output mp4 file name. The “*path/to/a/%03d.png*” contains a file name pattern. The pattern accepts “*path/to/a000.png*,” “*path/to/a001.png*,” “*path/to/a002.png*,” and so on.

```
ffmpeg -f image2 -framerate 5 -start_number 0
-i "path/to/a/%03d.png" -f mp4 "path/to/b.mp4"
```

The process “*make chap2*” has an input as chapter information for a Web slide media item and an output as chapter information for a sparse Web slide media item.

VI. EVALUATION

We have conducted an experiment to show that the Web presentation system proposed in section V synchronizes the slides at the same time for practical use. We measured the change in processing time and the data traffic in relation to the increase in the number of clients.

We evaluated a comparison of the file size of an original presentation file, a Web media slide file converted from the presentation file, a sparse Web media slide file generated from the Web media slide file, and an mp4 video file generated by the PowerPoint function.

A. Evaluation of Data Traffic

We evaluated the number of audience members within the range of 1 to 7, where the step of increase was at the rate of 1. We changed the slides 200 times and measured the time from when the Silhouette hall receive a change message from the presenter to when the Silhouette server received the message from the server. For experimental purposes, a Silhouette audience sent a response message after the Silhouette audience received a change message and also sent a finish message after the image was loaded by the audience.

The relevant specifications of the computer system used in the experiment are outlined below. The presenter system was performed on a MacBook Pro (Late 2013) that had an Intel Core i7 2.3-GHz CPU and a 16.0-GB DDR3 memory. The operating system running on the machine was Windows 7 Professional Service Pack 1. The server system was performed on an iMac (Mid 2010) that had an Intel Core i3 3.2-GHz CPU and an 8.0-GB DDR3 memory. The operating system running on the machine was OS X 10.9.2. The client system was an iPad Air. The operating system running on the iPad Air was iOS 7.0.4, and the Web browser was Safari. These machines were connected to a local network via an 802.11n wireless LAN adapter.

Figures 10 and 11 show the data traffic, where the number of audience members is seven, captured on the server by using Wireshark¹¹. The horizontal axis shows the time series, while the vertical axis shows the total traffic data quantity at the time. Figure 11 shows the data traffic where $85 < t < 130$ from Figure 10. The blue graph (HTTP) indicates the data traffic of Web media slides over HTTP. The red graph (WS) indicates the data traffic of synchronization messages over WebSocket. The total size of the data traffic is 32.7 MB.

¹¹<http://www.wireshark.org/>

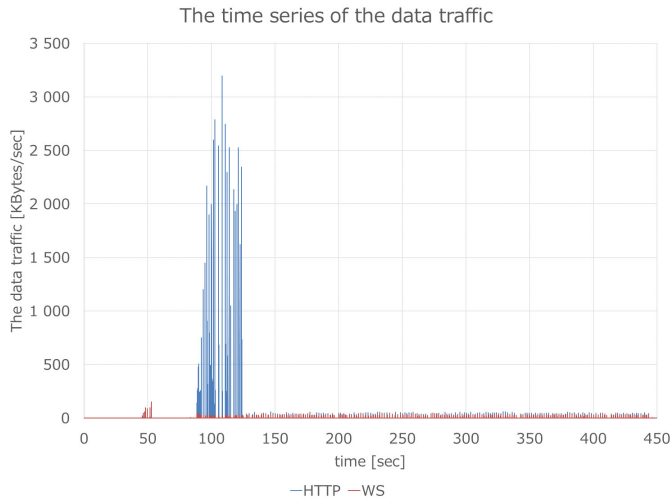


Fig. 10: Time Series of Data Traffic.

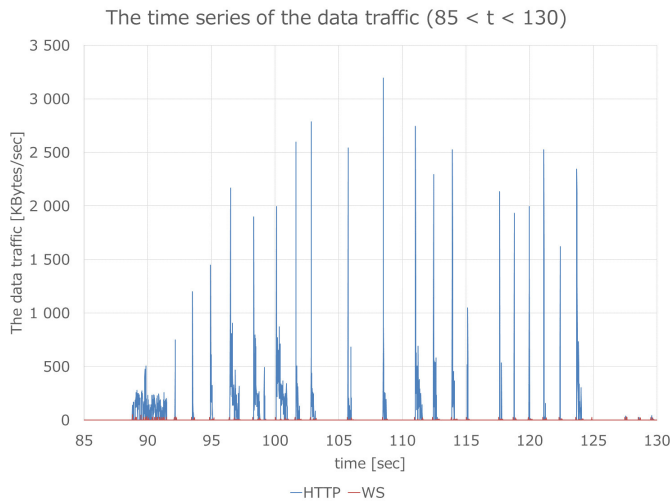


Fig. 11: Time Series of Data Traffic (85 < t < 130).

B. Evaluation of Making Web Slide Media

Table I shows the comparison of file size of Web slide media. pptx indicates the original presentation file. file1 has no animations and no slide transitions. file2 has animations and slide transitions. file3 has slide transitions but no animations. file1, file2, and file3 have the same images, shapes, and text. file3 is equal to file2, which excludes all animations. file1 is equal to file3, which excludes all slide transitions. We compared four types of videos: an original mp4 file (mp4), a Web slide media file (wwm), a sparse Web slide media file (wwm2) and a Web slide media file that excluded all slide transitions.

The mp4 video file was generated by PowerPoint with the parameter of a video frame resolution of 720 pixels. wwm, wwm2, and wwm3 were generated by the add-in with parameters of a video frame resolution of 720 pixels and a video frame rate of 10 frames per second. Generally, slide transitions increase the file size of a wwm. Therefore, we also

TABLE I: Comparison of File Size

Type	mp4	wwm	wwm2	wwm3	pptx
file1	1,618	2,260	928	2,260	4,693
file2	44,866	10,443	3,235	5,689	4,729
file3	43,788	8,303	1,743	2,260	4,724

TABLE II: Comparison of Rate of Change of File Size

Type	mp4	wwm	wwm2	wwm3
file1	34.5%	48.2%	19.8%	48.2%
file2	948.7%	220.8%	68.4%	120.3%
file3	926.9%	175.8%	36.9%	47.8%

made a Web slide media file that excluded all slide transitions of wwm3. It seems that slide transitions significantly increase the file size of a wwm, and excluding slide transitions helps to reduce the file size. In Table I, slide transitions of file3 are disabled because the wwm file size of file3 is equal to that of file1. Table II shows the rate of change of file size between an original presentation file and a Web slide media file. Table III shows the conversion times of making a Web slide media file from an original presentation file. Note that, in Table III, the times of wwm2 are the making times of a sparse Web slide media file from a Web slide media file. Therefore, the actual making time of a sparse Web slide media is the wwm2 time added to the wwm time. And the wwm2 making time is smaller than the one that, excluded slide transitions.

VII. DISCUSSION

A Silhouette audience sends a request to the Silhouette hall to download an image file each time the audience receives a notification of a slide change. The data traffic after 125 seconds is very low (less than 9.805 KBytes/sec) because the Web browser uses an HTTP If-None-Match header and an HTTP If-Modified-Since header. Moreover, the Silhouette hall reduces the data traffic by HTTP 304 response code if the requested image file is not changed. In Figure 11, the pulses between 40 seconds and 50 seconds occurred when the Silhouette audiences logged in. The Silhouette presenter sent the Web media slides sequentially because the pulses between about 88 seconds and 93 seconds continuously occurred. The pulses between 93 seconds and the end occurred when the Silhouette presenter sent synchronization messages to the Silhouette hall.

Our system is easy to configure for presenters and audience members to synchronize a current slide with a few settings. The following three operations are needed for the synchronization of slides. First, a presenter launches a Silhouette presenter system. Second, the presenter sets a Silhouette hall URL and a specified synchronization token. Third, an audience member accesses the Silhouette audience in a Web browser and logs in with the same token. After the set up, the presenter opens a presentation file and begins the presentation as usual in a PowerPoint slide show. After that, the synchronization of shown slides starts.

The ease of operation is implemented by the following procedures. The Silhouette presenter monitors whether a presentation slide show begins or not. The Silhouette presenter converts the slides to images and sends them to the Silhouette

TABLE III: Comparison of Making Times

Type	mp4	wwm	wwm2	wwm3
file1	2.3	45.4	10.6	46.6
file2	64.9	215.3	178.4	163.6
file3	26.3	103.9	36.4	45.2

hall when the presentation slide show begins. The displayed slide on the screen of an audience member is synchronized with the presenter. Moreover, a Silhouette audience is executable in a Web browser supporting HTML5. Since the system does not require special plug-ins due to the use of HTML5, it is easy to introduce and use.

Furthermore, the rich shape animations and effects of the existing presentation software, PowerPoint, are used without special settings. All users use their own PowerPoint program with the Silhouette presenter system. When users begin the slide show mode with their PowerPoint, the Silhouette presenter system automatically connects to the Silhouette hall system and the server system and uploads the slide images of the current slide. The audience members browse the slides by using their Web browsers.

Generating HTML snippets enables users to reuse the slides and share them on their Web pages. Users share reusable and re-editable presentation slides for the Web using SlideStack, which was developed in our laboratory. One of the key points of SlideStack is that users control its quality and controllability. When users want to share high-quality slides, they obtain Web slide media, which are animated and controllable Web slides. Web slide media is a new presentation slide format based on MPEG-4 with chapter information. The chapter information contains timing information of the beginning and end of slide transitions and shape animations. Users only need to install the add-on for creating Web slide media and to push the button of the add-on. Then the add-on system generates and upload a special movie containing control information from a current slide.

Existing methods of converting a presentation file into a set of image files, a PDF file, or an HTML file enable a program code to manipulate a shown slide and play back animations. The files, however, have appearance problems regarding animation effects. PowerPoint has a function to output a presentation file as a video file, but the video file has no chapter information. Thus, a video player makes a user seek the slide that the user wants until the slide is shown. We implemented a function that converts a presentation file to a video file and has chapter information. The chapter information indicates when a slide transition effect or a shape animation effect begins in the time sequence of the video. Generally, the Web slide media file size is larger than the original presentation file or the video file. To reduce the file size, we also implemented a function that removes all shape animation effects and slide transition effects from the Web slide media.

VIII. CONCLUSION

We implemented a PowerPoint add-in that builds a Web slide media file from a PowerPoint presentation file. The Web slide media has chapter information. We also implemented a

video player that can stop the video playback using the chapter information when an animation effect reaches the end. The video player can run in Web browsers that support HTML5. The Web slide media file size is larger than the original presentation file or the video file. To reduce the file size, we also implemented a function that removes all shape animation effects and slide transition effects from the Web slide media. We called the reduced video sparse Web slide media. We evaluated the file sizes of Web slide media files and sparse Web slide media files. The evaluations show that the file sizes are effective for practical use.

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Applying data mining in the context of Industrial Internet

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Abstract—Nowadays, (industrial) companies invest more and more in connecting with their clients and machines deployed to the clients. Mining all collected data brings up several technical challenges, but doing it means getting a lot of insight useful for improving equipments. We define two approaches in mining the data in the context of Industrial Internet, applied to one of the leading companies in shoe production lines, but easily extendible to any producer. For each approach, various machine learning algorithms are applied along with a voting system. This leads to a robust model, easy to adapt for any machine.

Keywords—*machine learning; data mining; k-nearest neighbour; neural network; support vector machine; rule induction.*

I. INTRODUCTION

According to [1], the access to the Internet has grown from an estimated 10 million people in 1993, to almost 40 million in 1995, to 670 million in 2002, and to 2.7 billion in 2013. The Internet has started to be used so much in industry, that a consortium has been founded, called Industrial Internet Consortium, which covers energy, healthcare, manufacturing, public sector and transportation (online at www.iiconsortium.org). According to the Industrial Internet Insights Report for 2015 [2], issued by General Electric and Accenture, the Industrial Internet can be described as a source of both operational efficiency and innovation that is the outcome of a compelling recipe of technology developments. However, the companies are not ready for predictive and innovative kinds of value-creating solutions, but 80% to 90% of the surveyed companies indicated that bug data analytic is in the top three priorities.

Therefore the article presents a specific application of data mining in the context of Industrial Internet. That means that some industrial equipments, wearing several sensors, are connected to the internet and the data can be analysed either locally or in a cloud. The target is to predict a specific behaviour based on the inputs from the sensors.

We treat the specific case of KLÖCKNER DESMA Schuhmaschinen GmbH (Germany), which is a leading vendor of machine systems for automated shoe production. An export rate of more than 95% results in a globally distributed customer and business partner network. Up-to-date, communication and information exchange with customers and business partners is mainly done in personal conversations almost inhibiting an efficient systematic analysis of customer feedback or product experience. By establishing an Industrial Internet connecting

customers, partners and products DESMA extends its leading position in the market but also targets to explore new business fields in data analysis and provision.

The article is structured as follows. After discussing the related work in section II, the general concepts of data mining in section III, which means applying machine learning (presented in section III-A) on big data. The specific algorithms used are described in sections III-A1 through III-A8, respectively III-B. The experimental setup and the results are detailed in sections IV and V.

II. RELATED WORK

The field of internet connectivity (along with Internet of Things and Industrial Internet) is at the start of the research because until recently there were no circumstances to do it (see [3], [4], [5]). As the Internet has deeply penetrated the domestic and industrial fields, more and more companies want to know how their equipments work. Therefore, in 2004, Olsson et al. [6] did fault diagnosis using case-based reasoning on sensor readings. A more statistical approach was developed by Giudici in [7]. One year later, Harding et al. publishes in [8] an overview about the possibilities of applying data mining in manufacturing. Many application are developed in the context of quality assurance in industry, such as the ones by Braha [9] and Koeksal [10]. Almagrabi et al. [11] publish in 2015 a very good survey on the quality prediction of product review.

III. DATA MINING PROCESS

The data mining process consists in collecting the data from a data source, preprocessing it and then applying a machine learning algorithm [12]. The inputs come from the sensors of a specific equipment, in our case direct soling machines, automated material flow with integrated robots (AMIR), robots and automatic laser processing cells. For more details, see section IV. The output is usually a desired outcome of the system (either an action of the equipment, or a warning etc.).

In our work, several machine learning algorithms have been applied, as described further. However, for improving the accuracy of the process, a voting system is also used. This means that the outputs of all algorithms vote for the most probable outcome. If it is a classification, the voting result is the class generated by most particular algorithms; if it is a regression, the global outcome is the average of each specific output. The strong point of this voting system is that

more approaches (algorithms) are used for the same purpose, therefore the result is expected to be more robust.

A. Machine learning algorithm

In our experiments we have used several algorithms along with their combinations. Only the ones with significant accuracy are presented in this paper, the other one being discarded because of the lack of interest and efficiency. Therefore, eight algorithms have some importance, such as:

- neural networks;
- Naive Bayes;
- support vector machine (SVM);
- fast large margin;
- k-nearest neighbour (k-NN);
- logistic regression;
- random forest;
- rule induction.

1) *Neural networks*: We apply the most common used model of neural networks, namely a multi-layer perceptron, which is a feed-forward neural network trained by a back propagation algorithm [13].

The learning is made using the back propagation algorithm, which is a supervised method that can be divided into two phases: propagation and weight update [14]. The two phases are repeated until the performance of the network is good enough. In back propagation algorithms, the output values are compared with the correct answer to compute the value of some predefined error-function. The error is then fed back through the network. Using this information, the algorithm adjusts the weights of each connection in order to reduce the value of the error function by some small amount. After repeating this process for a sufficiently large number of training cycles, the network will usually converge to some state where the error of the calculations is small [15].

For activation of the neurons, the sigmoid function is the most commonly used. Therefore, the values ranges of the attributes should be scaled to -1 and +1. This can be done through the normalize parameter. The type of the output node is sigmoid if the learning data describes a classification task and linear if the learning data describes a numerical regression task.

2) *Naive Bayes*: According to Lewis [16], a Naive Bayes classifier is a simple probabilistic classifier based on applying Bayes' theorem ([17]) with strong (naive) independence assumptions. In simple terms, a Naive Bayes classifier assumes that the presence (or absence) of a particular feature of a class (i.e. attribute) is unrelated to the presence (or absence) of any other feature.

The main advantage of the Naive Bayes classifier is that it requires a small amount of training data to estimate the means and variances of the variables necessary for classification. This is important because quite often the learning process is done on a very limited samples. Because independent variables are assumed, only the variances of the variables for each label need to be determined and not the entire covariance matrix.

3) *Support Vector Machine*: The support vector machine (SVM) is a fast algorithm with good results, which can be used for both regression and classification. Several kernel types include dot, radial, polynomial, neural, ANOVA, Epachnenikov, Gaussian combination and multiquadric [18].

An SVM model is a representation of the examples as points in space, mapped so that the examples of the separate categories are divided by a clear gap that is as wide as possible. New examples are then mapped into that same space and predicted to belong to a category based on which side of the gap they fall on.

More formally [19], a support vector machine constructs a hyperplane or set of hyperplanes in a high- or infinite-dimensional space, which can be used for classification, regression, or other tasks. Intuitively, a good separation is done by the hyperplane that has the largest distance to the nearest training data points of any class. It is often the case that the sets to discriminate are not linearly separable in a lower dimension space. For this reason, it was proposed that the original finite-dimensional space be mapped into a much higher-dimensional space, presumably making the separation easier in that space [20].

4) *Fast Large Margin*: Although the fast large margin provides results similar to those delivered by classical SVM or logistic regression implementations, this classifier, implemented as proposed by Fan et al in [21], is able to work on data set with millions of examples and attributes.

5) *k-Nearest Neighbour*: The k-Nearest Neighbor algorithm is based on learning by analogy, that is, by comparing a given test example with training examples that are similar to it [27]. The training examples are described by n attributes. Each example represents a point in an n -dimensional space. In this way, all of the training examples are stored in an n -dimensional pattern space. When given an unknown example, a k-nearest neighbor algorithm searches the pattern space for the k training examples that are closest to the unknown example. These k training examples are the k "nearest neighbors" of the unknown example. "Closeness" is defined in terms of a distance metric, such as the Euclidean distance [22].

The k-nearest neighbor algorithm is one the simplest of all machine learning algorithms: an example is classified by a majority vote of its neighbours, with the example being assigned to the class most common amongst its k nearest neighbours (k is a positive integer, typically small).

6) *Logistic Regression*: The logistic regression is based on the algorithm proposed by Keerthi et al. in [23]. The implementation uses the myKLR by Stefan Rueping [24]. Like most other SVM approaches, this one supports various kernel types including dot, radial, polynomial, neural, anova, epachnenikov, Gaussian combination and multiquadric.

7) *Random Forest*: The Random Forest algorithm generates a set of a specified number of random tree models. The number of trees parameter specifies the required number of trees. The resulting model is a voting model of all the random trees [26].

According to Safavian and Landgrebe [25], the representation of the data in form of a tree has the advantage

compared with other approaches of being meaningful and easy to interpret. The goal is to create a classification model that predicts the value of a target attribute based on several input attributes of the training set. Each interior node of the tree corresponds to one of the input attributes. The number of edges of a nominal interior node is equal to the number of possible values of the corresponding input attribute. Outgoing edges of numerical attributes are labelled with disjoint ranges. Each leaf node represents a value of the label attribute given the values of the input attributes represented by the path from the root to the leaf.

Pruning is a technique in which leaf nodes that do not add to the discriminative power of the tree are removed [30]. This is done to convert an over-specific or over-fitted tree to a more general form in order to enhance its predictive power on unseen datasets. In other words, pruning helps generalization.

8) *Rule Induction*: The Rule Induction operator works similar to the propositional rule learner named 'Repeated Incremental Pruning to Produce Error Reduction' [28]. Starting with the less prevalent classes, the algorithm iteratively grows and prunes rules until there are no positive examples left or the error rate is greater than 50% [29]. In the growing phase, for each rule greedily conditions are added to the rule until it is perfect (i.e. 100% accurate). The procedure tries every possible value of each attribute and selects the condition with highest information gain. In the prune phase, for each rule any final sequences of the antecedents is pruned with the pruning metric $p/(p+n)$.

B. Voting

For a more accurate result, a voting algorithm has been also applied. It uses a majority vote of all other algorithms. The results with most votes is the output of this algorithm.

The process of voting is depicted in figure 1.

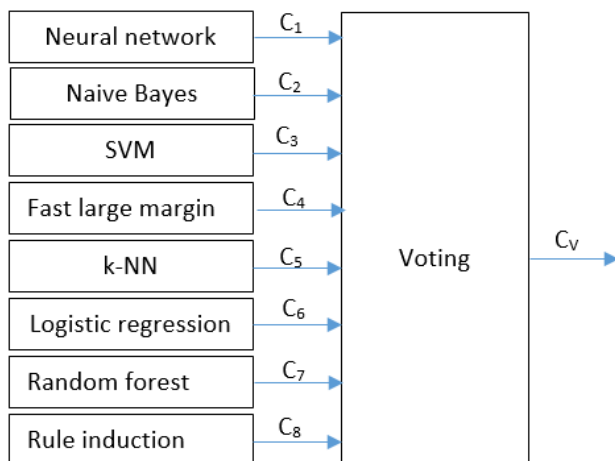


Fig. 1: The voting process

The result C_v of the voting is the result of the most of the specific outputs C_1, C_2, \dots, C_8 .

IV. EXPERIMENTAL SETUP

The experiments have been carried out on a set of simulated samples provided by one of the world leading manufacturer of

footwear production systems DESMA to validate the approach in a first step.

The data for the simulation is from following sensors which are representing a detailed status of a DESMA system or system component.

- Status (on/off) – showing the on/off status of a system or a system component;
- Actor Positions - represents the positions of actors inside of a system or a system module;
- Speed – is the configured system speed;
- Revolution speed - means the rotational speed of e.g. motors;
- Duration time - if the time of scheduled tasks;
- Temperature – shows material, ambient or system (component) temperatures;
- Liquid pressure - represents the pressure of liquid for the sole production;
- Air pressure - show the air pressure in the pipes.

The format of the data along with some sample values is described in table I.

These sensors are integrated in DESMA systems and system components as:

- Direct soling machines which directly injected the sole to the upper for single- or dual-density applications for the materials of rubber, polyurethan (PU), thermoplastic polyurethane (TPU) and other thermoplastics.
- Automated material flow with integrated robots (AMIR) systems which is an automation concept developed by DESMA used today in advanced footwear factories.
- Robots integrated in an automated production process
- Automatic Laser Processing cells which are roughing upper and offers new design opportunities in the direct soling technology.

Data mining results through (continuous) analyses of such sensor data of systems and system components as data base are useful to improve and optimizes company individual KPIs and are useful to improve extended system services as analysis, diagnosis and advanced monitoring of such systems.

A. Correlation matrix

The first important thing is to analyse the correlation between the attributes. Although this does not influence the behaviour of the algorithm, it is good to see if the inputs are dependent of each other. If so, some may be skipped. That would reduce the number of inputs, thus also the complexity of the system, increasing also the quality of the output.

As easily seen in table II, some fields are fully correlated, such as Status (on/off) with Speed, Duration time and Status air pressure. Other fields, such as Status (on/off) and Revolution speed or

TABLE I: A sample of the data set

Date	Status (on/off)	Actor Position	Speed	Revolution speed	Duration time	Temp.	Liquid pressure	Air pressure	Article number
10.9.15 10:17:01	0	0	0	0	0	25	0	0	6565
10.9.15 10:17:10	1	90	10	72	1	40	1	1	6565
10.9.15 10:18:04	1	50	10	74	1	41	10	1	6565

TABLE II: The correlation matrix

	Status (on/off)	Actor position	Speed	Revolution speed	Duration time	Temperature	Liquid pressure	Air pressure
Status (on/off)	1	0.399	1	0.763	1	0.586	0.406	1
Actor position	0.399	1	0.399	0.408	0.399	0.436	0.125	0.399
Speed	1	0.399	1	0.763	1	0.586	0.406	1
Revolution speed	0.763	0.408	0.763	1	0.763	0.762	0.352	0.763
Duration time	1	0.399	1	0.763	1	0.586	0.406	1
Temperature	0.586	0.436	0.586	0.762	0.586	1	0.241	0.586
Liquid pressure	0.406	0.125	0.406	0.352	0.406	0.241	1	0.406
Air pressure	1	0.399	1	0.763	1	0.586	0.406	1

Revolution speed with Speed and Duration time, are significantly correlated.

However, the desired output, namely Actor position, has rather low correlation with any of the attributes, as its highest correlation factor is 0.408.

Such observations are a good starting point for a data mining process, because otherwise the process itself does not make much sense.

V. EXPERIMENTAL RESULTS

Two sets of experiments have been carried out:

- one determining the alert temperature, which means determining if the temperature is more than 50 degrees, described in section V-A.
- one determining the temperature trend. If the trend is ascending and the temperature is above 50 degrees, an alert is emitted (as presented in section V-B).

Of course, in both set-ups, the temperature is neither an input nor an output. In the former case, the value of the logical expression $Temperature > 50$ is the output, whereas in the latter scenario, the trend of the temperature is the desired outcome of the data mining process.

A. Predicting the temperature

In this scenario, the temperature is the output of the data mining process.

The approaches with good outcomes are: neural networks, SVM, k-NN, logistic regression, random forest and rule induction. The accuracy for each of them is presented in table III. The first column displays the approach, the second column the accuracy and the third one displays whether the accuracy is above or below the average, within or out of the standard deviation. In both tables III and V, the notations are as follows:

- "(+)" if the accuracy is above the average and within the standard deviation ($accuracy \in [avg, avg + stdev]$);

TABLE III: Computational results for predicting the temperature

Method	Accuracy	
Neural nets	98.9%	(+)
SVM	96.7%	(+)
k-NN	98.9%	(+)
Logistic regression	7.8%	(-)
Random forest	95.6%	(+)
Rule induction	92.2%	(+)
Average	81.68%	
Stdev.	33.11%	

- "(++)" if the accuracy is above the average and outside the standard deviation ($accuracy > avg + stdev$);
- "(-)" if the accuracy is below the average and within the standard deviation ($accuracy \in [avg - stdev, avg]$);
- "(--)" if the accuracy is below the average and outside the standard deviation ($accuracy < avg - stdev$);

There are two important remarks related to the results:

- most of the results are very good (above 92%), which means that determining the temperature is very accurate and reliable;
- the accuracy in the case of logistic regression is extremely low (below 8%).

As the accuracies are so high, no further enhancements are considered.

B. Predicting the trend

In this scenario, the temperature can be measured, but it is not an input of the data mining process. The outcome is the trend of the temperature, which, corroborated with the value of the temperature will raise an alert if the temperature is above 50 degrees and the trend is upwards. The dataset looks like in table IV, where Temp. trend is the trend of the temperature to be predicted.

The eight approaches led to the following results regarding the accuracy of classification:

TABLE IV: A sample of the data set when predicting the temperature trend

Date	Status (on/off)	Actor Position	Speed	Revolution speed	Duration time	Temp. trend	Liquid pressure	Air pressure	Article number
10.9.15 10:17:01	0	0	0	0	0	up	0	0	6565
10.9.15 10:17:10	1	90	10	72	1	up	1	1	6565
10.9.15 10:18:04	1	50	10	74	1	up	10	1	6565

TABLE V: Computational results

Method	Accuracy		Optimized accuracy		Increase
Neural nets	52.9%	(+)	57.67%	(+)	9.02%
Naive Bayes	55.1%	(+)	62.14%	(+)	12.78%
SVM	56.9%	(++)	58.67%	(+)	3.11%
Fast large margin	51.5%	(-)	51.56%	(-)	0.12%
k-NN	48.2%	(-)	51.56%	(-)	6.97%
Logistic regression	57.1%	(++)	66.89%	(++)	17.15%
Random forest	48.6%	(-)	58.67%	(+)	20.72%
Rule induction	51.8%	(-)	46.56%	(-)	-10.12%
Average	52.76%		56.72%		7.49%
Stdev.	3.42%		6.52%		

In table V, the first column represents the machine learning algorithm used for data mining. The second column displays the accuracy of each algorithm and then their relative performance. The results marked with "(+)" are above the average, and the ones marked with "(++)" are above the average plus the standard deviation. Pairwise, the results marked with "(-)" are below the average and the ones marked with "(-)" are below the average minus the standard deviation. The next column represents the accuracy of each approach, after a parameters optimization process, along with the relative performance. The last column shows the improvements brought by the optimization, which is, in average, 7.49%. To be noticed that only one algorithm namely Rule induction, reduced its accuracy (with 10.12%), while all the others improved up to 17.15% in the case of logistic regression and 20.72% for random forest.

Determining the trend did not lead to impressive levels of accuracy and this is somehow disappointing. This is because of the low correlation between the input attributes and the temperature trend. In other words, we could not find a significant correlation between the trend and the other attributes.

1) *Best parameters:* It is of importance to show what parameters of the learners lead to the best results. Therefore they are described in table VI.

2) *Voting results:* Voting for between all previous data mining techniques, brings the accuracy of the system 59.6%, which is less than the best two techniques running separately, namely Naive Bayes (with an accuracy of 62.14%) and logistic regression (with an accuracy of 66.69%). Even the optimized techniques with respect to parameters has not brought any improvement. However, voting accuracy is slightly higher than the average accuracy of the six methods (neural network, Naive Bayes, SVM, fast large margin, k-NN and logistic regression), which is 58.081%.

Voting all eight methods lead to an accuracy of 51.4%, which is much less than the average accuracy of each separate

TABLE VI: The best parameters

Algorithm	Parameter	Value
Neural network	hidden layers	0
	training epochs	500
	learning rate	0.64
	momentum	0.0
Naive Bayes	estimation method	greedy
	number of kernels	10
	bandwidth	0.2
	minimum bandwidth	2.4
	number of kernels	10
SVM	kernel type	rbf
	γ	19.0
	C	0.0
Fast large margin	ϵ	0.001
k-NN	k	1
	numerical measure	correlation similarity
Logistic regression	kernel	radial
	kernel γ	5.0
	C	4.7
Random forest	number of trees	100
	maximal depth	46
	minimal leaf size	4
Rule induction	criterion	accuracy
	sample ration	0.9
	pureness	0.9
	minimal prune benefit	0.25

technique of 56.715%.

The very interesting thing is that removing the poorest quality method, which is rule induction (accuracy of 46.56%), the voting approach leads to an accuracy of 46.8%, thus even lower than using also rule induction.

VI. CONCLUSIONS

This article presents a data mining approach in the context of Industrial Internet. The specific studied case is a KLÖCKNER DESMA Schuhmaschinen GmbH, which is the leader company in shoe production lines. Their machines are equipped with several sensors and the collected data can be transferred into the cloud. Analysing the data thoroughly brings a lot of insight and useful knowledge about the usage of the machines, which can be further utilized for predictive maintenance and for improving the user experience.

The achieved accuracy is neither very high, nor very low. However, the experimental data set was pretty limited and the scope of the research was to achieve a robust and very easily extendible model. In this case, the accuracy will depend on the input/output data set. Therefore, the level of accuracy is not the most dominant here, if it is above some certain reasonable thresholds.

The implemented model and very robust and can be easily extended to other companies collecting data from their machines. We also show that two different approaches for

mining the same data set, lead to completely different results and accuracies.

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Complex-Valued Neural Networks Training: A Particle Swarm Optimization Strategy

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Abstract—QSAR (Quantitative Structure-Activity Relationship) modelling is one of the well developed areas in drug development through computational chemistry. This kind of relationship between molecular structure and change in biological activity is center of focus for QSAR modelling. Machine learning algorithms are important tools for QSAR analysis, as a result, they are integrated into the drug production process. In this paper we will try to go through the problem of learning the *Complex-Valued Neural Networks*(CVNNs) using *Particle Swarm Optimization*(PSO); which is one of the open topics in the machine learning society where the CVNN is a more complicated for complex-valued data processing due to a lot of constraints such as activation function must be bounded and differentiable at the complete complex space. In this paper, a CVNN model for real-valued regression problems is presented. We tested such trained CVNN on two drug sets as a real world benchmark problem. The results show that the prediction and generalization abilities of CVNNs is superior in comparison to the conventional real-valued neural networks (RVNNs). Moreover, convergence of CVNNs is much faster than that of RVNNs in most of the cases.

Keywords—Particle Swarm Optimization, Complex-Valued Neural Networks, QSAR, Drug Design, prediction.

I. INTRODUCTION

The problem of drug design is to find drug candidates from a large collection of compounds that will bind to a target molecule of interest. The development of a new drug is still a challenging, time-consuming and cost-intensive process and due to the enormous expense of failures of candidate drugs late in their development. Designing 'drug-like' molecules using computational methods can be used to assist and speed up the drug design process [1], [2], [3]. The major bottlenecks in drug discovery were addressed with computer-assisted methods, such as QSAR models [4], where the molecular activities are critical for drug design. The QSAR models used to predict the drug activity within a large number of chemical compounds using their descriptors that are often generated with high-noise in high-dimensional space. Nowadays, *machine learning* algorithms have been used in the modelling of QSAR problems [5], [6], [7]. They extract information from experimental data by computational and statistical methods and generate a set of rules, functions or procedures that allow them to predict the properties of novel objects that are not included in the learning set. Formally, a learning algorithm is tasked with selecting a hypothesis that best supports the data. Considering the hypothesis to be a function f mapping from the data space X to the response space Y ; i.e., $f : X \rightarrow Y$. The learner

selects the best hypothesis f^* from a space of all possible hypotheses \mathcal{F} by minimize errors when predicting value for new data, or if our model includes a cost function over errors, to minimize the total cost of errors. As shown in Fig. 1, the QSAR modelling is heavily dependent on the selection of molecular descriptors; if the association of the descriptors selected to biological property is strong the QSAR model can identify valid relations between molecular features and biological property/activity. Thus, uninformative or redundant molecular descriptors should be removed using some feature selection methods during (*filters*) or before (*wrappers*) the learning process. Subsequently, for tuning and validation of the predictively of learned QSAR model, one of the validation strategy can be applied likes cross-validation, leave-one-out or the full data set is divided into a training set and a testing set prior to learning (See [8] for a survey).

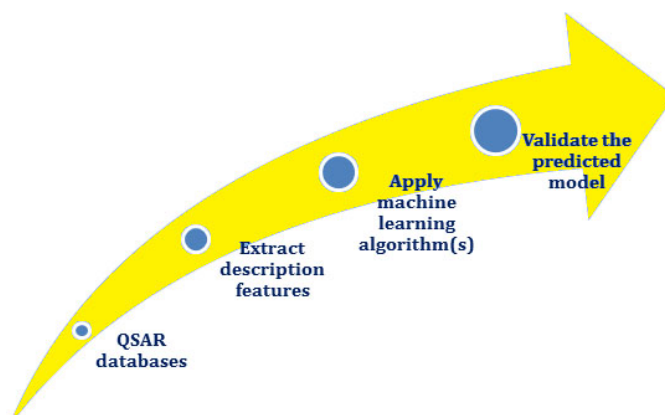


Fig. 1: General steps of developing QSAR models

Actually, the machine learning field [9], [10], [11] have versatile methods or algorithms such as decision trees (DT), lazy learning, k -nearest neighbours, Bayesian methods, Gaussian processes, artificial neural networks (ANN), support vector machines (SVM), and kernel algorithms for a variety of tasks in drug design. These methods are alternatives to obtain satisfying models by training on a data set. However, the prediction from most regression models - be it multiple regression, ANN, SVM, DT, etc. is a point estimate of the conditional mean of a response (i.e., quantity being predicted), given a set of predictors. currently, the complex numbers are very actively used in modern engineering and in modern physics, to

describe real-life phenomena which allow *feasible predictions* and *efficient control strategies*. Operations such as integration and optimization are only *feasible* when the corresponding functions can be extended to smooth functions in a complex domain and consequently feasible algorithms become possible. Moreover, sometimes, even in the situations when a real-valued feasible algorithm is possible, the use of complex numbers can speed up computations. A typical example of such a situation is the use of *Fast Fourier Transform*(FFT). In recent years, complex-valued neural networks (CVNNs) have widened the scope of application in telecommunications, imaging, remote sensing, time-series, spatio-temporal analysis of physiological neural systems, and artificial neural information processing. Also, multi-valued neural network is a special type of CVNN, its has a threshold function of multi-valued logic and complex-valued weights is considered [12], [13], [14]. The CVNN which has complex number processing structure that made the network have stronger learning ability, better generalization ability [15], superior reducing power [16], faster convergence [17], lower computational complexity and less data is needed for network training. There are many previously successful attempts to implement the PSO in generating QSAR models. One of the attempts, is the using of Binary PSO for feature selection followed by a neural network which is trained using back propagation (BP) for the construction of the QSAR models [18], [19]. The major disadvantage of BPSO-BP is the difficulty in choosing parameters for the *back-propagation* that can ensure efficient network training. Recently, [20] uses the PSO for training the RVNN to overcome such drawbacks. The main objective of this work is to exploit the complex-valued characteristic by apply the CVNN model that trained by PSO for real world forecasting problem (e.g. drug design). The key contributions of this paper are concluded as follows:

- We propose a new strategy for training the CVNN using PSO in QSAR modelling for drug design.
- We evaluate the proposed strategy by determine its prediction accuracy and convergence rate for real data through experimental results.

The rest of the paper is organized as follows. Section II presents the related work for training ANN and complex-valued NN. In sections III and IV an introduction to the CVNN and PSO algorithm are presented. Section V briefly introduces the problem formulation. Section VI describes the data set, their descriptions and processing step. Section VII describes an evaluation of QSAR modelling and prediction results using the proposed strategy. Finally Section VIII presents the findings and conclusions.

II. RELATED WORK

The training of neural networks is still one of the main issues in the machine learning researches. There are many different training algorithms (i.e. optimization methods) that are proposed and implemented, from gradient descent optimization to evolutionary algorithm and swarm optimization [21], [22], [23]. Several works that use evolutionary and swarm strategies for training ANNs have been reported in the literature. Wang *et. al.*, [24] combine PSO and ANNs for building QSAR models. The ANN structure is constructed using neurons, the functionality and limitations of these single neurons is

determined mainly by their activation function [21]. In general, the functionality of single neurons can be significantly increased by using complex-valued activation functions, which must have all the properties of a real-valued neuron (i.e. boundness, differentiability of errors etc.). The problem is that due to a lot of constraints this does not held in complex-valued case [25]. Aizenberg and Moraga [12], proposed the multi-valued neural network as a special type of CVNN, its has a threshold function of multi-valued logic and complex-valued weights is considered. They have also been successfully applied to associative memory tasks, classification tasks, to signal processing or signal generation [13] and prediction problem [14]. Alexandre [26] presents the adaptation of a single layer CVNN using entropy in the cost function instead of the usual mean squared error (MSE). Nitta [27], developed the *natural gradient descent method* for the multilayer stochastic complex-valued neural networks, and derived the natural gradient for a single stochastic complex-valued neuron. Usually, the multilayer CVNN is trained using the gradient descent learning method [28]. These CVNN used for time series forecasting utilizing the speed and direction of the for a wind power generation system [29]. The wind speed and direction were represented as a complex number on the complex co-ordinates for generating an input to the CVNN. The CVNN was trained using a complex back-propagation algorithm during training for predictions. Zimmermann *et.al.*, [30] combining the global (random search algorithm) and local optimization algorithm (gradient descent method) to calculate gradients of the CVNN.

III. COMPLEX-VALUED NEURAL NETWORKS

The CVNNs are simply the generalization of the RVNN in the complex valued domain as shown in Fig. 2, where all the parameters including weights, biases, inputs, outputs could be complex variables, and the activation function and its derivatives have to be well behaved everywhere in the complex plane. The complex back-propagation (BP) algorithm, which is the complex-valued version of the real valued back propagation algorithm, is widely used to train the CVNNs [28], [31]. Let us take the CVNN which has n inputs, m neurons in the hidden layer and k outputs. The o networks output could be calculated as follows:

$$O_k = f(B_k + \sum_{i=1}^m W_{km} \times H_m) \quad (1)$$

and the output of each hidden neuron m is given like:

$$H_m = f(B_m + \sum_{i=1}^n W_{mn} \times X_n) \quad (2)$$

Where X_n complex valued input and B_k, W_{km}, B_m, W_{mn} are biases and the weights from the input to hidden and from the hidden to the output layers, respectively. $f(\cdot)$ is a complex valued activation function. According to the *Liouville's theorem*, in which the analytic and bounded functions on entire complex plane are constant, the $f(\cdot)$ takes a great attention and several complex activation functions proposed in the literature [31]. In this paper the split sigmoid function is taken and it is given as follows:

$$f = \frac{1}{1 + e^{-Re(z)}} + j \frac{1}{1 + e^{-Im(z)}} \quad (3)$$

Where $z = x + jy$. It should be noted that the use of the split sigmoid function rather than the non-split function could avoid the problem of functions singularity [31].

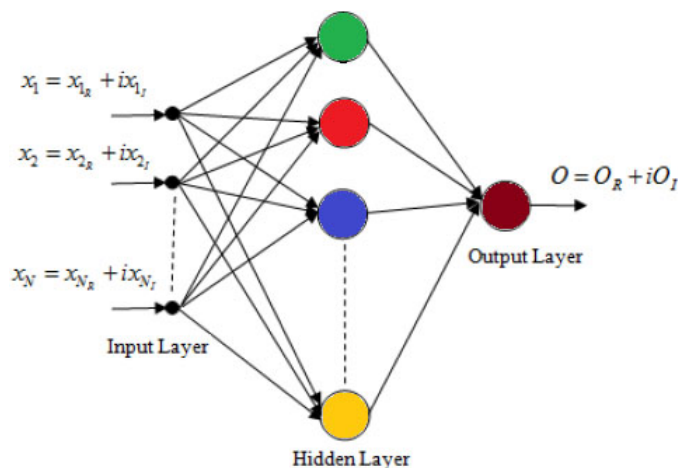


Fig. 2: The CVNN model

IV. PARTICLE SWARM OPTIMIZATION

The particle swarm optimization algorithm [32] is based on two socio-metric principles. Particles fly through the solution space and are influenced by both the best particle in the particle population and the best solution that a current particle has discovered so far. The best particle in the population is typically denoted by *global best*, while the best position that has been visited by the current particle is denoted by *local best*. The *global best* individual conceptually connects all members of the population to one another. That is, each particle is influenced by the very best performance of any member in the entire population. The *local best* individual is conceptually seen as the ability for particles to remember past personal success. The PSO makes use of a velocity vector to update the current position of each particle in the swarm. The position of each particle is updated based on the social behaviour that a population of individuals adapts to its environment by returning to promising regions that were previously discovered. The PSO is one of the methods that can be used to update the weights of a neural network during training. Weight adjustment between processing units in PSO is carried out according to the difference between the target value and the output value of the neural model. At the end of each iteration, the smallest fitness value is remembered by PSO, and the corresponding particle is retained as global best.

V. PROBLEM FORMULATION

QSAR models are in essence a mathematical function that relates features and descriptors generated from small molecule structures to some experimentally determined activity or property [15]. The structure-activity study can indicate which features of a given molecule correlate with its activity, thus making it possible to synthesize new and more potent compounds with enhanced biological activities. QSAR analysis is based on the assumption that the behaviour of compounds is correlated to

the characteristics of their structure. In general, a QSAR model is represented as follows:

$$activity = \beta_0 + \sum_{i=1}^n \beta_i x_i \quad (4)$$

where the parameters x_i are a set of measured (or computed) properties of the compounds and β_0 through β_i are the calculated coefficients of the QSAR model. Using both RVNN and CVNN for QSAR modelling where a network of nodes and connecting weights is used to represent the interaction between input and output parameters in a prediction model. The primary function of a QSAR neural network model is to assign appropriate weights to the input nodes of a network so that a weighted function of the input nodes predicts the outputs. By formulating the QSAR neural network design problem as an optimization and search problem. This objective function $Q(S)$ needs to be optimized. Where $Q(S)$ represents the quality measurement for a solution S_i given $\forall S_i Q(S_i) \geq 0$. The problem is to find the *best solution* (i.e., QSAR model) \hat{S} such that:

$$Q(\hat{S}) = \max_S Q(S) \quad (5)$$

The validation of a QSAR relationship is probably the most important step of all. The validation estimates the reliability and accuracy of predictions before the model is put into practice. Poor predictions misguide the direction of drug development and turn downstream efforts meaningless. To verify QSAR model quality in regression tasks, we employ the commonly used mean squared error (MSE) given by,

$$MSE = \frac{1}{n} \sum_{i=1}^n (y_i - \hat{y}_i)^2 \quad (6)$$

where, \hat{y}_i , values of the predicted values, and y_i , values of the actual values. However, it is necessary to get a large number of testing compounds in order to draw statistically convincing conclusion. The main steps of the QSAR optimization using PSO algorithm listed in Algorithm 1, where there are three main steps 1) Initialize a swarm of particles (position and velocities); 2) Updating velocities; 3) Updating positions.

VI. DATA DESCRIPTION AND PROCESSING

The chemical structure is susceptible of many numerical representations, commonly known as molecular descriptors. These molecular descriptors map the structure of the molecules into a set of numerical or binary values that characterize specific molecular properties which can explain an activity. The datasets used in this study are obtained from the UCI Data Repository [33]. For this study, two data sets including 74 instances of Pyrimidines with 28 features and 186 instances of Triazines with 61 features were collected. Features were arranged according to the positions of possible substitutions and contained molecular descriptors like polarity, size, flexibility, hydrogen-bond donor, hydrogenbond acceptor, π donor, π acceptor, polarizability, effect, branching and biological activity. In the first dataset the aim of a learning algorithm is to predict the inhibition of dihydrofolate reductase by pyrimidines with low probability of error. In the second dataset the aim is to predict the inhibition of dihydrofolate reductase by triazines. In order to reproduce cancer cells triazines inhibit dihydrofolate enzymes. To make data useful to the CVNNs, it

Algorithm 1: QSAR optimization using PSO algorithm

Input: Particles, a set of QSAR models
: Fitness-Fn, a function that calculate the QSAR model error function

Output: Predictive QSAR model

1. Initialize swarm $P(t)$, each position $X_i(t)$ of each particle $P_i \in P(t)$ (random)
2. Evaluate performance $F(X_i(t))$ of each particle (using current position $X_i(t)$)
3. Compare performance of each individual to its best performance
4. Compare performance of each particle to the global best

while ($error \geq \epsilon$) **do**

for ($i = 1$ to $Size(Particles)$) **do**

 Change velocity vector of each *particle*

$$v_i(t+1) = wv_i(t) + \eta_1 \cdot rand() \cdot (p_i(t) - x_i(t)) + \eta_2 \cdot rand() \cdot (p_g(t) - x_i(t))$$

 Move each *particle*

$$x_i(t+1) = x_i(t) + v_i(t)$$

return *Best QSAR*

should be transformed into the complex valued domain. In this paper, each real-valued input is encoded into a phase between 0 and π of a complex number of unity magnitude [34]. One such mapping for each element of the vector x can be done by the following transformation:

$$\text{Let } x \in [a, b] \text{ where } a, b \in \mathbb{R} \text{ then } \theta = \frac{\pi(x - a)}{b - a} \quad (7)$$

and

$$z = \cos \theta + i \sin \theta \quad (8)$$

This transformation can be regarded as a preprocessing step. The preprocessing is commonly used even in RVNNs in order to map input values into a specified range, such as Min-Max normalization.

VII. EXPERIMENTAL RESULTS

In order to evaluate the fittest neural network model, the training algorithms are conducted through several pre-experiments to determine the parameters setting per algorithm that yields the best performance with respect to all data sets. For PSO, all swarm particles start at a random position (i.e., weights). The velocity of each particle is randomized to a small value to provide initial random impetus to the swarm. The swarm size was limited to 60 particles. The most important factor is maximum velocity parameter, which affects the convergence speed of the algorithm, is set to 0.1. For the BP the learning rate is 0.1 and activation function is sigmoid. The two algorithms are runs of 500 objective function evaluations. Both of the CVNN and RVNN models consists of three-layer neural networks with one input layer, one hidden layer containing 3 processing nodes, and one output layer. The most important factor is maximum velocity parameter w which affect the convergence speed of the algorithm is set to 0.2. The

η_1 and η_2 are 2.0 and 2.0 respectively. Using 10-fold cross-validation accuracy to measure prediction quality for each of the QSAR models in the two datasets in order to establish their true learning and generalization capabilities. The results of runs on the two drug data set summarized in Table I in terms of the *mean square error*(MSE).

TABLE I: Ten fold cross validation MSE on the Pyrimidine and Triazine datasets

	PSO Training RVNN	PSO Training CVNN	BP Training RVNN
Triazines	0.016	0.015	0.04
Pyrimidines	0.0033	0.002	0.012

As shown in Fig. 3 and 5, the actual activity of Pyrimidines data set and predicted activity using both CVNN and RVNN as QSAR models and their learning rate, that CVNN is superior for a complex non-linear prediction in comparison with RVNN. Moreover, CVNN has a faster learning rate. From these results it is concluded that the neural network model that trained by PSO is superior for a complex non-linear prediction in comparison with BP which trapped in local minima. This due to the PSO training algorithm escape from local minimal, explore interesting areas of the search space in parallel and maintain multiple solutions during the search. Also, the CVNN is get better results than RVNN and its learning rate is faster than the RVNN as shown in Fig. 5.

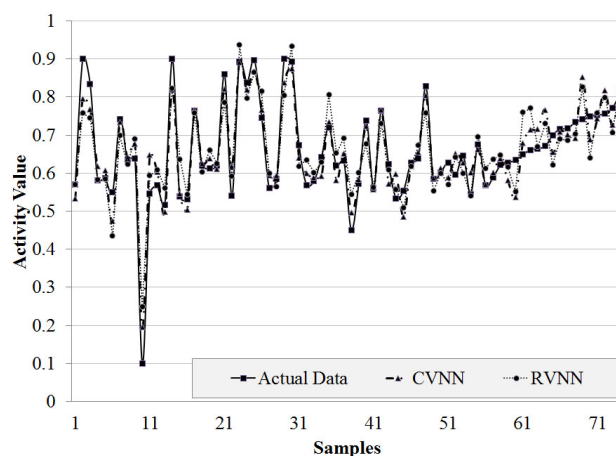


Fig. 3: The predicted activity for Pyrimidines data set using QSAR model

VIII. CONCLUSIONS

One important problem in modern drug design is to predict the activity of a compound of the drug to a binding target using its descriptors, which can be accomplished using machine learning approaches. Computationally, using efficient algorithms in the implementation, since drug datasets are noisy and has high dimensional space. This research demonstrated that complex-valued multilayer networks based on complex-valued neurons are a powerful prediction model for predicting QSAR data using PSO. The new strategy for training the CVNN based on the using of PSO as an efficient learning and optimization algorithm for real-valued regression problems. The proposed

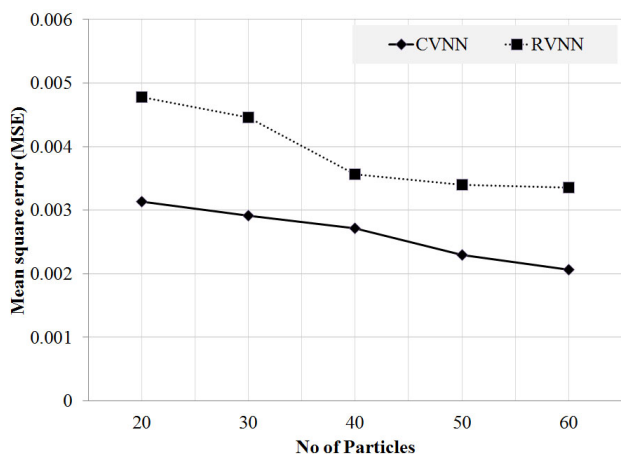


Fig. 4: The effect of particles numbers on the predicted error with QSAR model

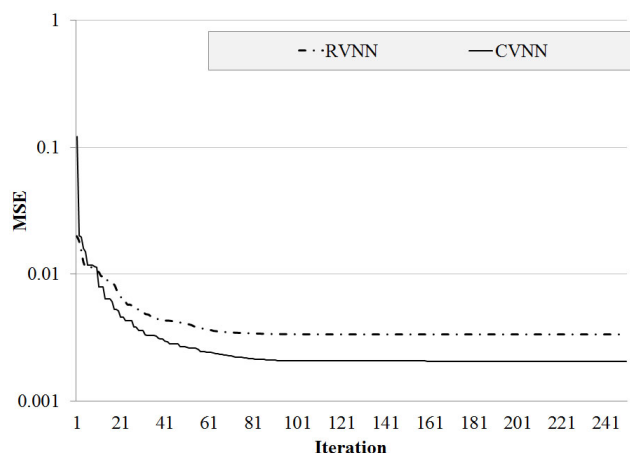


Fig. 5: The learning curves of both PSO for both CVNN and RVNN

strategy for training CVNN is implemented and tested on two drug datasets as a real world benchmark problems and compared with BP algorithm and RVNN trained by PSO. It was found that the BP has a poor performance in a comparison with CVNN and RVNN. This because the BP uses a predefined way of updating the weights and usually fall in local minima, while the weights in PSO is evolved and influenced by random factors make it escape from local minimal. Moreover, the PSO explores a larger candidate solution space than BP. The RVNN is suffer from this slow rate of learning and gave worst results than CVNN due fact that the complex-valued representation can make the network more flexible in the mapping process from input to output. This idea inspired in kernel-based learning algorithms that use non-linear mappings from input spaces to high dimensional feature spaces should be mentioned.

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Translation of the Mutation Operator from Genetic Algorithms to Evolutionary Ontologies

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Abstract—Recently introduced, evolutionary ontologies represent a new concept as a combination of genetic algorithms and ontologies. We have defined a new framework comprising a set of parameters required for any evolutionary algorithm, i.e. ontological space, representation of individuals, the main genetic operators such as selection, crossover, and mutation. Although a secondary operator, mutation proves its importance in creating and maintaining evolutionary ontologies diversity. Therefore, in this article, we widely debate the mutation topic in evolutionary ontologies, marking its usefulness in practice by experimental results. Also we introduce a new mutation operator, called relational mutation, concerning mutation of a relationship through its inverse.

Keywords—Evolutionary ontologies; Genetic algorithms; Mutation; Ontology

I. INTRODUCTION

Evolutionary ontologies (EO) represent a new concept, introduced very recently by Matei et al. in [1]. Vast and complex information used in artificial intelligence (AI) increasingly requires the use of ontologies as a manifestation of knowledge. But static information is of no use in the areas of AI, always subjected to change. Therefore evolutionary ontologies are the next natural step that is required to be made in AI.

EO are genetic algorithms using ontologies as individual items instead of classical data structures such as strings of bits, different values (real numbers, characters, objects) or programs.

Also in EO is followed the pattern of the evolutionary process: it is selected an initial population that undergoes genetic operators such as crossover and mutation, then the offspring are subjected again to selection for resuming the algorithm until the condition of the problem to be solved is fulfilled or the maximum number of epochs is reached.

Although originated in classical genetic operators, crossover and mutation that are used in evolutionary ontologies are new operators designed to meet the needs of complex structuring of knowledge contained in ontologies.

The genetic operators of evolutionary ontologies were introduced in [1], but was only traced the outline. The involvement of mutation in evolutionary ontologies diversity requires the detailing of the operator, which is the purpose of this article.

The rest of the paper is organized as follows: section II contains references to the genetic mutation operator and to the ontologies, but we have to emphasize that the two concepts are first used together in this article. Section III introduces mutation operators for evolutionary ontologies, the absolute novelty being relational mutation. In section IV we demonstrate the utility of mutation operators in practice, on an application for automatically generated scenes. Finally, the conclusions are marked in section V.

II. RELATED WORK

Imitating the biological model, genetic mutation operator changes the offspring resulting from crossover of the chromosomes [2]. Often, genetic algorithms tend to block during searching the solution at a local optimum. The purpose of the mutation is to prevent this by not allowing the population to become too similar [3].

Mutation operator does not apply to all chromosomes, but a certain percentage of the population, called mutation rate [4]. Sahoo et al. [5] state that mutation rate generally ranges between 0.05 and 0.02, but that sometimes is calculated as $1/n$, where n is the number of chromosomal genes. Mutation rate is an important parameter of genetic algorithms, because, as Cao et al. show in [6] and Maaita et al. in [7], too many or too few mutations adversely affect convergence.

Mutation operator depends on the chromosome coding technique [8]. For the binary representation is used, in particular, *bit-flipping* mutation [9], which consists in changing the values of some genes from 0 to 1 or vice versa [10].

For the integer encoding is used the *random resetting* mutation operator, where the value of a gene is replaced by a value chosen at random from an allowed range [11] or *creep* mutation, where a gene value is increased or decreased by a small value [12].

Permutation encoding involves chromosome representation as a string of integer or real values [13]. The most suitable mutation operators for this type of encoding are *swap* mutation and *inversion* mutation [14]. Under the swap mutation, two randomly chosen genes interchange their values [15]. Regarding inversion mutation, two random points are selected within the chromosome and all the genes between the two points are reversed [15].

Uniform and non-uniform mutation are two types of mutations used for real encoding of the data. The positions of the genes uniform mutated are chosen with equal probability [16], and the values of these genes are replaced with uniform random numbers [17]. Introduced by Michalewicz and presented in [18], non-uniform mutation, is to explore solutions space uniformly at first, and finally to become local search.

If we discuss coding tree, we meet *subtree mutation*, where a randomly selected subtree is replaced by another randomly created subtree (the process is broadly described in [19]) or *point mutation*, which changes a randomly selected node with a another node [20].

In addition to ordinary mutation operators, various types of mutation have been researched, such as discrete [21], [22], model-based [23] and fuzzy [24].

Although, so far, the mutation was considered a secondary genetic operator, Lehre and Yao proved in [25] that there is a close link between mutation and selection, which influences the running time of genetic algorithms. Moreover, mutation is such an important genetic operator, that significant research has been done for implementing it in different programming languages, such as C++ [26]. Also, mutation was used with particle swarm optimization technique [27], whose role is to optimize the difficult problems of mathematics in continuous or binary space [28]. Lastly, the importance of mutation operator is emphasized in practical applications of genetic algorithms, like the ones regarding the alignment of multiple molecular sequences [29], the detecting of human faces in an image with complex background [30] and the modeling of neural image compression [31].

The multitude of information requires the use of new data structures, enabling knowledge sharing and reuse, therefore, in recent years there has been an increase in the use of ontologies [32]. An ontology can be seen as a set of related concepts [33].

Matei et al. used, for the first time, the ontologies in the context of genetic algorithms, introducing the concept of evolutionary ontologies in [1]. Our goal was to enrich the ontology knowledge already held, this being achieved by applying genetic operators on ontologies. Due to the complex nature of mutation operator, we believe that the general framework introduced in [1] is not enough, therefore we detail mutation in this article.

III. MUTATION IN EVOLUTIONARY ONTOLOGIES

In [1] was defined the operating framework of ontologies under the aegis of the evolutionary algorithms. The ontological space (onto-space) is an ontology that includes all the elements of a domain of knowledge at information level with all their characteristics and relationships. Formally, an onto-space is defined as follows:

$$OS = (C, P, I) \quad (1)$$

where C is the set of classes, P is the set of properties and I is the set of instances.

An individual of the evolutionary ontologies is a subset of the ontology, meaning

$$Ch = (SC, SP, SI) \quad (2)$$

where $SC \subset C$ is a subset of classes in OS, $SP \subset P$ is a subset of properties in OS and $SI \subset I$ is a subset of instances in OS.

A population consists of (μ) such individuals, but not necessarily cover the whole onto-space, therefore

$$\bigcup_{i=1}^{\mu} Ch_i \subset OS \quad (3)$$

Due to the complex nature of individuals, is required the instantiation mutation, resulting three types of mutation operators, namely class mutation, instance mutation and property mutation.

A. Class mutation

The set of classes of onto-space, C , can be seen as a reunion of all ontology classes

$$C = \bigcup_{i=1}^{n_c} C_i \quad (4)$$

where n_c is the number of classes in the ontology.

In an ontology, therefore also in onto-space, classes are organized hierarchically [34], a class whether or not containing one or more subclasses.

Any class or subclass C_i may have one or more subclasses, which in turn can have subclasses, going as deep as much require the domain of knowledge that is represented.

$$C_i = \bigcup_{j=1}^{n_{c_i}} C_{i,j} \quad (5)$$

where n_{c_i} is the number of subclasses of class C_i .

Any class or subclass C_i can be regarded as a set of instances, composed by its own instances and all instances of all its subclasses.

$$C_i = \bigcup_{k_1=1}^{n_{inst_{c_i}}} I_{i,k_1} \bigcup_{k_2=1}^{n_{c_i}} \bigcup_{k_3=1}^{n_{inst_{k_2}}} I_{i,k_2.k_3} \quad (6)$$

where $n_{inst_{c_i}}$ is the number of the own instances of class C_i .

Mutation of a class C_i takes place by replacing all its own instances with the instances of a randomly chosen subclass of class C_i . In other words, the instances $\bigcup_{k_1=1}^{n_{inst_{c_i}}} I_{i,k_1}$ will be replaced by the instances $\bigcup_{k_3=1}^{n_{inst_{q_1}}} I_{i,q_1.k_3}$, where q_1 is a random number with values in the range $[1, n_{c_i}]$ representing the serial number of the selected subclass.

B. Instance mutation

The set of instances of onto-space, I , is the reunion of all ontology instances

$$I = \bigcup_{i=1}^{n_{inst}} I_i \quad (7)$$

where n_{inst} is the number of instances in the ontology.

The instances are grouped together in classes, as can be seen in (6). We consider an instance I_{i,q_1} , where $q_1 \in$

$[1, n_{inst_{c_i}}]$, of a class C_i . Mutation of the instance $I_{i.q_1}$ means replacing it with another instance of class C_i , namely $I_{i.q_2}$, where $q_2 \in [1, n_{inst_{c_i}}]$, $q_2 \neq q_1$ and $I_{i.q_2} \notin \bigcup_{k_2=1}^{n_{c_i}} \bigcup_{k_3=1}^{n_{inst_{k_2}}} I_{i.k_2.k_3}$.

C. Data Property Mutation

In an ontology, thus also in onto-space, are two types of properties: data properties and object properties [35]. We note with P_{d_i} a data property, $\forall i \in [1, n_{p_d}]$, where n_{p_d} is the number of data properties in OS and with P_{o_j} an object property, $\forall j \in [1, n_{p_o}]$, where n_{p_o} is the number of object properties in OS. Therefore, the set of onto-space properties is defined as follows:

$$P = \bigcup_{i=1}^{n_{p_d}} P_{d_i} \cup \bigcup_{j=1}^{n_{p_o}} P_{o_j} \quad (8)$$

The mutation may be applied differently, depending on the type of property.

A data property is characterized by a data type, that indicates the kind of value for the corresponding instance, which have such property data. Typically, the data types most commonly used for data properties are numerical types and boolean.

We consider a data property P_{d_i} , with $i \in [1, n_{p_d}]$. If I_j , with $j \in [1, n_{inst_i}]$ is one of the instances that have this data property, and the value corresponding to I_j for property P_{d_i} is val , with $val \in [val_1, val_k]$, then applying mutation on data property means changing the val value. Changing the val value depends on the data type of the property P_{d_i} , as follows:

- 1) If data property P_{d_i} has an integer data type, then there are two possibilities:

- a) the val value will be changed with val' , another value from the range $[val_1, val_k]$, ie

$$val = val', \text{ where } val' \in [val_1, val_k] \quad (9)$$

- b) the val value will be changed by adding a value (positive or negative), which is part of a symmetrical distribution to zero, ie

$$val = val + \alpha, \text{ where } \alpha > 0 \text{ or } \alpha < 0 \\ \text{such that } val + \alpha \in [val_1, val_k] \quad (10)$$

- 2) If data property P_{d_i} has a real data type, then there are two possibilities:

- a) first possibility is similar to the first possibility for integer values, hence (9) takes place also in this case
- b) the val value will be changed by adding a value (positive or negative), based on non-

uniform mutation, ie

$$val = val + \alpha \cdot |\beta - val| \cdot r \cdot \left(1 - \frac{g}{G}\right)^b$$

where

r is a uniformly distributed random number $[0, 1]$,

g is the current generation,

G is the maximum number of generations,

b is the degree of non-uniformity usually of value 5, moreover

when $\alpha > 0$, $\beta = val_k$

and when $\alpha < 0$, $\beta = val_1$ (11)

- 3) If data property P_{d_i} has a boolean data type, then mutation acts as a switch by changing the value val with its complement, ie

$$val = val', \text{ where } val' = \text{FALSE if } val = \text{TRUE or} \\ val' = \text{TRUE if } val = \text{FALSE} \quad (12)$$

D. Object Property Mutation

An object property is designed to establish relations between the classes of the onto-space and thus between their instances. Each object property has behavior of a binary relation [36], so each object property is defined on a domain and has values in a range.

We consider an object property P_{o_j} , with $j \in [1, n_{p_o}]$, which has the domain, respectively the range one class or a class reunion.

$$(P_{o_j}, \bigcup_{k_1=1}^{n_{dom_j}} C_{k_1}, \bigcup_{k_2=1}^{n_{r_j}} C_{k_2}) \quad (13)$$

where n_{dom_j} is the number of domain classes and n_{r_j} is the number of range classes for object property P_{o_j} .

According to (6), an instance of the domain is $I_{i_1.k_1}$, where $k_1 \in [1, n_{inst_{c_{i_1}}}]$ or $I_{i_1.k_2.k_3}$, where $k_2 \in [1, n_{c_{i_1}}]$ and $k_3 \in [1, n_{inst_{k_2}}]$ and an instance of the range is $I_{i_2.k_1}$, where $k_1 \in [1, n_{inst_{c_{i_2}}}]$ or $I_{i_2.k_2.k_3}$, where $k_2 \in [1, n_{c_{i_2}}]$ and $k_3 \in [1, n_{inst_{k_2}}]$.

For the sake of simplicity, we note, the P_{o_j} domain with D_j , the P_{o_j} range with R_j , an instance of the domain with I_{i_1} , an instance of the range with I_{i_2} , the results being the same regardless of chosen notations. Thus

$$I_{i_1} P_{o_j} I_{i_2} \quad (14)$$

where $I_{i_1} \in D_j$ and $I_{i_2} \in R_j$.

The inverse of the P_{o_j} , if any, is $P_{o_j}^-$ with the domain R_j and the range D_j , therefore

$$I_{i_2} P_{o_j}^- I_{i_1} \quad (15)$$

Applying mutation operator on the object property P_{o_j} means one of the following:

- 1) Replacing the object property P_{o_j} with its inverse $P_{o_j}^-$, if it exists, very suitable for spatial relations,

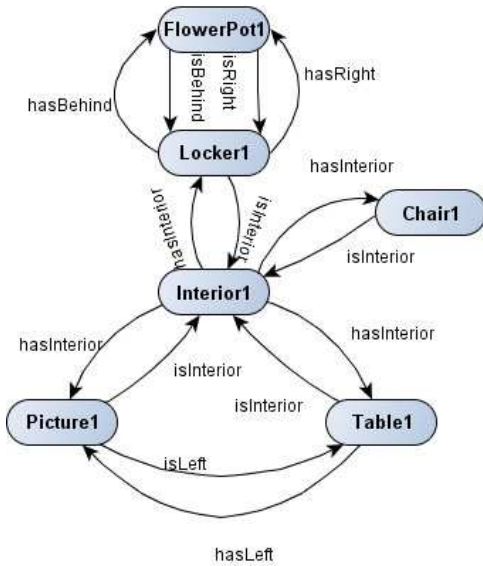


Fig. 1: Scene before relational mutation

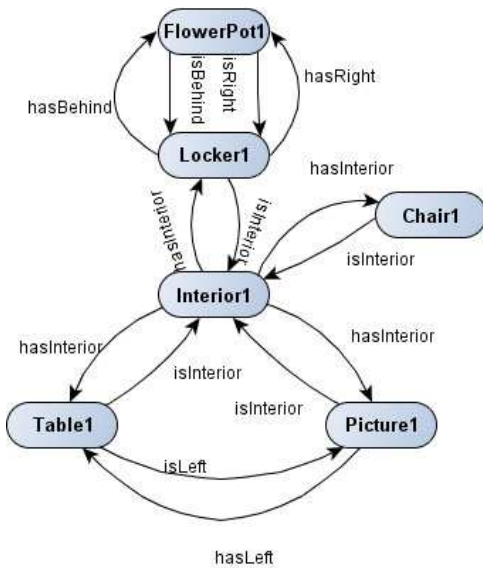


Fig. 2: Scene after relational mutation

getting a mirror effect, as you can see in Fig. 1 and Fig. 2.

We call this type of mutation, **relational** mutation, but it does not apply in all conditions. The relations (14) and (15) occur before mutation. After mutation:

$$\begin{aligned} I_{i_1} P_{o_j}^- I_{i_2} \\ I_{i_2} P_{o_j} I_{i_1} \end{aligned} \quad (16)$$

with $I_{i_1} \in R_j$ and $I_{i_2} \in D_j$. So I_{i_1} and I_{i_2} belong both to the domain and range, when applying relational mutation. For relational mutation to be

valid, must occur:

$D_j = R_j$, in which case relations P_{o_j} and $P_{o_j}^-$ swap or

$D_j \cap R_j \neq \perp$, in which case the relational mutation applies only when relations link instances of intersection. (17)

Of course that relational mutation may also occur in other particular cases, which must be checked manually to match the onto-space rules, but those shown in (17) may be automatically applied as imposed by evolutionary ontologies.

- 2) Changing the object property P_{o_j} with another object property P_o , which is not part of the onto-space, but suitable for domain knowledge, so as to have the same domain and range as the object property P_{o_j} .

$P_{o_j} = P_o$, so that $P_o \notin OS$, but

$$(P_o, \bigcup_{k_1=1}^{n_{dom_j}} C_{k_1}, \bigcup_{k_2=1}^{n_{co_j}} C_{k_2}) \quad (18)$$

If the object property P_{o_j} has inverse, we identify two situations:

- a) The new object property P_o has inverse, in which case the inverse of object property P_o will substitute the inverse of object property P_{o_j} ;
 - b) The new object property P_o has no inverse, in which case the inverse of object property P_{o_j} will be deleted from the onto-space.
- 3) Replacing one of the instances I_{i_1} or I_{i_2} with another instance I'_{i_1} , respectively I'_{i_2} , belonging to the domain, respectively to the range.

starting from $I_{i_1} P_{o_j} I_{i_2}$, is obtained by the application of mutation

$$\begin{aligned} I'_{i_1} P_{o_j} I_{i_2}, \text{ where } I'_{i_1} \in \bigcup_{k_1=1}^{n_{dom_j}} C_{k_1} \text{ or} \\ I_{i_1} P_{o_j} I'_{i_2} \text{ where } I'_{i_2} \in \bigcup_{k_2=1}^{n_{co_j}} C_{k_2} \end{aligned} \quad (19)$$

The third situation may seem similar to instance mutation. However, is another type of mutation because in the case of instance mutation, an instance I_i is replaced by another instance of the same class as I_i , while in the case of object property mutation, an instance is replaced by another instance of a class reunion, which is the domain or the range of the object property.

IV. EXPERIMENTAL RESULTS

The concept of *evolutionary ontologies* was applied on user centered automatically generated scenes application, like in all previous similar experiments, reported in [37], [38]. This way, the methodology is consistent throughout all the experiments. Cycle of the algorithm takes place in the following stages: it sets the initial population as a list of 10 randomly generated scenes. For each scene it selects a frame. We have considered

TABLE I: The number of differences between epochs without, respectively with mutation

Epoch	Without mutation			With mutation			Improvements		
	Median	Min.	Max.	Median	Min.	Max.	Median	Min.	Max.
5	11	9	13	13	10	16	18,18%	11,11%	23,08%
10	26	22	28	30	25	33	15,38%	13,64%	17,86%
15	34	30	38	39	34	45	14,71%	13,33%	18,42%
20	38	32	40	41	35	45	7,89%	9,38%	12,50%
25	36	32	38	39	35	43	8,33%	9,38%	13,16%
30	25	23	27	29	27	30	16,00%	17,39%	11,11%
35	26	24	27	30	29	31	15,38%	20,83%	14,81%

TABLE II: The number of differences between epochs when using a simple mutation, respectively applying also the inverse operator

Epoch	Without inverse			With inverse			Improvements		
	Median	Min.	Max.	Median	Min.	Max.	Median	Min.	Max.
5	13	10	16	14	10	16	7,69%	0,00%	0,00%
10	30	25	33	30	27	33	0,00%	8,00%	0,00%
15	39	34	45	40	34	45	2,56%	0,00%	0,00%
20	41	35	45	42	37	45	2,44%	5,71%	0,00%
25	39	35	43	39	36	44	0,00%	2,86%	2,33%
30	29	27	30	30	30	31	3,45%	11,11%	3,33%
35	30	29	31	31	31	32	3,33%	6,90%	3,23%

three types of frames sufficient for the power of example: land, water and interior. Every user (the algorithm is executed by 10 different users) gives each scene a grade of 1-10, according to his/her preferences. Then it selects scenes involved in the evolutionary process, by Monte Carlo technique, where the grades of users are the fitness function values. The selected scenes are subjected to crossover and mutation operators, whereupon the algorithm repeats until at least one scene is marked with 10 or the maximum threshold of 35 epochs is reached.

The first set of experiments were designed having in mind the importance of the mutation operator in the economy of evolutionary ontologies. The results are summarized in Table I. Two successive lines of the table, in column *Epoch*, denote the epochs range for which we have recorded values.

From Table I it is noticeable that the mutation improves the diversification with 7% to 24%, which is significant comparing with the complexity of the individuals (which is the number of components of a scene and the relationships between them).

A next set of experiments targeted the influence of the completely new type of mutation, namely *relational mutation*. The results are summarized in Table II. The header of Table II has the same meaning as the header of Table I.

The inversion itself brings interesting diversification, counting for up to almost 8%, which is significant for a simple operator, taking into account that it can be applied only for invertible relations. A graphical representation of the results reported in Table II is depicted in figure 3.

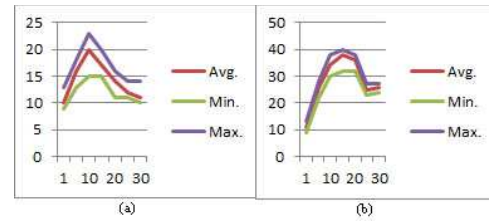


Fig. 3: The median, minimum and maximum number of differences in the two cases: (a) without inverse; (b) with inverse as mutation

V. CONCLUSION

This article puts a good mathematical foundation to the mutation operator in the context of evolutionary ontologies. Actually, we define this operator at the level of:

- classes;
- instances;
- data properties;
- object properties (class relationships).

The most complex mutation operators are at the level of object properties, where we define three specific operators:

- replacing the whole relation with its inverse when this is possible;
- replacing the property itself;
- changing relations between the instances.

The complete novelty of the article is the application of the inverse relationships as relational mutation in the context of evolutionary ontologies. This type of mutation has proved highly efficient in application for automated generation of scenes, causing the innovations in scenes due to its mirror effect.

As future work we intend to use relational mutation in an application for automatic design of industrial products, where its role is to maintain the diversity of relationships between components of products.

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Faster Scalar Multiplication Algorithm to Implement a Secured Elliptic Curve Cryptography System

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Abstract—Elliptic Curve Cryptography provides similar strength of protection comparing other public key cryptosystems but requires significantly smaller key size. This paper proposes a new faster scalar multiplication algorithm aiming at a more secured Elliptic Curve Cryptography scheme. This paper also proposes a novel Elliptic Curve Cryptography scheme where maximum length random sequence generation method is utilized as data mapping technique on elliptic curve over a finite field. The proposed scheme is tested on various bits length of prime field and key sizes. The numerical experiments demonstrate that the proposed scheme reduces the computation time compared to conventional scheme and shows very high strength against cryptanalytic attack particularly random walk attack.

Keywords—Cryptography; Elliptic curve cryptography; Scalar multiplication; Random walk; Elliptic curve discrete logarithm problem

I. INTRODUCTION

Recently, *Elliptic Curve Cryptography (ECC)* [1], [2] has gained popularity in the field of public key cryptosystem for its smaller key size, faster processing time and robust security against popular cryptanalytic attacks comparing to other *Public Key Cryptography(PKC)* systems. These features engrossed the attentions of manufacturers of small processing devices like smart cards, Raspberry computers, wireless devices, pagers, smart phones and tablets [3]. ECC is mainly used for key exchange, digital signature and authentication [4]. However, it can be applicable to any security applications where computational power and integrated circuit space is limited.

The unique idea of ECC was proposed independently by Koblitz [5] and Miller [6] in 1985. Since then on, a lot of attention has been paid to ECC, it has been studied thoroughly and still there are lots of scopes of research. Several researches are conducted to reduce the computational cost or to increase the level of security of the ECC scheme. For example, Abdalhossein Rezai [7] et al. proposed an efficient scalar multiplication algorithm for ECC using a New Signed-Digit Representation. D. Sravana Kumar [8] et al. proposed a new encryption algorithm using Elliptic Curve over finite fields. F. Amounas [9] et al. proposed an algorithm to generate a data sequence and applied it on ECC encrypted message over the finite field $GF(p)$. During this time, cryptanalysis of ECC went on with the same pace.

This paper studies the basics of ECC and some existing algorithms on it to move forward to the proposed approach. This paper proposes a faster scalar multiplication algorithm and a new scheme for ECC. In addition to these, a data mapping technique on elliptic curve over a finite field is

proposed using maximum length random sequence generation algorithm. Within our knowledge, the same approach has not yet been reported neither for ECC scheme nor for scalar multiplication algorithm. The message to *points on elliptic curve* mapping is done using random sequence generation algorithm that increase the security of the proposed scheme to higher level.

The rest of the paper is organized as follows: Section II describes the preliminary studies for proposed approach. Section III describes the proposed ECC scheme and scalar multiplication algorithm along with the necessary algorithms needed to implement the proposed ECC scheme. Section IV presents the experimental results and discussions for the proposed approach. Section V provides the conclusion and future work.

II. PRELIMINARY STUDY

In this section, preliminary theories and studies for ECC scheme are briefly described. An elliptic curve E over \mathbb{F}_p , for a prime $p > 3$ is defined with the short Weierstrass equation [10]

$$E : y^2 = x^3 + ax + b \text{ with } x, y, a, b \in \mathbb{F}_p \quad (1)$$

where a, b are integer modulo p , satisfying: $4a^3 + 27b^2 \neq 0 \pmod{p}$, and include a point \mathcal{O} called *point at infinity*. The basic condition for any cryptosystem is that the system is closed, i.e., any operation on an element of the system results in another element of the system. In order to satisfy this condition for elliptic curves, it is necessary to construct nonstandard *addition* and *multiplication* operations.

A. Geometric Rules of Addition

Let $P(x_1, y_1)$ and $Q(x_2, y_2)$ be two points on the elliptic curve E . The sum $R(x_3, y_3)$ is defined as: first draw a line through P and Q , this line intersects the *elliptic curve* at a third point. Then the reflection of this *point of intersection* about X -axis is R which is the sum of the points P and Q . The same *geometric interpretation* also applies to two points P and P , with the same X -coordinate. The points are joined by a *vertical line*, which can be viewed as also intersecting the curve at the *infinity point*. We, therefore, have $P + (-P) = \mathcal{O}$, the identity element which is the *point at infinity*.

B. Point Doubling

First, draw the *tangent line* to the *elliptic curve* at P which intersects the curve at a point. Then the reflection of this point about X -axis is R . As an example the addition of two points

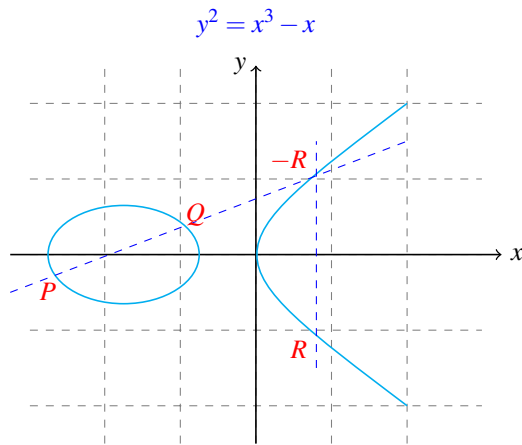


Fig. 1: Point addition on elliptic curve

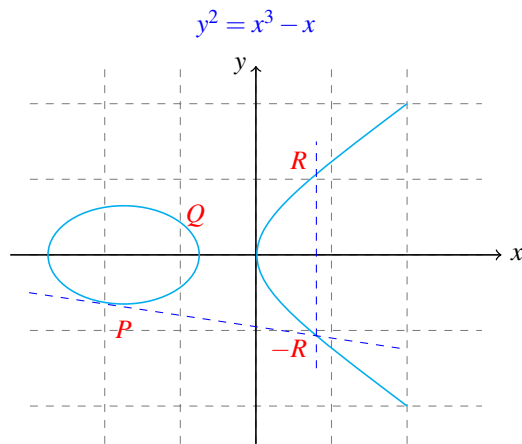


Fig. 2: Point doubling on elliptic curve

and doubling of a point are shown in Fig.1 and Fig.2 for the elliptic curve $y^2 = x^3 - x$. Point $R(x_3, y_3)$ can be derived as

$$\begin{aligned} x_3 &= \lambda^2 - x_1 - x_2 \\ y_3 &= \lambda(x_2 - x_3)y_2 \end{aligned} \quad (2)$$

where

$$\lambda = \begin{cases} \frac{y_1 - y_2}{x_1 - x_2} & \text{if } P \neq Q \\ \frac{3x_1^2 + a}{2y_1} & \text{if } P = Q \end{cases}$$

C. Conventional ECC Scheme

The conventional ECC scheme is shown in Fig. 3. Suppose Alice wants to send a message to Bob and E is an elliptic curve over \mathbb{F}_p . G is an agreed upon (and publicly known) point on the curve. Bob chooses integer b and calculates $P_b = b \times G$ and makes it public. Alice maps the plaintext m to point M on curve and secretly chooses a random integer k .

Alice encrypts M as $C_1 = k \times G$ and $C_2 = M + k \times P_b$.

Bob decrypts by calculating

$$\begin{aligned} M &= C_2 - b \times C_1 \\ &= C_2 - b \times kG \\ &= M + k \times P_b - k \times P_b \\ &= M \end{aligned}$$

D. Random Number Generation

Random number can easily be generated using Linear-Feedback Shift Registers (LFSR) [11] from maximum length polynomial. For polynomial $f(x) = x^4 + x + 1$, it has a shift register of length $m = 4$. So, it can produce a sequence of length $2^m - 1$, i.e., 15. In the numbers, sequence of bits appears random and has a very long cycle. For the given polynomial, random number sequence can be generated by calculating

$$x^i \text{ mod } f(x) \text{ for } i = 0, 1, 2, \dots, 14. \quad (3)$$

The generated random number sequence will be like Fig. 4. Stream of values produced by registers in LFSR is completely determined by its current or previous states and the Exclusive-OR operation.

III. PROPOSED APPROACH

In this section, the proposed ECC scheme, proposed scalar multiplication algorithm and the necessary algorithms to implement the ECC scheme are described.

A. Proposed ECC Scheme

Suppose, Alice and Bob want to communicate using ECC scheme. They have to agree on some issues related to elliptic curve parameters and base point. The proposed ECC scheme for covert communication is described in Algorithm 1 and shown in Fig. 5.

B. Proposed Scalar Multiplication Algorithm

The efficiency of an ECC implementation mainly depends on the way it implements the Scalar or Point Multiplication [12]. Most of the existing algorithms focus on the minimization of Hamming weight [13] of the given value by converting it to binary or sign binary numbers [14]. The proposed algorithm also works with a view to making the hamming weight minimal choosing whatever is suitable between sign binary or binary multiplication without conversion overhead. The proposed Scalar Multiplication algorithm is described in Algorithm 2.

C. Rational Point Generation

To generate rational point $P(x, y)$ from the elliptic curve equation $E : y^2 = x^3 + Ax + B$ over \mathbb{F}_p for a large prime p , it is necessary to satisfy the condition, $4A^3 + 27B^2 \neq 0 \text{ mod } p$. Any point (x_i, y_i) for $i = 0, 1, \dots, p - 1$ is a rational point on the curve if $y_i^2 = x_i^3 + Ax_i + B \text{ mod } p$ holds. The rational point generation is described in Algorithm 3.

0 0 0 1 (=01)
0 0 1 0 (=02)
0 1 0 0 (=04)
1 0 0 0 (=08)
0 0 1 1 (=03)
0 1 1 0 (=06)
1 1 0 0 (=12)
1 0 1 1 (=11)
0 1 0 1 (=05)
1 0 1 0 (=10)
0 1 1 1 (=07)
1 1 1 0 (=14)
1 1 1 1 (=15)
1 1 0 1 (=13)
1 0 0 1 (=09)

Fig. 4: Random number sequence.

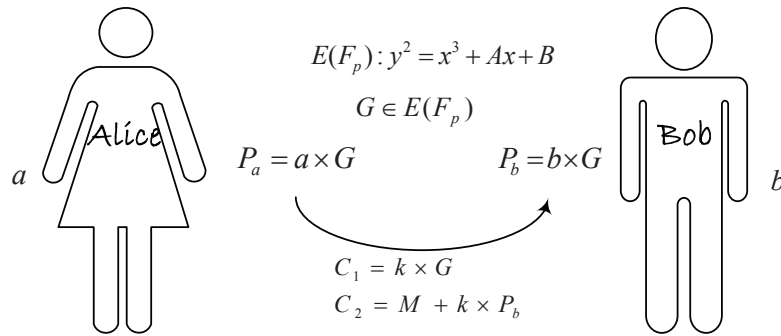


Fig. 3: Conventional ECC scheme

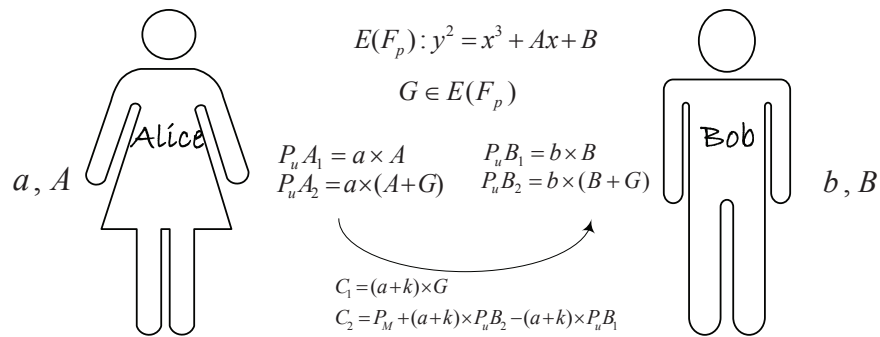


Fig. 5: Proposed ECC scheme

Algorithm 1 Proposed ECC Scheme

- 1: Both agree on curve $E : y^2 = x^3 + Ax + B$ on large prime field \mathbb{F}_p for prime p and a common point G .
- 2: Alice chooses a *random number* a and a *random point* A on the curve, keeps them as her *private key* $PrA\{a, A\}$. She calculates $PuA_1 = aA$ and $PuA_2 = a(A + G)$ and made them public.
- 3: Bob chooses *random number* b and a *random point* B on the curve, keeps them as his *private key* $PrB\{b, B\}$. He calculates $PuB_1 = bB$ and $PuB_2 = b(B + G)$ and made them public.
- 4: If Alice wants to send message M to Bob, then Alice encrypts the message in following way
 - a) maps message M to point P_M using *random sequence generation method*, $M \rightarrow P_M$.
 - b) generate a *random number* k .
 - c) Calculate $C_1 = (a + k)G$ and $C_2 = P_M + (a + k)PuB_2 - (a + k)PuB_1$.
 - d) Alice sends $\{C_1, C_2\}$ to Bob.
- 5: Bob decrypts the message in the following way
 - a) Calculate $P_M = C_2 - bC_1$.
 - b) Then message is derived, $P_M \rightarrow M$.

Algorithm 2 Scalar Multiplication, kP

```

1: procedure SCALAR MULTIPLICATION( $k, P$ )
2:    $R \leftarrow 0$ 
3:    $S \leftarrow 1$ 
4:   while  $k > 0$  do
5:      $x \leftarrow \lfloor \log_2 k \rfloor$ 
6:     if  $(k - 2^x) > (2^{x+1} - k)$  then
7:        $R \leftarrow R + (s)2^{x+1} \cdot P$ 
8:        $k \leftarrow 2^{x+1} - k$ 
9:        $s \leftarrow -s$ 
10:    else
11:       $R \leftarrow R + (s)2^x \cdot P$ 
12:       $k \leftarrow k - 2^x$ 
13:    end if
14:  end while
15:  Return  $R$ 
16: end procedure

```

D. Rational Point Addition

Algorithm provision for Rational Point Addition has already been described in section II-A. The algorithm is presented in Algorithm 4.

E. Message to Point Mapping

This section describes how a message M is mapped to point P_M . First random sequences from *maximum length polynomial*

Algorithm 3 Rational Point Generation

```

1: procedure RATIONALPOINTGENERATION( $p, A, B$ )
2:   for  $i = 0$  to  $p$  do
3:     for  $j = 0$  to  $p$  do
4:       if  $(i^3 + a \times i + b) \bmod p == j^2 \bmod p$  then
5:          $P_i.x \leftarrow i$ 
6:          $P_i.y \leftarrow j$ 
7:          $P_{i+1}.x \leftarrow i$ 
8:         if  $j \neq 0$  then
9:            $P_{i+1}.y \leftarrow p - j$ 
10:        else
11:           $P_{i+1}.y \leftarrow j$ 
12:        end if
13:      end if
14:    break
15:  end for
16: end for
17: return  $P$ 
18: end procedure

```

Algorithm 4 Rational Point Addition

```

1: procedure RATIONALPOINTADDITION ( $P(x,y), Q(x,y)$ )
2:   if  $P == 0$  then
3:      $R \leftarrow Q$ 
4:   end if
5:   if  $Q == 0$  then
6:      $R \leftarrow P$ 
7:   end if
8:   if  $P.y == -Q.y$  then
9:      $R \leftarrow 0$  ▷ when,  $P = -Q$ 
10:  end if
11:  if  $P == Q$  then ▷ Point Doubling
12:     $\lambda \leftarrow \frac{3 \times P.x^2 + a}{2 \times P.y}$ 
13:     $R.x \leftarrow \lambda^2 - 2 \times P.x$ 
14:  else ▷ Point Addition
15:     $\lambda \leftarrow \frac{P.y - Q.y}{P.x - Q.x}$ 
16:     $R.x \leftarrow \lambda^2 - (P.x + Q.x)$ 
17:  end if
18:  if  $\lambda \neq 0$  then
19:     $R.y \leftarrow -P.y + \lambda \times (P.x - R.x)$ 
20:  else
21:     $R \leftarrow 0$ 
22:  end if
23:  return  $R$ 
24: end procedure

```

are generated. For random sequence, this paper used LFSR technique on the polynomial $x^7 + x^6 + 1$ to generate random sequences. This polynomial has maximum period of 127 values ranging from 1 to 127. So it can represent 127 characters without any repetition. This paper uses only alphanumeric letters where every letter is assigned a value in the order it is generated in the sequence. For letters starts with numbers [0 – 9], small letters [a – z] and capital letters [A-Z] .

IV. EXPERIMENTAL RESULTS AND DISCUSSIONS

In this section, numerical experiment results of proposed ECC scheme and the scalar multiplication algorithm are de-

Algorithm 5 Random Sequence Generation

```

1: procedure RANDOMSEQUENCEGENERATOR( $x^7 + x^6 + 1$ )
2:    $flag \leftarrow 0x01$ 
3:    $data \leftarrow flag$ 
4:    $index \leftarrow 1$ 
5:   do
6:      $newbit \leftarrow ((data \gg 6) \wedge (data \gg 5)) \text{ AND } 1$ 
7:      $data \leftarrow ((data \ll 1) \vee newbit) \text{ AND } 0x7f$ 
8:      $R_{index} \leftarrow data$ 
9:      $index \leftarrow index + 1$ 
10:  while  $data \neq flag$ 
11:  return  $R$ 
12: end procedure

```

scribed. This paper proposes a new ECC scheme with a view to achieving robust securities. The results of the proposed approach are compared with the existing approach to find the effectiveness of the proposed approach in terms of number of operations and computational costs. Running time of the algorithm depends on the prime number and the message to be encrypted. This can also vary machine to machine and compiler to compiler. So this study implements some existing algorithms for comparisons. The approach presented in this paper is coded using C on an Intel laptop with speed of 2.13 GHz and 2GB of RAM under *Ubuntu 14.04 LTS* using *gcc - 4.9* compiler. For the operations of large bits this paper uses GMP library [15] , version-6.0.0a. The large primes are taken from *The Prime Pages library* [16] . It is tested on a message of 2.7 kb file containing only numbers and alphabets.

A. Experimental Results of Proposed ECC Scheme

Proposed ECC scheme is compared to the conventional ECC scheme on computational cost and the level of security they offer against popular cryptanalytic attacks. Conventional ECC uses only one secret number as a private key and one point on the curve as a public key. On the other hand, the proposed ECC scheme uses one number with a point on the curve as private key and two points on the curve as public key. The additional points increase the computational cost but strengthen the security higher than the conventional ECC. Table I shows the elliptic curve operations needed for conventional ECC [17] and proposed ECC Scheme. The

TABLE I: Comparison of required operations between [17] and proposed ECC

Algorithm	Operations	Addition	Subtraction	Multiplication
[17]	Key Generation	0	0	1
	Encryption	1	0	2
	Decryption	0	1	1
Proposed ECC	Key Generation	1	0	2
	Encryption	1	1	3
	Decryption	0	1	1

comparison of computational costs of Key generation, ECC encryption and ECC decryption methods of proposed scheme with the conventional ECC scheme are presented in Fig. 6, Fig. 7 and Fig. 8.

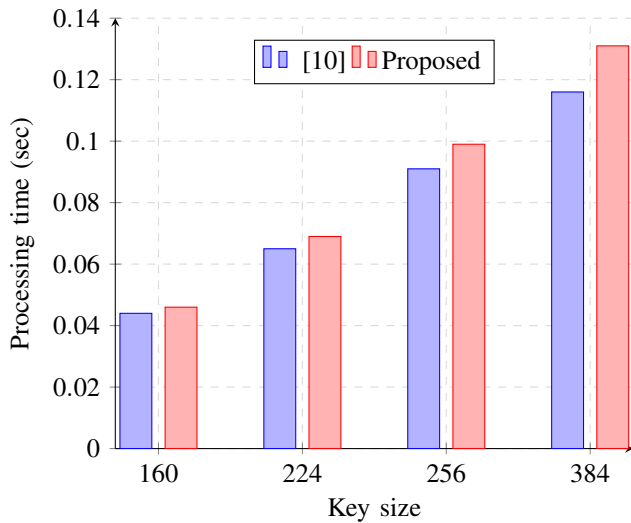


Fig. 6: Comparison of key generation cost between [10] and proposed ECC

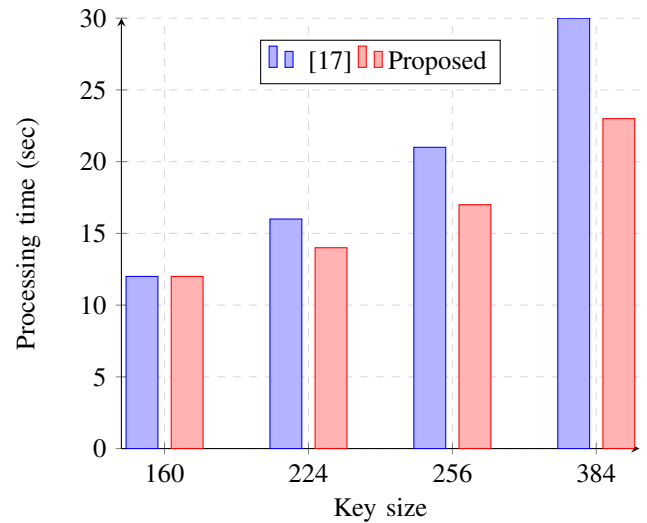


Fig. 8: Comparison of decryption cost between [17] and proposed ECC

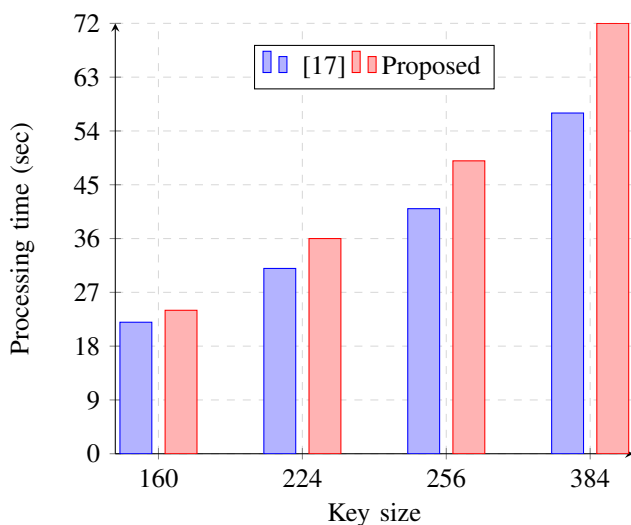


Fig. 7: Comparison of encryption cost between [17] and proposed ECC

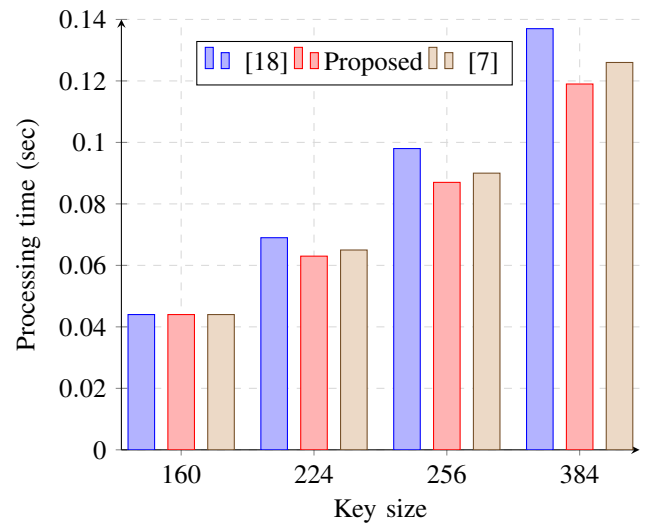


Fig. 9: Comparison of computational cost of scalar multiplication among [18], proposed ECC and [7]

B. Experimental Results of Proposed Scalar Multiplication

The efficiency of an ECC scheme depends largely on the scalar multiplication. Most of the existing algorithms have the overhead of converting the scalar number to binary or sign binary presentation to minimize the Hamming Weight. Proposed algorithm doesn't have such overhead except it needs pre-computed doublings and minimizes the Hamming Weight. The computational cost of the proposed Scalar Multiplication is shown in Fig. 9. The experimental results show that the computational cost of the proposed scalar multiplication algorithm is less than that of [18] and [7] which proves the efficiency of the proposed scalar multiplication algorithm.

C. Resistance against Attack on Proposed ECC Scheme

Proposed ECC has higher level of security than conventional ECC and experimental results for different attacks confirm it. The standard attack on ECC is *Random Walk* [19] which uses *Pollard-Rho* method for solving Elliptic Curve Discrete Logarithm Problem (ECDLP) [20]. The strategy behind the algorithm is to produce a sequence of randomly generated terms (R, a, b) where R is a point on the curve E and a, b lie in \mathbb{F}_p . As $E(\mathbb{F}_p)$ is periodic, eventually it will back again to some point. Using this technique, the secret is calculated. Pollard-Rho [21] proved that the expected running time of the method is $\sqrt{\frac{\pi \times n}{2}}$ steps, where a step here is an elliptic curve addition. As the proposed algorithm has two different secret keys the expected running time will be twice of the stated cost.

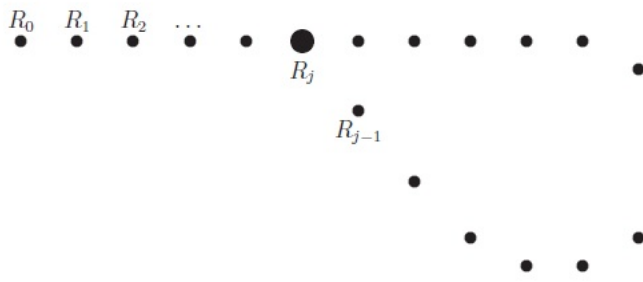


Fig. 10: Finding repetition of points using random walk technique

TABLE II: Comparison of computation cost required to break ECC

Key Size	ECC [22]		Proposed ECC	
	Cost	MIPS years	Cost	MIPS years
160	2^{80}	9.6×10^{11}	2×2^{80}	1.92×10^{12}
186	2^{93}	7.9×10^{15}	2×2^{93}	1.58×10^{16}
234	2^{117}	1.6×10^{23}	2×2^{117}	3.2×10^{23}
354	2^{177}	1.5×10^{41}	2×2^{177}	3.0×10^{41}
426	2^{213}	1.0×10^{52}	2×2^{213}	2.0×10^{52}

So the expected cost of breaking the proposed ECC scheme will roughly be $2 \times \sqrt{\frac{\pi \times n}{2}}$ steps. A MIPS (Million Instructions Per Second) year is presented as the computational power of a computer that is rated at 1 MIPS and utilized for one year. The comparison of computation cost required to break the proposed ECC with the conventional ECC [22] are presented in Table II.

V. CONCLUSION AND FUTURE WORK

This paper proposes a new ECC scheme and a scalar multiplication algorithm for it. When developing ECC scheme, this paper aims the higher level of security to be the foremost criteria to improve than the conventional ones. For scalar multiplication, reduction of computation cost is chosen to be the primary criteria and this is achieved including the advantages of both binary and signed binary presentation in the proposed algorithm. Future study aims to integrate the advantages of double scalar multiplications in the proposed ECC scheme to further minimize the computation cost. Then, the proposed scheme will be evaluated on different key sizes against different cryptanalytic attacks for further improvement of the proposed scheme.

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FPGA Prototype Implementation of Digital Hearing Aid from Software to Complete Hardware Design

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Abstract—The design and implementation of digital hearing aids requires a detailed knowledge of various digital signal processing techniques used in hearing aids like Wavelet Transforms, uniform and non-uniform Filter Banks and Fast Fourier Transform (FFT). In this paper the design and development of digital part of hearing aid is divided into three different phases. In the first phase review and Matlab simulation of various signal processing techniques used in the digital hearing aids is presented. In the second phase a software implementation was carried out and the firmware was designed for the Xilinx MicroBlaze softcore processor system. In the third phase everything was moved into hardware using VHDL hardware description language. The implementation was done on Xilinx Field Programmable Gate Array (FPGA) Development Board.

Keywords—Hearing Aid; FPGA; CODEC; MicroBlaze; Wavelets; Filter Banks; FFT

I. INTRODUCTION

It is in this context interesting to know how the human ear works because the underlying working principle of a hearing aid device is the same [3]. The human ear is a very complex organ. The basic structure is divided into three parts. Outer ear, middle ear and inner ear [7]. The outer ear receives the signal and directs it towards the middle ear. In the hearing aid this is usually done by directional microphones. The middle ear performs three basic functions. Firstly it acts as an impedance matching network, secondly it acts as an amplifier, thirdly and most importantly it splits the signal into different frequencies. A hearing aid device use some kind of Independence matching network, it also contains pre-amplifier stages and post-amplifiers, and also uses some kind of signal processing techniques, that is different frequency bands could be manipulated according to the patients need. The third part which is the inner ear works like a spectrum analyzer. It encodes the signal at different frequencies, makes different nerve cells resonate and transmit short pulses to the brain. In hearing aids after analyzing different frequencies we process them and then feed them to a digital analog converter in case of digital hearing aids. Fig. 1. and 2. shows a pictorial comparison between the human ear and a digital hearing aid. The frequency range of hearing for a normal human is 20 Hz to 20 kHz. Human hearing is most sensitive in the range of 1 kHz to 4 kHz [12]. There are two important parameters which effects hearing. One is loudness that is the intensity of sound and the other one is pitch, which is the frequency of the fundamental component in the sound. Human ears are designed to perceive sound on a logarithmic frequency and amplitude scale. The doubling of frequency will be perceived the same no matter

what the frequency will be, for instance the doubling from 200Hz to 400Hz will be perceived the same as from 2kHz to 4kHz. Same is true for amplitude, if we double the amplitude with the same ratio no matter what the amplitude is, it will be perceived as same. We normally express sound level in dB. Normal speech is around 60dB. If the ear is exposed to 85dB the ear is at risk and the sound pressure level is harmful. Above 140dB the hearing will most certainly get damaged [13].

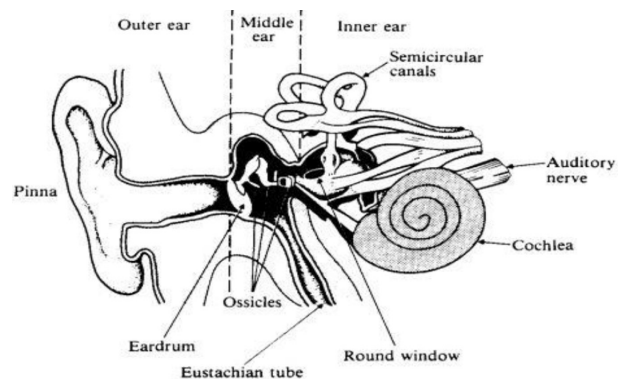


Figure 1: Structure of Human Ear

Normally the patient can suffer from three types of hearing losses [17], [18]. Patients suffering from *sensory neural loss* have some kind of damage to the auditory nerve. Some of the auditory nerves do not properly resonate with their designated frequencies and are not able to send short pulses to the brain. The other kind of loss is *conductive loss* in which the middle ear gets damaged and one does not get sound signals properly. The third kind of hearing loss is due to both conductive and sensory neural loss and is called *mixed hearing loss*. Sensory neural loss could be treated by using hearing aids, which magnifies the sound signals or cochlear implants [2], which stimulate the nerve cells directly. Nerve cells convert them to pulses which are then decoded by the brain. In case of conductive loss if the middle ear is not totally damaged hearing aid device would help by amplification of the signal in band of interest. The Hearing Aids cannot solve all hearing related issues of patients.

II. TYPES OF HEARING AIDS

We can divide hearing aids on the basis of functionality and placement [20], [22]. In *Analog Hearing Aids* every thing is composed of analog components. Analog processing is carried

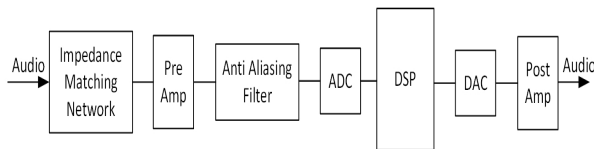


Figure 2: Block Diagram of Hearing Aid

out. The designer designs according to the patients needs adjusting the gains as needed. In *Programmable Analog Hearing Aids* the audio circuitry is built using analog circuitry but there is additional programmable control circuitry which one could use to program gain and adjust frequency settings. In *Digital Hearing Aids* almost every thing is digital from audio section to control circuitry. Very complex signal processing algorithms are there to remedy hearing losses, and also to overcome noisy environment. Humans have two ears 180 degree apart so humans can listen to sound from every direction. The sound signal could be reflected back from side walls, windows and many other objects. The human ear can detect the signal as an echo if it is received back after more than 12ms. Loose fitting of the earplug in the hearing aid could also cause feedback which is undesirable. Different echo cancellation techniques could be implemented to cancel the echo or feedback [10], [11]. Hearing aids also use compression [4], [5], [6] which basically amplify the low level sounds more than the high level sounds. If the compression is not used the hearing aid will also amplify louder sound which will be painful.

Hearing aids could also be categorized on the basis of how they are placed on the human body [21]. Following is a brief description of different hearing aids. *Completely in the Canal (CIC)*: As the name suggests they are completely in the ear canal. They are not visible from outside and are for people who don't want other people to see that they are suffering from some kind of hearing loss. They are very costly. *In the Canal (ITC)*: It is hardly visible from outside and is also a costly choice. It is good for conductive hearing losses. *Behind The Ear (BTE)*: These kinds of hearing aids rest behind the ear. They are not costly and they are visible to other people. *Body Worn Hearing Aids*: They are most simple and bigger in size. They are the least expensive. These hearing aids consists of a case, cord and an ear mold. *Bone Anchored Hearing Aid*: Another type of hearing aid is the bone anchored hearing aid. It is implanted into the skull through operation. This kind of hearing aid can treat people having conductive loss. This kind of hearing aid transmits sound directly to the hearing nerve.

III. SIGNAL PROCESSING TECHNIQUES

Consider telecommunication, avionics, medical and many others, all contain very sophisticated electronics [19]. It contains everything from software running on some embedded processor to ASIC. Power, timing and area are major concerns for many of these electronic designs. Most of the electronic devices that interact with the environment or the physical world contain some kind of signal processing technique in them. It could be an algorithm or a digital filter. Digital hearing aids do contain very sophisticated algorithms and filters [20], [23]. They may also contain different transforms for time to frequency domain conversion. In all hearing aid devices one of the basic purposes is to amplify sound in specific frequency

bands of interest. There are some well know digital techniques to do this. Most common are uniform filter banks, non uniform filter banks, wavelet transform and fast Fourier transform. In all of the techniques the frequencies are split into different bands. Some patients have hearing problems in the lower frequency bands whereas some patients have problems in high frequency bands. After splitting the sound into different frequency bands amplification can be done according to the patients needs. The next sections contains an overview of these signal processing techniques.

A. Uniform Filter Banks

A filter bank is nothing but a group of parallel low pass, band pass and high pass filters [30]. They are used commonly in audio systems among them in hearing aids. The basic idea is to give a common input and analyze the spectrum by splitting it into different frequency bands. In uniform filter banks all the filters have equal bandwidth. All filters use a common input signal sampled at the same frequency. Fig. 3. shows the basic architecture of the uniform filter bank. The input is common to all filter banks and the frequency split signals are then multiplied by the amplification factors and at the end the signals are added together. The best thing about uniform filter



Figure 3: Uniform Filter Bank

banks is that the architecture is very simple. It is also very easy to implement in hardware. Lets say that the sampling frequency is 8kHz and we have 4 filter banks each with a bandwidth of 1Khz. Fig. 4. shows the basic idea of band division. The bands in uniform filter banks are uniformly distributed therefore it does not provide sufficient resolution at low frequencies for the logarithmic behavior of hearing. Increasing the number of bands will fix the resolution problem at low frequencies but will give unnecessary high resolution at high frequencies. Another point to consider is the number of filter coefficients in each filter. If the number of coefficients is greater in number, no doubt the frequency response will be better but at the same time it will increase the latency which is the delay from input to output and the acceptable latency of human ear is 12ms. The dynamic switching power will also be worst in the case of uniform filter banks because of the number of multiplications and additions needed. There could also be leakage at the cut off frequencies of the filter due to the overlapping of bands.

B. Non-Uniform Filter Bank

The best thing about non-uniform filter banks is that they can exhibit the same behavior as the human ear does i.e. it can be designed logarithmically [31]. It is simpler than FFT or wavelets to implement in hardware. Fig. 5. shows the basic architecture of non uniform filter bank which is the same as for the uniform filter banks only the filter cutoff frequency changes. Lets consider a sampling frequency of 8

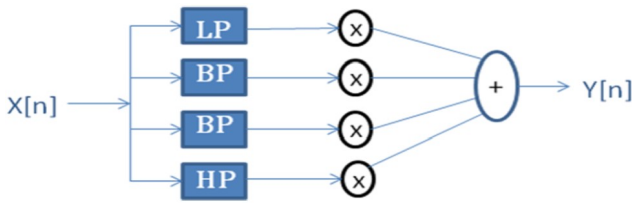


Figure 4: Filter Bank Architecture

kHz and the division of bands is non-uniform and in this case is logarithmic. The behavior is more close to the human ear as it is logarithmically divided. By using this approach better resolution could be achieved at lower frequencies. Fig. 5. shows the graphical representation of the basic idea of non-uniform filter banks. As the figure shows the resolution is better at low frequency.

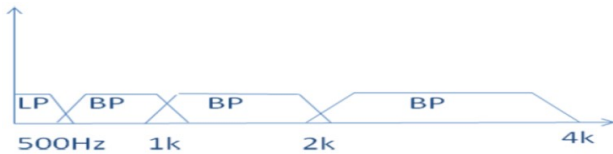


Figure 5: Non-Uniform Filter Bank

C. Wavelet Transform

Wavelet is another type of very well known transform used in hearing aids. It suits many real time applications. Wavelet works logarithmically therefore it has better frequency resolution at low frequencies [37], [38]. At the same time it has better timing resolution at higher frequencies. The architecture of the wavelets is a bit harder to implement as compared to filter banks. Fig. 6. shows the basic architecture of the wavelet transform. The signal passes through a high pass filter and a low pass filter in parallel and then gets down-sampled by a factor of two. The high pass and low pass filters split the frequency

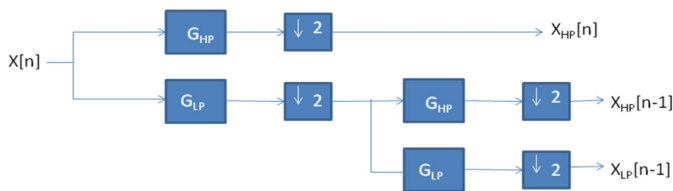


Figure 6: Wavelet Sub band Division

range into two equally sized bands. The signal from the low pass filter is again passed through a high pass filter and a low pass filter and down sampled. This is how the system gets better frequency resolution at lower frequencies. The process can go on further with more bands. For the inverse wavelet the signal is up-sampled and passes through high pass and low pass filters and the signals are then added as shown in Fig. 7.

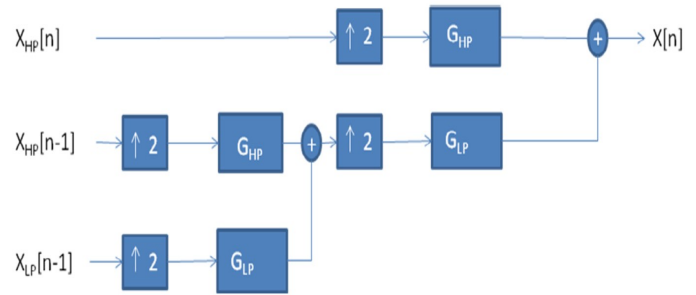


Figure 7: Wavelet Synthesis

D. Discrete Fourier Transform

The Discrete Fourier Transform (DFT) is another well known transform. It converts the time domain signal into a frequency domain signal [24], [27]. Equation (1) shows the mathematical representation of the DFT.

$$X[K] = \sum_{n=0}^{N-1} x[n]e^{-j2\pi kn/N} \quad (1)$$

The complexity of the DFT is N^2 where N is the number of samples. For example if we take a 16 point DFT then it will require 256 complex multiplications and 256 complex additions which is really expensive in terms of computations. It would take a lot of power to compute bigger DFT's and it is not realistic to use DFT. Discrete Cosine Transform (DCT) is another choice which is basically a DFT but only contain the real part of the signal. Still DCT requires lots of computations. In terms of hardware DFT and DCT are both very expensive.

E. Fast Fourier Transform

There is a highly efficient way of calculating the DFT, the well known Fast Fourier Transform (FFT) [28], [29]. It reduces the complexity of the calculations to $N * \log_2 N$. To compute a 16 point FFT requires only 64 additions and multiplications which is much less compared to computing DFT. Increasing the number of FFT points will give even better calculation gain. The signal on which the FFT is to be applied should have $N = 2^n$ samples. FFT is really efficient in terms of less hardware i.e. it will reduce both power and area. In hearing aids the basic idea is to transform the time domain signal into the frequency domain, then amplify the frequency bands of interest and then transform the signal back to the time domain. This is exactly what a healthy cochlea [25], [26] does, a biological Fourier transform which separate sound into high and low frequency bands. To take care of some of the hearing problems FFT could be used to amplify specific band of interest. As FFT is used to transform the time domain signal into the frequency domain one could use some noise reduction algorithm [39], [40] after the FFT to manipulate frequencies for better hearing in case of noisy places like markets etc.

Directionality is another issue, after the FFT some directional algorithms could be of help to improve directionality as well [15], [16]. This could be achieved in such hearing aids where dual microphones are present [8], [9]. In many cases where voice could be reflected from walls and windows and

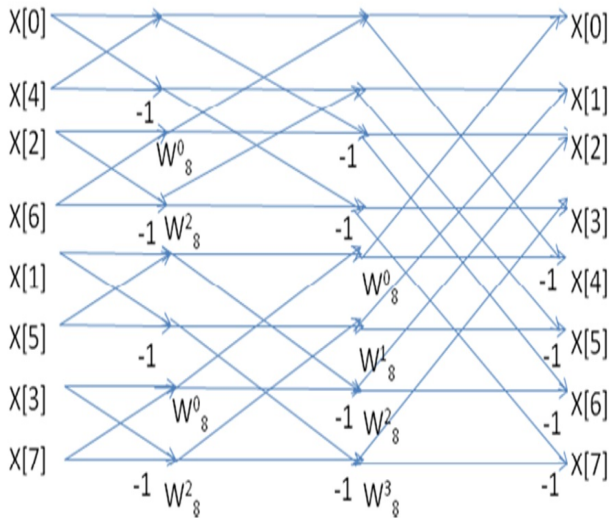


Figure 8: Decimation in Time

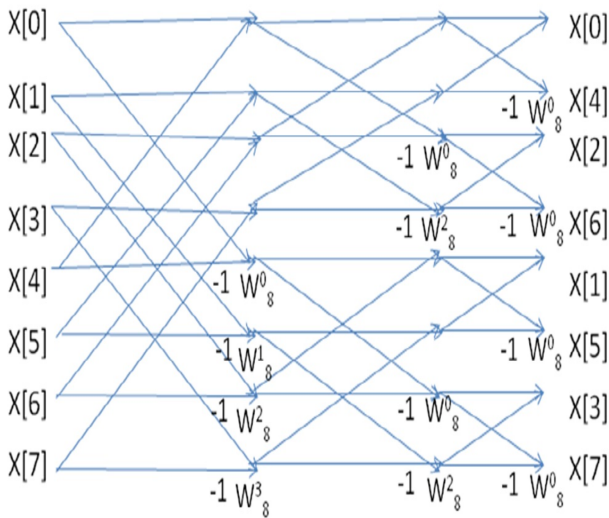


Figure 9: Decimation in Frequency

other objects may cause echo in hearing aid devices and can be noticeable, echo cancellation algorithms could be used to overcome such problems [10], [11]. The problem with FFT in hearing aids is that frequency resolution is linear so at lower frequencies the resolution will be low for a FFT with few points. It is a bit harder to implement than filter banks. FFT with non-uniform filter banks at low frequencies could give really good results. This could be achieved by diving the signal into sub bands before calculating the FFT and using more points in the lower frequency FFT which will increase the frequency resolution. Radix 2 is a very popular algorithm for computing FFT. Fig. 8. shows the FFT Radix 2 decimation in time algorithm while Fig. 9. shows FFT radix 2 decimation in frequency for N=8. An important thing to note here is the order of the input and the output samples. Decimation in time takes bit reverse inputs and generate natural order outputs whereas decimation in frequency take natural order inputs and generates an output which is in bit reversed order.

After reordering analysis is performed on the resultant outputs. The FFT algorithm in case of decimation in time consists of three basic steps:

- Invert the bit order of the index which separate even and odd indexes as shown in table I.

Table I: Bit Order DFT vs FFT

Sample #	Natural Order	Reversed	Sample #
0	000	000	0
1	001	100	4
2	010	010	2
3	011	110	6
4	100	001	1
5	101	101	5
6	110	011	3
7	111	111	7

- Calculate the twiddling factor using Equation (2) below:

$$W = \sin \theta + \cos \theta \quad (2)$$

$$\theta = -2 * \pi * (\#ofGroups) * (\#ofButterflies)/N$$

- Apply a butterfly operation which will finally represent the frequencies on specific indexes as shown in Fig. 10. The butterfly operation can be mathematically

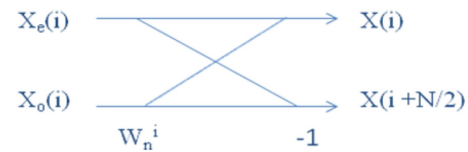


Figure 10: Butterfly Operation

represented in the form of Equations (3) and (4) below:

$$X(i) = X_e(i) + W_i^N X_o(i) \quad (3)$$

$$X(i + N/2) = X_e(i) - W_i^N X_o(i) \quad (4)$$

Real world signals are continuous in time. The problem with FFT is that it works on a finite length of the signal. Latency is an important issue in real time systems. In case of audio if the latency is more than 12ms then its not recognizable by human ears. In such cases its important to take latency into account before designing such systems. If large point FFT needs to be computed then time to gather samples and processing time should be less than 12ms. Another problem is that we have limited amount of memory and processing power. The output signal is not smooth if the sections are taken one by one. In order to make it smooth there are two methods which can be used to handle these signals. One is overlap and add and the other one is called the overlap and save method.

1) *Overlap and Add Method:* The basic algorithm of overlap and add consists of following four steps:

- Break down the signal of infinite length into finite blocks of length L.
- Pad zeros to these non overlapping blocks which makes the size of the new blocks L+M where L+M must be 2^n .

- Take FFT of the block including padded zeros.
- The signal can be resynthesized by IFFT and overlapping and adding the result, as shown in Fig. 11.

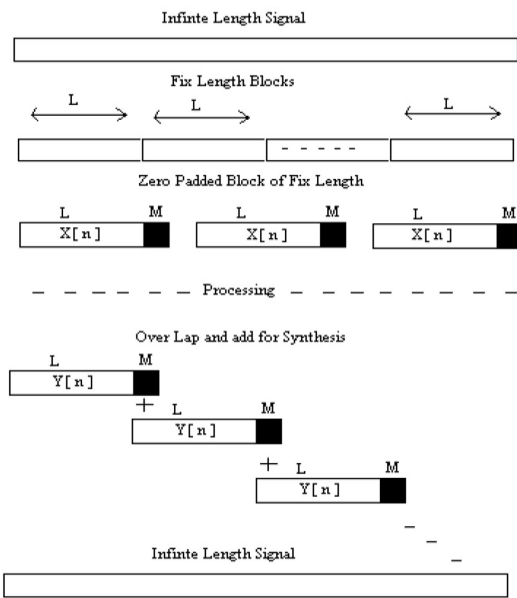


Figure 11: Overlap and Add Method

2) *Overlap and Save Method*: The basic algorithm of overlap and save consists of following five steps:

- Break down the signal of infinite length into finite blocks of length L .
- Place M zero samples before the first L input samples.
- In the following calculations take the last M samples from the proceeding block and place them before the L new samples.
- Do FFT analysis on $L+M$ sample.
- The signal can be re-synthesized by IFFT and then discarding the first M values from each group of processed samples, as shown in Fig. 12.

IV. MATLAB SIMULATIONS

Basic simulations were carried out to make sure that the FFT and the IFFT results match the original signal. The IFFT signal was compared to an IDCT output as well. The IFFT was similar to the IDCT output. As the complexity of implementation of the DCT is many folds, it was decided to implement FFT as time to frequency domain transformation. Later the bands of interest could be processed according to user needs. Fig. 13. shows simulation results of the comparison between DCT and FFT. One can clearly see that the reconstructed signal is equal to the original signal.

In order to manipulate the signal in the frequency domain it is necessary to be able to analyze the frequency spectrum. Different simulations were carried out in order to understand the frequency spectrum. Fig. 14. shows the spectrum of the simulation where the sum of sin waves at 500Hz, 1200 Hz and

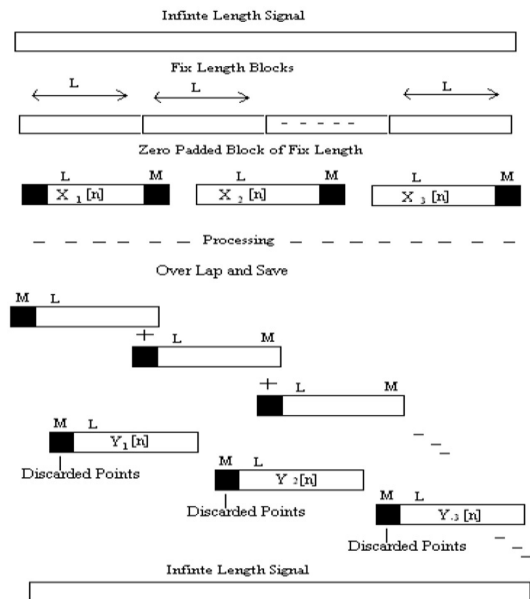


Figure 12: Overlap and Save Method

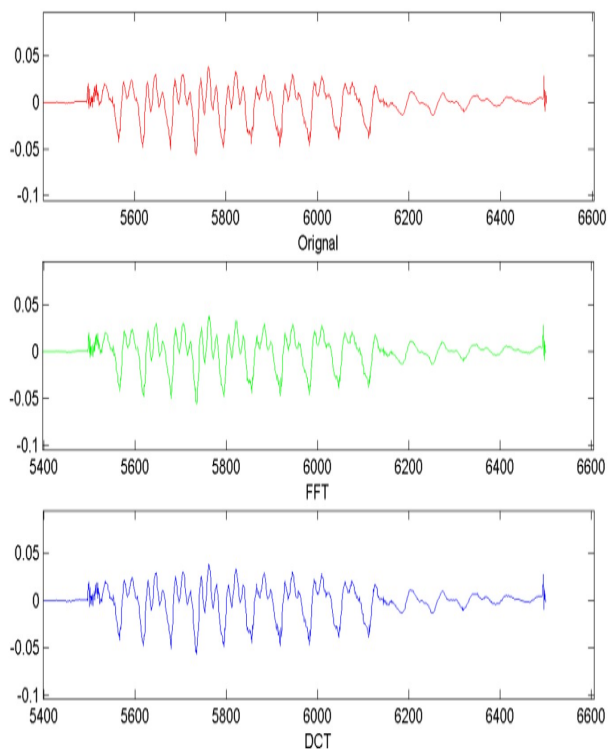


Figure 13: DCT vs FFT vs Original Signal

2900 Hz were summed and input to the FFT where the number of samples were 8192. The signal is sampled at 8.192Khz. As its clear from the result that the resolution is 1Hz and one can see the signal in the spectrum at exactly 500, 1200 and 2900 Hz.

Another simulation result is shown in Fig. 15. The input signal is a sine wave at 1200Hz and a 16 point FFT is carried

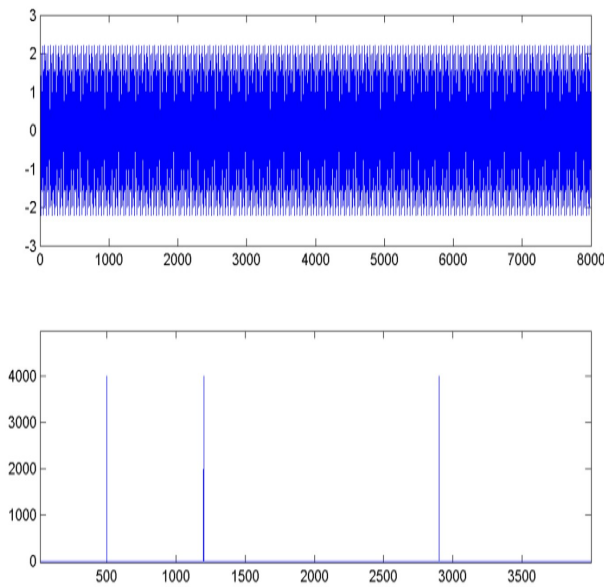


Figure 14: Spectrum of 500,1200, 2900 Hz

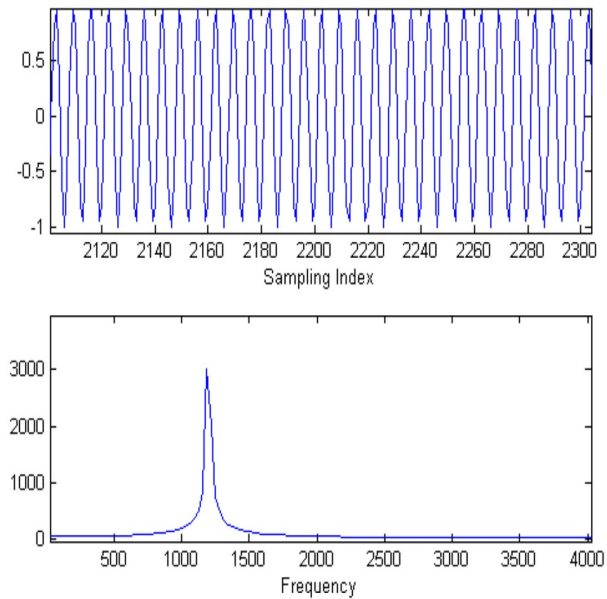


Figure 16: Spectrum of 1200 Hz with 256 points

out. The signal is sampled at 8 kHz. The resolution is 500Hz. It is obvious from the Fig. 15 that the frequency resolution is not very good. The spectrum doesn't show the signal located at 1 kHz because there is no frequency slot there. The closest frequency slots are at 1 kHz and 1.5 kHz.

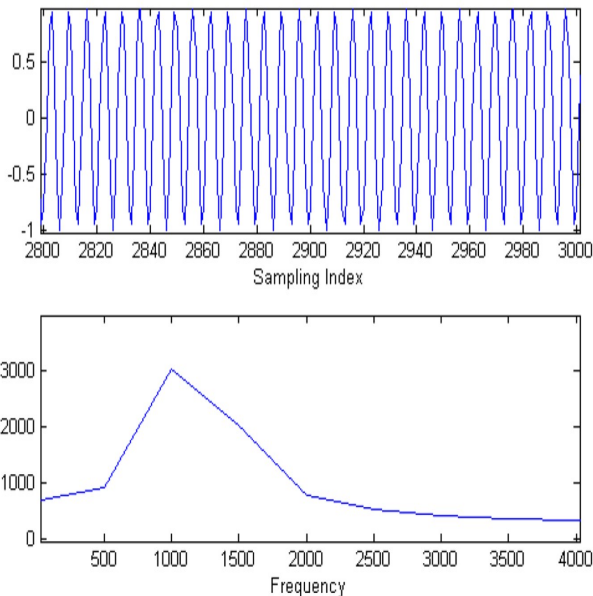


Figure 15: Spectrum of 1200 Hz with 16 points

Spectrum analysis of Fig. 16. shows that the resolution in frequency improves if we take more points in the FFT. The same signal of 1200Hz sine wave was fed to the FFT at a sampling frequency of 8 kHz using a 256 point FFT. Fig. 16. clearly shows that the resolution has increased to 31.25 Hz. It is obvious that the signal is located near 1200 Hz on the frequency scale.

V. BASIC DESCRIPTION OF SYSTEM

All of the digital hearing aids must contain some common basic blocks [1], [3]. It includes microphone, impedance matching network, preamplifiers, antialiasing filters, analog to digital converter, some kind of signal processing technique using custom hardware or an embedded processor, digital to analog converter, post amplifier, speaker. Fig. 2. shows a basic diagram of a digital hearing aid. Normally directional microphones are used in hearing aids. Some kind of impedance matching network is designed to deliver proper signal level from microphone to the preamplifier stage. The preamplifier stage amplifies the input signal and feeds the signal to the analog to digital converter. Every digital hearing aid also have an antialiasing filter before the ADC. The digitized signal is then processed. Modern digital hearing aids implement very complex algorithms for feedback and echo cancellation[10], [11]. Directivity is improved using different techniques[14], [15], [16]. Adaptive filtering techniques are used so that the hearing aid device should work in an intelligent manner i.e. it should be able to work in noisy places like markets and crowded places. The basic purpose of any hearing aid device is to amplify the signal in the bands of interest. All the latest digital devices come with some kind of user interface so that the user can adjust the gains according to the patient's needs, then the processing signal is fed to a digital to analog converter or pulse width modulator(PWM). PWM is the simplest type of DAC. The time domain discrete signal is fed to the PWM, the duty cycle of the output pulses change and then pass through a low pass analog filter. The simple architecture of PWM reduces the complexity of hardware. A speaker is used at the end of the device chain.

For the purpose of implementation the Digilent XUP II Virtex-II Pro system [32] is used. The FPGA on the board is a Xilinx XC2VP30 [33] with 30,816 logic cells. It contains 136 18bit multipliers and 2448 kb of block RAM. The board contains an audio codec the LM4550 chip from National

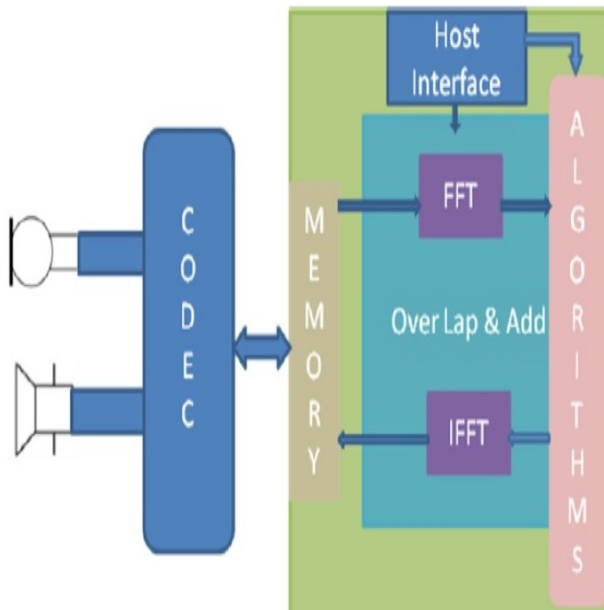


Figure 17: System Level Diagram of Hearing Aid

Semiconductor [34], which is used as an ADC and DAC. The board contains an amplifier at the output stage as well. The board is running at 100MHz. As the FFT is easier to implement than the DCT and needs very few computations FFT became the choice of implementation. For the purpose of prototyping the signal is fed through a microphone to the input of channel one of the codec. The digitized output from the ADC of the codec is fed to memory which is basically a FIFO. Fig. 17. shows the system level view of Digital Hearing Aid implementation. After receiving samples they are divided into blocks of 16 samples. The 16 sample block is then fed to a FFT calculation block which transforms the information from time domain to frequency domain. After this the bands of interest are amplified. The gain values in this prototype were changed using the push buttons of FPGA board. Two kinds of host interfaces were also tried. First a USB phy host interface was tried. A free open core source implementation was tried but it did not match the exact electrical characteristics of the USB phy interface. Another serial UART implementation was later tried at 9600 bps. After converting the audio signal from time domain to frequency domain, different kinds of complex signal processing algorithms can be used to improve hearing. It includes echo and feedback cancellation, noise reduction, filtering, adaptive noise cancellation etc [39], [40]. The implementation of the algorithm will of course affect the implementation complexity, latency, power and area. To synthesize the signal back to the time domain an inverse fast Fourier transform (IFFT) is used. The 16 samples were fed to the DAC of the codec. The board contains an amplifier at the output stage. The output is then fed to a speaker. Fig. 17. shows the block level diagram of the system implemented.

VI. SOFTWARE IMPLEMENTATION

On the basis of the simulation results the whole system was implemented in software using the Xilinx Microblaze softcore processor system [35]. It could be of great help in some

cases to implement part of the design in software and part of it in hardware to achieve acceleration [41], [42], [43], [44], [45]. The program code was written in C programming. The implementation was further divided into two phases. In the first phase the MicroBlaze processor general purpose input output (GPIO) was used to configure the LM4550 codec [34]. Due to the sequential nature of the MicroBlaze processor, it was not possible to configure the LM4550 codec properly. In the second phase a separate wrapper for the LM4550 codec was added as a peripheral which took care of the timing issues. The Peripheral Local Bus (PLB) was used to connect the LM4550 codec wrapper to the MicroBlaze processor. A FIFO was used to synchronize the data transfer between the MicroBlaze processor and the LM4550 codec wrapper. The FFT algorithm was implemented in the MicroBlaze processor. Both blocking and non-blocking approaches were tried. Blocking means that

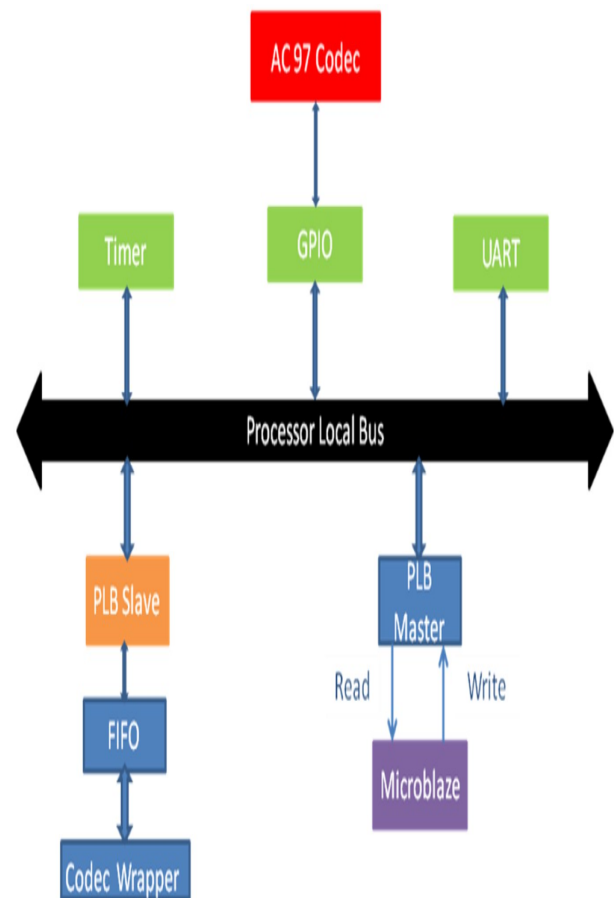


Figure 18: Block Diagram of Software Implementation

data is received through the data bus and is stored in an array before it is processed further. Non blocking means that data is received and processed continuously. The LM4550 codec wrapper as a user peripheral was not only responsible for programming of the codec but also for reading digitized inputs from the codec chip and writing output samples to the codec. Fig. 18. shows the basic architecture of the software implementation. Various problems were faced during implementation. It required lots of effort to debug various issues using the provided debugger. The slow behavior of the debugger was one

of the major reasons. The codec's serial clock was at 12.288 Mhz whereas the debugger response was very slow. It became nearly impossible to debug the serial protocol in real time, that was one of the main reasons to add the LM4550 codec wrapper as a peripheral. As printf functions takes too much memory, almost 1k, Xilinx has provided a special function to print to the terminal but it was still taking too much memory and was printing the wrong results. After some debugging it was found that if one increases the stack size it could solve the problem. The FFT algorithm was working fine, but the stack size was the problem. Another really big issue was to connect the GPIO's to the user hardware. The solution to the problem was really tedious. It took some manual changes to some of the design files including user logic in the hdl folder and the file with extension pao in the data folder. GPIOs and user peripherals are connected to the PLB, therefore if a user peripheral wants to communicate with some external hardware, in this case the LM4550 codec, then the user peripheral has to go through the PLB to communicate with the GPIO's. The flow of communication goes like this. User peripheral writes command and data to the PLB FIFO which writes to the PLB bus. The GPIO's are also connected to the PLB so data is read and written to the external LM4550 codec chip. The data read from the LM4550 codec is sent to the FFT for frequency domain conversion and after processing the bands of interest, an IFFT is carried out and written back to the codec through the PLB and GPIOs.

VII. HARDWARE IMPLEMENTATION

The main building blocks in the hardware implementation includes the Xilinx FFT core [36] and the Xilinx Dual port RAM [33]. Fig. 19. shows the block diagram of hardware implementation of Digital Hearing Aid. The architecture chosen for the Xilinx FFT core was pipelined streaming I/O 16 point FFT. Input resolution of the codec is 20 bit. The output resolution is 25 bit but the DAC only uses 20 bits. The order of the output was natural. The same core was used for both the FFT and IFFT. Latency of the system is 333usec which is far less than the limit of 10ms for hearing aids, the latency recognizable by normal human ear. There is still lots of room and time to implement other complex algorithms in hardware for instance to reduce noise, echo cancellation [39], [40], improving directionality [15] etc. Two Xilinx Dual port RAMs were used. The width of the RAMs was 20 bits and the Depth of the RAMs was 32 samples. Both RAMs were configured in Read After Write (RAW) mode. To configure the LM4550 codec chip a wrapper was used which not only configure the chip but also receives the samples from the ADC of the codec and output serial data to the DAC of the codec over a five wire serial interface. It includes serial data in, serial data out, reset for the LM4550 codec chip reset and serial clock at 12.288 MHz. The LM4550 codec chip could be programmed to sample at a rate from 4 kHz to 48 kHz in steps of 1 kHz. Different sampling rates in the range of 16kHz to 48kHz were tried during implementation. The LM4550 codec contains a sigma delta ADC whose dynamic range is 90dB. The dynamic range of the DAC is 89dB. The LM4550 codec gain is programmable and could be programmed from 0dB to 22.5dB in 1.5dB steps. The board is running at 100MHz. The LM4550 codec chip provides a 5 line serial programmable interface. At the very beginning reset is applied to the LM4550

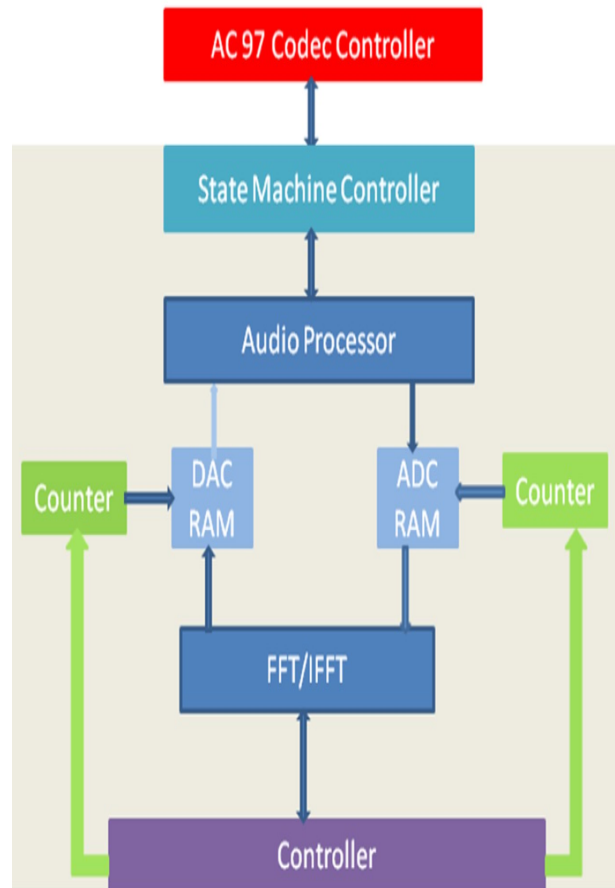


Figure 19: Block Diagram of Hardware Implementation

codec which resets all registers and internal circuitry to a default state. At the start of each frame a serial synchronization pulse goes from low to high which is used as a soft reset for the LM4550 codec chip. It is used to clear the value in the status register of the LM4550 codec. Each bit is received and sent on the bit clock over the serial lines of the interface. The frequency of the bit clock is 12.288MHz. Each serial programmable protocol frame consists of 13 slots. Each frame consists of 256 bits. Slot 0 consists of 16 bits while the rest of the slots consists of 20 bits each. Each bit in the slots has a specific purpose. They indicate status of the frame or data. For detailed description refer to the data sheet of LM4550 [34]. Fig. 19. shows a block level diagram of the main modules of hardware implementation of Hearing Aid.

After each hard reset and configuration of the LM4550 controller the audio processor block gathers samples from the LM4550 ADC. The samples are then fed to Xilinx Dual port RAMs. A counter was implemented which keeps track of if 16 samples have arrived. As soon as 16 samples are gathered these samples are fed to the FFT block for time domain to frequency domain conversion. After that the FFT bands of interest can be amplified. As the FFT core is running at 100 MHz and a 16 point FFT is calculated, there is plenty of time available to implement complex signal processing algorithm for better hearing. A controller was implemented which keeps track of the control signals from the core. As soon as the transform

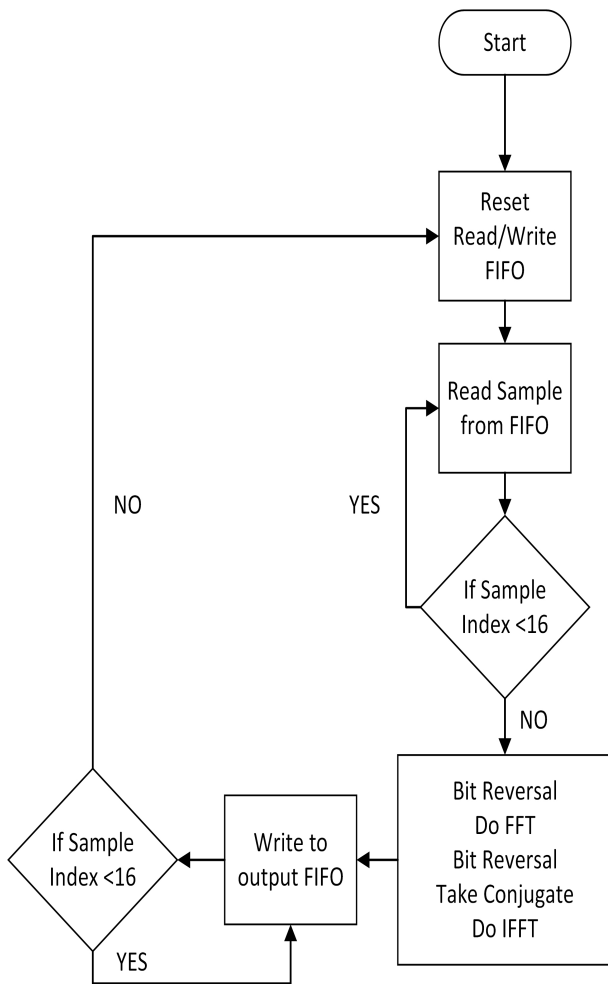


Figure 20: Flow chart of Hardware Implementation

is completed the core asserts a done signal. The same core is then used to calculate the IFFT. The core has an input signal which tells the core to calculate forward transform or reverse transform. The core needs three cycles delay before it is fed back to the FFT core to calculate the inverse transform. Three flip flops were added to take care of this issue. As soon as the inverse Fourier transform is finished the core asserts the done signal again. As the natural order for the output was selected therefore there was no need for reordering of the output samples. The samples are then fed to the RAM block. The 16 processed audio samples are then serially fed back to the LM4550 audio codec. Basic Simulation of the FFT was carried out in Modelsim. Fig. 20 shows the flow chart of hardware implementation. The results of the core were verified by comparing the results in Matlab. Debugging was carried out using both ChipScope and Modelsim. Average resource utilization of the FPGA is about 12% which includes Flip-Flops, Shift Registers, BRAMS, and Multipliers.

VIII. CONCLUSION

The main purpose of the paper was to implement a prototype for a digital hearing aid. In the first phase different signal processing techniques were analyzed which includes wavelet

transform, uniform filter banks, non-uniform filter banks, DCT and FFT. In the second stage a software implementation was carried out on Xilinx Microblaze soft core processor system. It could be of great help in some cases to implement part of the design in software and part of it in hardware to achieve acceleration. In the third stage everything was implemented in hardware. Power, area and timing are the main issues of modern electronics. These are more critical in the case of hearing aids. When considering ASIC some kind of techniques for power reduction should be used for instance gate clocking could be used. Dynamic voltage and frequency scaling could save a lot of power. Area is another important issue. Register size, multipliers, adders, resolution filter coefficient directly effect area. More hardware can also result in more power consumption. The system could be designed to run at the minimum possible frequency which will reduce dynamic switching power. Different signal processing algorithms like echo cancellation, feedback compensation, noise reduction, adaptive filtering, better directionality etc could be implemented to make hearing aid work better.

IX. ACKNOWLEDGMENTS

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Innovative Framework for e-Government adoption in Saudi Arabia: A Study from the business sector perspective

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Abstract—E-Government increases transparency and improves communication between the government and the users. Providing e-Government services to business sector is a fundamental mission of governmental agencies in Saudi Arabia. However, the adoption of e-Government systems is less than satisfactory in many countries, particularly in developing countries. This is a significant factor that can lead to e-Government failure and, therefore, to the waste of budget and effort. One pertinent, unanswered question is what are the key factors that influence the adoption and utilisation level of users from business sector. Unlike much research in the literature that has utilised common technology acceptance models and theories to analyse the adoption of e-Government, which may not be sufficient for such analysis, this study proposes a conceptual framework following a holistic approach to analyse key factors that influence the adoption and utilisation of e-Government in Saudi Arabia. The developed framework, E-Government Adoption and Utilisation Model (EGAUM), was developed based on critical evaluation of several common models and theories related to technology acceptance and use including Technology Acceptance Model (TAM) and Unified Theory of Acceptance and Use of Technology (UTAUT), in conjunction with analysis of e-Government adoption literature. The study involved 48 participating business entities from two major cities in Saudi Arabia, Riyadh and Jeddah. The descriptive and statistical analyses are presented in this paper and the results indicated that all the proposed factors have degree of influence on the adoption and utilisation level. *Perceived Benefits, Awareness, Previous Experience, and Regulations & Policies* were found to be the significant factors that are most likely to influence the adoption and usage level of users from business sector.

Keywords—E-Government; E-Services; Saudi Arabia; Technology Adoption; Influential Factors; Users' Intention; Business Sector Perspective

I. INTRODUCTION

Information and Communication Technologies (ICTs) are considered to be the backbone of many activities used nowadays. They have a considerable potential to provide solutions and to solve problems in different aspects, which then lead to enhanced quality of life. Given the fact that the accelerated development of Information Technologies (ITs) has led to a rapid increase in the number of websites and services provided by governments, nearly all governments of countries around the world have at least a web presence, or so-called e-

Government [1]. Currently, the role of ICTs is crucial in governance processes, where they can help to create a structured network for service delivery [2], effectiveness and efficiency [3], interactivity, accountability and transparency [4].

E-Government refers to the utilisation of various Information and Communication Technologies (ICTs) for facilitating communication between the government and the stakeholders; citizens, businesses and governmental agencies, providing effective, efficient and integrated e-Services that enhance the interaction between the government and the stakeholders through multiple and flexible channels that lead to an increased engagement. E-Government initiatives are still in the early stages in most developing countries, and face many issues related to adoption, implementation and utilisation. Adoption and utilisation level is fundamental in terms of measuring the success in the implementation of e-Government systems. While governments develop e-Government systems to provide e-Services to different stakeholders, the adoption and usage level is still low especially in developing countries [5][6][7]. Successful implementation of ICTs in government processes and satisfactory usage level by all government stakeholders are the main goals of e-Government. Thus, analysing the significant factors that influence the adoption and utilisation of e-Government becoming a necessity. This study focused on exploring determinants and factors that would contribute to the increase of users' adoption and use and also to the success of e-Government systems implementation.

One of the main targeted stakeholders when providing e-Government services is business sector. Business sector, or what so called private sector in Saudi Arabia, is growing considerably in the recent years. The growth rate of the private sector is the highest between the three main sectors in Saudi Arabia, namely, governmental sector, private sector and oil sector [8]. Therefore, facilitating the communication and interaction between government agencies and business sector (business firms) is very important especially in the current advanced IT era. Many online e-Services have been provided to the private sector in the recent years in Saudi Arabia and there are more e-Services under development. Besides the need to increase the adoption and utilisation level of the implemented e-Government systems, it is also crucial to understand the

factors that can influence the adoption and usage of the new e-Services. The analysis of such factors will bring e-Government service provided to business sector to a more successful level and also will draw the path for developing new e-Government services.

In order to explore and understand the determinant factors of high adoption and usage level of e-Government, we need to utilise a comprehensive framework. Thus, this paper developed a conceptual framework for this purpose, namely, E-Government Adoption and Utilisation Model (EGAUM). Early stages of the development process of this model was presented at Federated Conference on Computer Science and Information Systems, see [9]. The rest of this paper will be divided as follows: the second section highlights the essentials of the research model and the third section will present the study methodology. The fourth section will present and discuss the findings including the descriptive and statistical analysis whereas a conclusion will be presented in the fifth section.

II. THE PROPOSED RESEARCH MODEL (EGAUM)

In order to explore and analyse key factors that influence e-Government adoption and utilisation from business sector perspective, this study developed a comprehensive model we called the E-Government Adoption and Utilisation Model (EGAUM). This model provides a holistic framework to analyse key factors that have crucial influence on the adoption and diffusion of e-Government. EGAUM was developed based on critical analysis of the literature on e-Government adoption, in conjunction with insights from several models and theories that are commonly used to analyse the acceptance and usage of technologies including the Theory of Reasoned Action (TRA), the Technology Acceptance Model (TAM), the Diffusion of Innovation Theory (DOI), Perceived Characteristics Innovation (PCI) and the Unified Theory of Acceptance and Use of Technology (UTAUT) [10][11][12][13][14]. The main goal of the proposed model (EGAUM) is to determine factors that could influence the users' beliefs and intentions, as well as the behaviour that influence their adoption and usage levels. In the development of EGAUM, several significant constructs were integrated from well-studied frameworks such as Perceived Benefits, Perceived Simplicity and Social influence whereas several other constructs were introduced and developed specifically for EGAUM [9].

The EGAUM model consists of two dependent variables, namely Intention to Use E-Government (ITU) and Perceived E-Readiness of E-Government (PER) which lead to the Actual Adoption and Utilisation of e-Government. EGAUM also contains four groups of independent variables, which are Personal Factors, Motivational Factors, Technical Factors and Reliability Factors. These independent variables represent the fundamental groups of factors that have a critical influence on the adoption and usage levels of e-Government. The E-Government Adoption and Utilisation Model (EGAUM) is shown in Fig. 1. Detailed information about its developing process and its constructs is presented and published in [9].

III. THE RESEARCH METHODOLOGY

The study surveyed 53 business entities from different cities in Saudi Arabia, namely, Riyadh and Jeddah. Most of

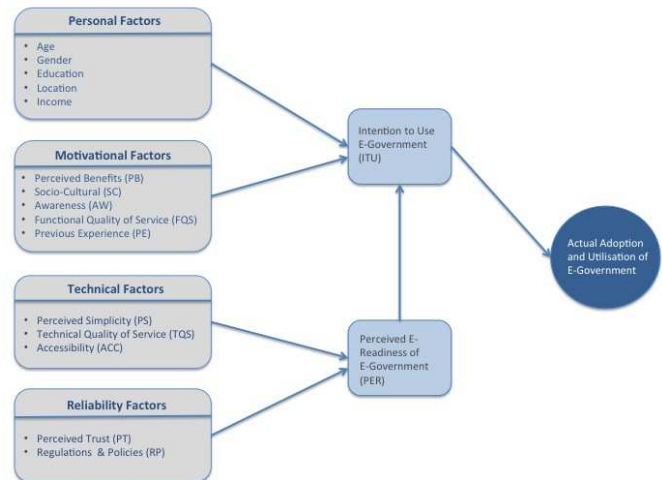


Fig. 1: E-Government Adoption and Utilisation Model (EGAUM) [9].

them were leader and large companies. Several medium and small business entities have also been involved in this study sample. The data has been collected in the period between August and October 2014. The business activities of the participating companies were different in order to provide more comprehensive results. The sample included both business owners and employees who work in business firms and deal with government agencies with regard to their companies' transactions. Incomplete questionnaires were also excluded since all the questionnaires' items need to be answered as they represent the research model constructs. Therefore, the total valid responses were from 48 business entities.

A 99-item questionnaire was distributed to the participants to respond to. The questionnaire comprised different forms of questions including 5-points Likert scale and 5-points importance degree questions. Although the questionnaire was relatively long, it collected fundamental data that led to efficient and sufficient analysis. The questionnaire was distributed in person for several reasons. For example, users from business sector are usually busy and it is likely that they would not response completely to questionnaires. Thus, meeting them in person at their appropriate time would ensure high response rate. Moreover, meeting such participants in person would help to clarify questions when they need. The data has been collected from different cities in Saudi Arabia. Although this way consumed effort, cost and time, it enabled us to obtain more comprehensive and useful data.

Since the factors in the research model (EGAUM) are abstract [9], number of items was developed for each factor to measure its influence on the adoption of e-Government. Raubenheimer stated that number of items per factor is crucial specifically for scales with one factor which requires at least four items to be identified. However, most of the scales measure more than one factor. Scales that measure more than one factor, like this research scale, can be identified with as little as two items [15]. Table 2 shows the number of items for each factor in the scale developed for this study.

In terms of validity and reliability of the study instrument, Blunch stated that a research instrument is evaluated by its reliability. The reliability of a measuring instrument means its ability to provide identical results if it is repeated under identical conditions [16]. This study used the internal consistency method to test the reliability with the use of Cronbach's Alpha, see Table 1. With regards to the instrument's validity, Bhattacharyya defined validity as "the degree to which a test measures what it intends to measure" [17]. Validity of an instrument does not mean that the instrument is either valid or not but it is a matter of degree. The greater the evidence that an instrument produces valid results the greater the likelihood that we will get information that we need [18]. Validity cannot be calculated or measured directly, it is judged by the existing evidence [19]. This study utilised face validity and content validity methods to evaluate its validity degree. Face Validity method provides useful information about the measure instrument and determines to what extent the instrument meets the intended purpose [18]. Test in which its purpose is clear, even for simple persons who have elementary knowledge, is judged to have high face validity. Where on the other hand, test in which its purpose is unclear would be judged to have low face validity [20]. Most of the items developed in this study were accurately face validated during the pilot study phase by the participants in accordance to the model factors' purposes, whereas some of them were not validated due to lack of understanding them clearly. Invalid items were either reworded if they were reported as unclear or deleted if they were reported as irrelevant. Furthermore, 5 academic members who are experts in interactive systems field have also reviewed the research instrument to have high content validity. Thus, it has been reviewed, tested and revised several times in order to have high validity.

IV. RESEARCH FINDINGS AND DISCUSSION

This section provides an overview of the demographic characteristics of the respondents including age groups, education level, income, the use of the Internet and the use of e-Government systems. This is followed by a brief descriptive analysis of each factor as well as the statistical analysis of the collected data which explains the correlation between the model's constructs that led to the hypotheses fit. For more details about the descriptive analysis see [21].

TABLE 1: Internal consistency of the study instrument [21]

Measured variable	# of items	Cronbach's α
Perceived Benefits (PB)	7 items	.825
Socio-Cultural (SC)	5 items	.686
Awareness (AW)	9 items	.822
Functional Quality of Services (FQS)	10 items	.800
Previous Experience (PE)	3 items	.601
Perceived Simplicity (PS)	6 items	.638
Technical Quality of Service (TQS)	5 items	.624
Accessibility (ACC)	4 items	.619
Perceived Trust (PT)	9 items	.792
Regulations and Policies (RP)	4 items	.899
Intention to use e-Government (ITU)	2 items	.659
Perceived e-Readiness of e-Government (PER)	2 items	.667

TABLE 2: Demographic data [21]

Variables	%
Participants gender	
Male	89.6
Female	10.4
Participants age in years	
18 - 30	20.8
31 - 45	64.6
46 - 60	12.5
Over 60	2.1
Participants education level	
Secondary school or less	29.2
Diploma	20.8
Bachelor degree	41.7
Master	8.3
Proficiency of using computer	
Average	8.3
Good	41.7
Excellent	50
Internet usage rate	
Several days a week	6.3
Several days a month	2.1
Everyday	91.6
Participants relationship to the business	
Owner	27.1
Representative	18.8
Employee	54.2
Business age in years	
0 - 5	16.7
6 - 10	16.7
11 - 20	10.4
Over 20	56.3
Number of employees in business	
0 - 10	18.8
11 - 50	8.3
51 - 250	4.2
More than 250	68.8
Annual net profit in SAR (1 GBP \approx 5.7 SAR)	
0 - 60000	10.4
61000 - 12000	4.2
121000 - 180000	2.1
181000 - 240000	4.2
More than 240000	35.4
Unknown	43.8
Business field	
Constructing and building	14.6
Restaurants	2.1
Food supply and grocery	4.2
Cars trade (sale and lease)	12.5
Communication	6.3
Health and medical supply	14.6
Other	45.8

A. Respondents' demographic data

This section presents the demographic data obtained from the respondents (see Table 2). As per the questionnaire results, the average age group was ranging between 31 and 45 with males accounting for 89.5% of the participants and 10.4% were female. It is clear that the percentage of male participants was more than female participants. These percentages were expected for several reasons. Most of the employees who work in business sector (private sector) and use e-Government services in their jobs' activities are male employees. This is mainly because of the jobs that involve dealing with government agencies to perform governmental transactions were almost exclusive to men before implementing e-Government in Saudi Arabia, therefore, they dominate these kinds of jobs due to their experience of dealing with the government services. Furthermore, it is difficult to collect data in person from female in Saudi Arabia, either business owners or employees, due to religious and cultural reasons.

TABLE 3: The most significant factors.

Hypothesised relationship	Direction of relationship	Hypothesis test	Relationship strength [42]
PB → ITU	Positive	Supported	Strong ($r_s=.543^{**}$) Sig
AW → ITU	Positive	Supported	Medium ($r_s=.463^{**}$) Sig
PE → ITU	Positive	Supported	Medium ($r_s=.354^*$) Sig
RP → PER	Positive	Supported	Medium ($r_s=.402^{**}$) Sig

*, Correlation is significant at the 0.05 level. **, Correlation is significant at the 0.01 level.

B. Data analysis and hypotheses fit

In this section, the statistical analysis of the collected data will be presented alongside with brief about the descriptive analysis. The findings will be investigated in order to test the acceptance/rejection of the research hypothesis. Moreover, the correlation between the independent and dependant variables will be investigated. The results indicated that 4 factors found to be the most significant factors that are very likely to influence the adoption and utilisation of users from business sector. These factors were *Perceived Benefits*, *Awareness*, *Previous Experience* and *Regulations & Policies* (see Table 3).

Perceived Benefits (PB)

The Likert score of *PB* factor obtained from the descriptive analysis was ($M=1.37$) indicating that this factor is very likely to influence the adoption and utilisation of users from business sector. The analysis of all *PB* items revealed that such factor would increase the intention to adopt and use e-Government services provided to the business sector [21]. This result supports the hypothesised relationship between the *Perceived Benefits* factor and the *Intention to Use* e-Government systems.

H1: There would be a positive relationship between perceived benefits and intentions to use e-government services.

The results suggest that the intention to use e-Government services is very likely to increase if users from business sector perceive the e-Services to be beneficial for their businesses. This indicates that to get business owners and business representatives to adopt and use e-Government services, these services must be useful for their business. Such e-Services should be implemented efficiently and effectively in order to meet the needs of this category of stakeholders.

The relationship between the two variables, namely, *Perceived Benefits* factor and *Intention to Use* has been assessed statistically. Spearman's correlation procedure was run to assess the relationship between *PB* and *ITU* and the results showed that there is a strong positive correlation between *PB* and *ITU*, $r_s = .543$, $n = 48$, $p < .0005$, with high level of perceived benefits associated with high level of intention to use e-Government. In other words, as the users from business sector perceive the benefits that they can gain from using e-Government systems, they are very likely to adopt and use them for their businesses' transactions. Fig. 2 shows the linear trend line that represents the positive relationship between the two variables.

The findings from this factor results are in accordance with other studies' results reported in the literature. Carter and Belanger have conducted a study to understand factors that influence the adoption of e-Government. They found that users' intention to use e-Government services would increase

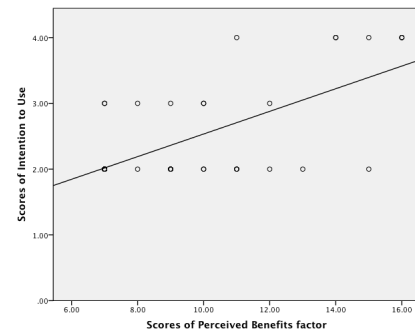


Fig. 2: The relationship between PB factor and ITU.

if they perceive the e-Services to be useful [22]. Moreover, Dimitrova and Chen conducted a study to examine non-demographic characteristics on the adoption of e-Government. One of their key findings was that the perceived usefulness would be positively related to e-Government adoption. They suggested that choosing to adopt e-Government services is rooted in the perceived benefits of using such e-Services [23]. Rogers in his Diffusion of Innovation theory also suggested that the benefits perception of using an innovation has a positive influence on the adoption decision [24].

Furthermore, AlAwadi and Morris have studied the factors that influence the adoption of e-Government services in Kuwait which is a close neighbour country to Saudi Arabia that have similar nature and culture. They found that if the users perceive e-Government services to be beneficial and useful, their intention to use such e-Service is likely to increase. They suggested that to get users to adopt and use e-Government services, these e-Services must be genuinely beneficial for them, such e-Services should be implemented effectively to satisfy their users [25]. This suggestion is emphasised when providing online governmental services to business sector where its users require high volume of transactions and services.

Businesses entities are normally required to perform many transactions that are provided by government agencies. Obtaining commercial permeations, applying for importing goods certificates and completing workers' paperwork are examples of many activities that users from business sector need to complete. An additional method of doing these tasks, the Internet, is a very useful method. Since increasing users' perception of benefits increases their intention to use e-Government services, government agencies should disseminate the benefits and advantages of using such online services and also present the potentials of implementing such e-Services. Government organisations should play a proactive role in popularizing the benefits of online government e-Services to the users from business sector. An understanding of the e-Government potentials and its benefits to the business sector will increase users intention to adopt and use it.

Socio-Cultural (SC)

The Likert score of *SC* factor obtained from its descriptive analysis indicated an impact of *SC* on the intention to adopt and use e-Government services. Although the Likert score of 2.44 that was obtained from the descriptive analysis did not represent a high influence level, it gave us an indication of a possible positive effect of this factor on users from business

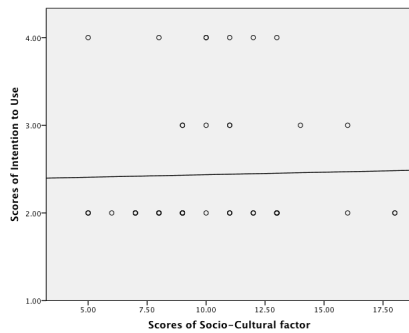


Fig. 3: The relationship between SC factor and ITU.

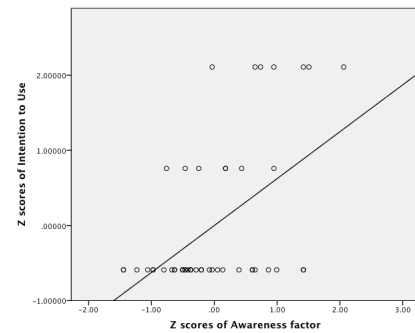


Fig. 4: The relationship between AW factor and ITU.

sector [21]. It is likely that users' intention to adopt and use e-Government would not be influenced much by the social and cultural aspects.

The relationship between *Socio-Cultural* factor and *Intention to Use* was investigated using Spearman's correlation coefficient. The results indicated that there is a positive relationship between *SC* and *ITU*, $r_s = .098$, $n = 48$, $p > .05$. Fig. 3 shows the linear trend of this relationship. Although the relationship between *SC* and *ITU* is considered to be weak, the result of the correlation coefficient shows a positive influence of the *Socio-Cultural* factor. The findings from the descriptive analysis of this factor as well as the correlation result supports the hypothesised relationship between *SC* and *ITU*.

H2: There would be a positive relationship between Socio-Cultural influence and intentions to use e-government services.

We can conclude that the social and cultural aspects do not significantly influence the adoption and utilisation of users from business sector when using e-Government services for their businesses' transactions. This is likely because they are practical users more than other users due to the busy environment that they work in. In other word, users from business sector normally need to perform a lot of governmental transactions and also need to obtain many services for their business entities regardless any other external factors such as the social life or the society culture. The influence of the *Socio-Cultural* factor on the intention to use e-Government services was not supported in several studies in the literature. For example, Alshehri et al. found that social influence do not have a significant effect on intention to use e-Government services [26]. Moreover, AlAwadi and Morris found that social influence was not significant to many participants in their study [25]. As stated earlier that users from business sector might not affected much by the social and cultural norms due to their high desire for online government services.

Awareness (AW)

Two groups of items were involved in *AW* measurement, namely, *AW* part1 and *AW* part2. The descriptive analysis revealed that it is likely that *Awareness* would positively influence the adoption and usage level of users from business sector. The Likert score of some items showed a significant influence of such factor. For example, the vast majority of the respondents agreed on that offering awareness campaigns would encourage them to attend to know more about e-Government potentials and benefits and this would positively

influence their adoption and utilisation [21].

With regard to the statistical analysis, the correlation between *AW* and *ITU* was assessed using Spearman's correlation procedure and the results revealed that there is a relatively strong relationship between them, $r_s = .46$, $n = 48$, $p < .0005$. Fig. 4 shows the positive correlation trend line between *AW* and *ITU*. We can conclude from the descriptive analysis and the correlation assessment that the *Awareness* factor positively influences the adoption and utilisation of users from business sector. The findings support the hypothesised relationship between *Awareness* and *Intention to Use*.

H3: There would be a positive relationship between awareness and intentions to use e-government services.

These findings are similar to those reported in Al-Awadi and Morris study where they found that *Awareness* plays a vital role in the use of e-Government [25]. Moreover, Alshihhi in his study also found that users' awareness about e-Government correlate positively with the willingness to use e-Government [27]. Baker and Bellordre stated that lack of awareness, that a given technology is exist or that it could of benefits, is a primary concern associated with the diffusion and use of technologies in general [28]. This is emphasised in the case of diffusion, adoption and utilisation of e-Government systems, as they are national and multi users systems.

Sufficient and efficient awareness campaigns are needed to achieve high level of adoption and usage of e-Government services. This is even more important for e-Services provided to business sector as they have large volumes of governmental transactions need to be completed. Increasing the awareness of the benefits and potentials of using e-Government services would significantly decrease the loads on the government agencies as customers from business sector would be encouraged to use online services rather than visiting agencies.

Functional Quality of Service (FQS)

The influence of *FQS* was also measured with two groups of items, namely, *FQS* part1 and *FQS* part2. The *Functional Quality of Service* refers to the external functional aspects that affect on the quality of the provided e-Service such as payment methods and the correspondence methods of the required documents. The Likert scores of the descriptive analysis of the two parts were 2.21 and 1.74 respectively. The descriptive analysis results revealed a positive impact of the high quality of the functional aspects on the users' adoption and use [21].

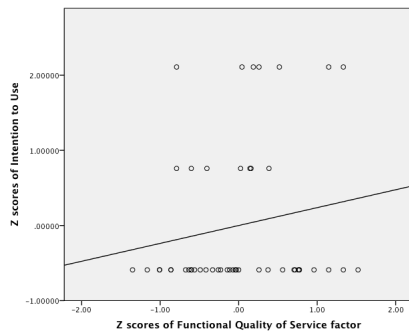


Fig. 5: The relationship between FQS factor and ITU.

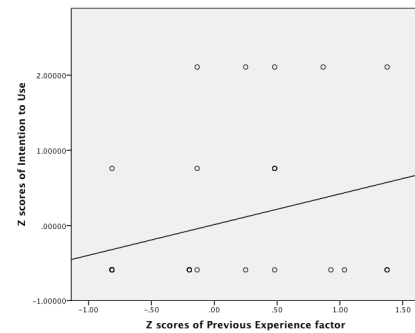


Fig. 6: The relationship between PE factor and ITU.

In terms of assessing the correlation between *Functional Quality of Service* factor and *Intention to Use*, Spearman's correlation coefficient was utilised. The result of the correlation test showed a positive relationship between *FQS* and *ITU* although the correlation was weak, $r_s = .165$, $n = 48$, $p > .05$. Fig. 5 shows the correlation trend line between both variables. The findings of both the descriptive analysis and correlation assessment can be summarised as that *FQS* positively influence the adoption and utilisation of users from business sector. This means the higher functional quality of e-Service it is, the more users from business sector will intend to adopt and use e-Government services. Thus, the hypothesised relationship between *FQS* and *ITU* was supported.

H4: There would be a positive relationship between functional quality of service and intentions to use e-government services.

E-Government should provide high quality services and also high quality delivery processes [29]. The finding of this factor is in accordance with those reported Abdelsalam et al. and Elkadi studies study. They found that service quality positively affects the use of e-Government [30][31]. Rehman et al. also found that the quality of e-Government services significantly influences the intention to perform transactions with e-Government [32]. In the current study, the influence of this factor was not found to be significant as in the discussed studies but it found a positive impact as they found. This is likely because those studies have been conducted with different e-Government stakeholders, employees in the former and citizens in the later. This means that FQS has less impact, on users from business sector, than public users or employees.

Previous Experience (PE)

PE was measured with three items with three different scales based on the question goal. All the three items were dependent on other prior items in the questionnaire and they measured the possible influence of the previous experience of using governmental online services and also the previous experience of using non-governmental online services such as online banking and online purchases. All of *PE* items were analysed separately in the descriptive analysis and the results indicated a high likelihood that the previous experience of using online services would significantly influence the intention to use e-Government services [21]. In order to statistically analyse the relationship between *PE* and *ITU*, Spearman's correlation coefficient was produced. The correlation between the *Previous Experience* factor and *Intention to Use* for the respondents who have used different kinds of e-Services, either governmental

and non-governmental, found to be positive and significant $r_s = .354$, $n = 33$, $p < .0005$. Thus, the following hypothesised relationship between *PE* and *ITU* is supported. Fig. 6 shows the relationship trend line between both variables.

H5: There would be a positive relationship between previous experience and intentions to use e-government services.

The previous experience is important in terms of using online e-Services. It can either encourage or discourage users to use such e-Services in the future. In other word, previous experience can predict and determine the users' intention to use online services. This is because different perceptions are likely built based on the previous experience. These perceptions include trust perception, ease of use perception and usefulness perception. AlAwadi and Morris found in their research that number of participant held negative view of e-Government due to negative previous experience. Those participants reported that such negative experience would deter them from using such e-Services in the future. They also found that trust in e-Commerce would lead to trust in e-Government [25]. This means that trust perception gained from previous experience of using e-Commerce would influence the trust perception when using e-Government. The importance of the previous experience was also supported in the descriptive analysis in this research.

Perceived Simplicity (PS)

PS factor was also measured with two groups of items and two different scales [21]. The relationship between *PS* and *ITU* was indirect through *Perceived E-Readiness (PER)*. Therefore, the relationship between *Perceived Simplicity (PS)* and *Perceived E-Readiness (PER)* was investigated using Spearman's correlation coefficient. There was a weak but positive correlation between the two variables, $r_s = .195$, $n = 48$, $p > .05$. Although the correlation was not strong, the result shows a positive relationship that indicates high level of perceived simplicity is associated with high level of perceived e-Readiness. Thus, the results supported the hypothesised relationship.

H6: There would be a positive relationship between perceived simplicity and perceived e-Readiness of e-Government.

Besides the results from the statistical analysis, the descriptive analysis showed similar results of the positive influence of *PS*. The results obtained from the descriptive analysis of *PS* factor can be interpreted as that the *PS* is an influential factor that positively affect the adoption and utilisation of users from business sector.

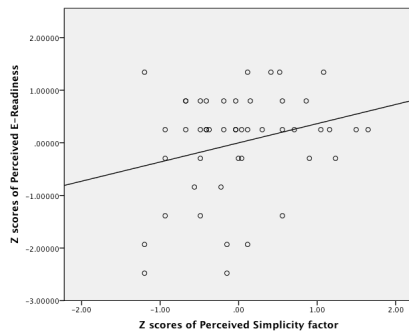


Fig. 7: The relationship between PS factor and PER.

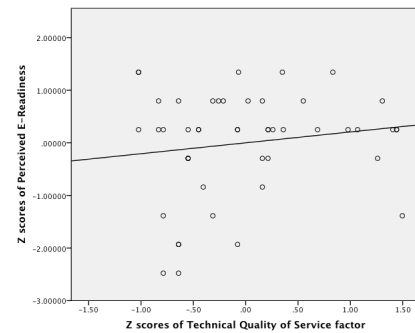


Fig. 8: The relationship between TQS factor and PER.

Although *Perceived Simplicity* factor was not found to be significant and the correlation between *PS* and *PER* was weak, a positively association was found between both of them. Fig. 7 presents the relationship trend line between *Perceived Simplicity* when using e-Government and *Perceived E-Readiness* of e-Government. Since the majority of the participants were from large business entities (around 70%) where they likely perform large amount of e-Services and e-Transactions and also most of them had a very good Internet experience (91% daily Internet users), the simplicity factor was expected not to be a significant factor due to the experience of daily usage of Internet and online services [21]. AlAwadi and Morris suggested in their research work that the greater one's use Internet experience, the easier it would be to learn and use e-Government services [25]. Moreover, Carter and Belanger found that ease of use, which is similar to simplicity in the current research, were not significant in the intention to use e-Government of users with Internet experience [22]. However, e-Government services need to be straightforward and simple to use in order to enable all potential users, either with high or low experience, to benefit from such e-Services. This simplicity is emphasised when it comes to providing e-Services to business sector where large number of e-Services and e-Transactions are involved.

Technical Quality of Service (TQS)

Two groups of items were involved in the measurement of *TQS* factor. The results from the descriptive analysis of both parts indicated that *TQS* is an influential factor. It did not reveal a significant effect but it gave an indication that there is a positive influence on the adoption and utilisation level [21].

In terms of statistical analysis, Spearman's correlation procedure was utilised to assess the relationship between *TQS* and *PER*. It was found that there is a weak relationship between the two variables, $r_s = .108$, $n = 48$, $p > .05$. However, the relationship was positive which means high level of technical quality of service positively influence the perceived e-Readiness of e-Government. Although the correlation was weak, the result supported the positive hypothesised relationship between the *Technical Quality of Service* and e-Readiness perception. The relationship trend line is shown in the scatterplot graph presented in Fig. 8 .

H7: There would be a positive relationship between technical quality of service and perceived e-Readiness of e-Government.

Technical aspects in e-Government systems are crucial in terms of judging the quality of service especially when

they are visible to the users such as interface design and website errors. Moreover, the technical features that reflect the quality of service such as the expected time to complete the e-Transactions and the last update time of the required procedures are also significant. A high quality of the technical side including the design, structure and layout of e-Government portals is needed for the successful adoption and utilisation [33]. Technical issues in e-Government systems might interrupt service transactions and processing performance which can cause severe delay in all governmental agency work and then low adoption and utilisation [25]. Several studies in the literature found that the quality of technical aspects are crucial and has significant impact on the usage level. For example, Alshehri et al. found that website quality, which is one of the technical aspects, directly impacts usage behaviour of e-Government. Their study confirmed the importance of quality government websites as one of the significant factors of e-Government adoption [34].

Nevertheless, the findings show no significant impact of the *Technical Quality of Service* on the E-Readiness perception of users from business sector. This indicates that *TQS* is not a fundamental factor that determines their adoption and utilisation but this does not diminish the positive influence of high technical quality. Although these technical and visible aspects were reported as an important in the descriptive analysis, they found to be not significant in the statistical analysis. This is possibly because the majority of the participants were employees in the participating business entities (81%) and they likely did not consider the technical aspects measured in this research as a fundamental factor that influences their adoption and utilisation of e-Government services [21]. These technical aspects might be more important for other e-Government users but not the users from business sector.

Accessibility (ACC)

The measurement of the influence of *Accessibility (ACC)* factor consisted of two groups of items with two different scales. The influence of different aspects of accessibility were measured in this study including the provision of mobile applications and the ability to access and perform e-Government transactions 24 hours/7 days [21]. Spearman's correlation procedure was utilised to assess the correlation between *ACC* and *PER*. The result showed that the hypothesised positive relationship between *Accessibility* and *Perceived E-Readiness* was not supported, $r_s = -.198$, $n = 48$, $p > .05$. The test also showed no significant relationship between them. This indicates that

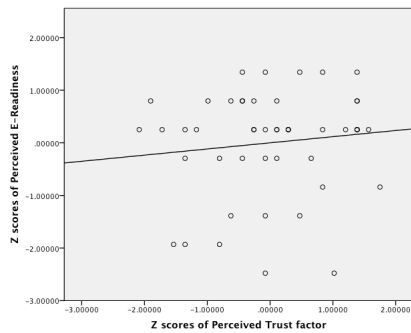


Fig. 9: The relationship between PT factor and PER.

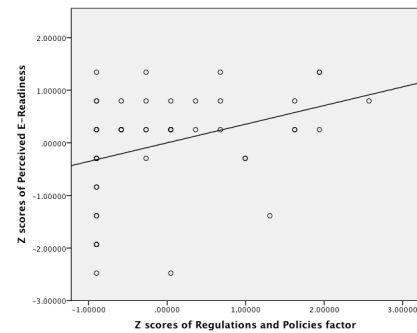


Fig. 10: The relationship between RP factor and PER.

Accessibility factor do not positively influence the e-Readiness perception of users from business sector.

H8: There would be a positive relationship between accessibility and perceived e-Readiness of e-Government.

This is likely because the majority of the participating business entities (66%) were old companies that had been established before the e-Government initiative was introduced to Saudi Arabia [21]. Such old companies might still have negative views from performing governmental transactions using the traditional ways and thus, they believed that governmental agencies would not be able to provide sophisticated features that are related to e-Government accessibility such as e-Government support offices, mobile applications and also the ability to perform e-Services 24 hours/7 days a week.

Perceived Trust (PT)

This factor was measured with nine items and the single composite score has been computed for them, $M = 2.34$ [21]. The Spearman's correlation test was performed between PT and PER to explore the possible positive relationship between them which was proposed in EGAUM. Similar to PS, TQS and ACC factors, there was a weak positive relationship between PT and PER, $r_s = .097$, $n = 48$, $p > .05$. Nevertheless, the hypothesised positive relationship was supported and its trend line can be seen in the scatterplot graph presented in Fig. 9.

H9: There would be a positive relationship between perceived trust and perceived e-Readiness of e-Government.

This insignificant impact of PT was also indicated from the descriptive analysis of this factor [21]. This weak influence is likely because the majority of the participants in this sample were employees (72%) and they possibly do not pay a lot of attention on trust issues as owners. Performing e-Government transactions and services are considered to be their job activities and trust aspects would not affect their e-Readiness perception or their intention to use.

Perceived Trust factor, which include perceived security and perceived privacy, is crucial factor when it comes to use virtual systems such as e-Government. Being able to place trust in a system is critical to consumers' successful interaction with that system [35]. Several research in the literature found that Perceived Trust is a significant factor that influence the adoption and intention to use e-Government [36][37][38][39]. However, the significance influence of PT in these studies was reported from different e-Government stakeholders. The

Perceived Trust factor might not be a significant factor for users from business sector. As stated earlier that the majority participants were employees and the trust perception is likely not influencing their adoption and utilisation. This does not mean that the data that they provide online when performing e-Services is not important to them or they are careless about it, but the influence of trust is a degree matter and its influence differs from users group to another.

Regulations and Policies (RP)

Regulations and policies factor (RP) was measured with four items and a single composite score has been produced for correlation test. The relationship between RP and PER was assessed utilising Spearman's procedure and the result showed a positive correlation between both variables. This relationship was statistically significant as the test showed, $r_s = .402$, $n = 48$, $p < .05$. Thus, the hypothesised relationship between RP and PER was supported and the relationship trend line is presented in Fig. 10.

H10: There would be a positive relationship between regulations & policies and perceived e-Readiness of e-Government.

The descriptive analysis of this factor also indicated a strong influence of RP factor [21]. This means that the implemented regulations and policies for using e-Government services is likely to have significant influence on the adoption and utilisation of e-Government and its e-Readiness perception. The findings showed how implementing visible, strict and clear regulations and policies are important to users from business sector. This indicates that such users are eager to have such legislative and control perspective when using online governmental services and this is likely because conflicts and regulatory violation would cause negative impact on their business.

Moreover, the existence of clear regulations and laws for both parties' rights (business users and government agency) and making them available and visible to the users is crucial. This importance would be more emphasised when large business entities, like the participating entities, are using such online governmental systems. E-Government environment requires a comprehensive set of regulations, terms, conditions and laws that need to be applied by the e-Services providers to allow them to leverage the full potential of automation and online services provision while ensuring users perception, privacy and security [40]. Regulations and policies related to the use of e-Government systems are needed to create a

favourable environment for promoting trusted governmental services over the Internet. Additionally, these regulations and policies are required to be updated regularly to encourage users to read and follow them.

V. THEORETICAL CONTRIBUTIONS AND PRACTICAL IMPLICATIONS

This study examined the adoption of e-Government services by potential users from business sector in Saudi Arabia. The contribution of this paper is twofold. First, guided by number of technology acceptance theories and models, the study develops an integrated theoretical model to investigate users' intention to adopt e-Government services from the business sector perspective. The new research model (EGAUM), was developed based on critical evaluation of such models and theories including TAM and UTAUT, in conjunction with analysis of technology acceptance literature [9]. It provides insight into number of crucial factors and determinants of higher adoption and use level. Second, the current study empirically validates this integrated model by analysing data that has been collected from targeted respondents and from different places. Eight of the 9 hypotheses specified in the model were supported. The results attest to the value of this research model.

The current study also informs practice. We briefly discuss the major implications derived from our research hypotheses in the following discussion. Perceived Benefits was found to be a direct influential factor on the intentions to use e-Government services. Users who have higher perceptions of e-Government benefits are more likely to adopt and use its services and transactions. Their intention to use e-Government services is very likely to increase if they perceive the e-Services to be beneficial for their businesses. Such e-Services should be implemented efficiently and effectively in order to meet the needs of this category of stakeholders. Government agencies should disseminate the benefits and advantages of using such online services and also present the potentials of implementing such e-Services. Moreover, they should play a proactive role in popularizing the benefits of online government e-Services to the users from business sector.

Furthermore, Awareness was also found to be a significant factor that is likely to increase the intention to adopt and use. If users from the business sector are informed about the existing e-Services and their potentials and benefits, they are likely to appreciate e-Government services and thus, they would be encouraged to adopt them and use them. Providing e-Services without their users' knowledge or without promoting the awareness of their benefits and potentials would limit achieving the goals of implementing e-Government. Different methods were found to be significant in terms of increasing the awareness of users from business sector. For example, Social media found to be very influential method that would increase users' willingness to use e-Government services for their business. Albayari stated that social media revolutionised the business sector in Saudi Arabia since it creates sufficient ways to communicate with customers [41]. This indicates that most of business entities have accounts in social media networks and also new businesses are expected to follow the same trend of using social media. The result of this study and the status of social media spread in business sector in Saudi

provide government with an excellent way of marketing e-Government services and increase the awareness about their benefits and potentials.

We anticipated that the previous experience of online activities would have an impact on the future adoption and use of e-Government services. The results found that the past use of e-Government services would influence users' intention to use such online services repeatedly. Moreover, the past use of non-governmental online services was also found to be an influential factor. Generally, the online activities that involve transactions and services are still new trend in Saudi Arabia. The study results alert the e-Services providers to enhance users experience. This should be considered from the first introduction of e-Services. Furthermore, improving the ICTs infrastructure would be positively reflected on the online activities.

The implemented regulations and policies also found to be a significant factor that influences users' adoption and use. The results revealed that implementing regulations, policies, conditions and terms of using e-Government systems is very important to users from business sector. They should be implemented properly by making them clear, strict, applicable to all parties (users, employees and agencies), accessible, comprehensive and visible. Formulating rules, policies, terms and conditions of using e-Government systems is not the only main thing, but also providing them in a way that encourage users to read and understand them would increase their confidence which in turn increase their adoption and use. This would also help to control such online dealings especially with business sector that contributes to the country's economy

VI. CONCLUSION

The current study attempts to understand users' intention to adopt and utilise e-Government services from the business sector perspective in Saudi Arabia. The adoption and utilisation of e-Government considered to be a challenge, and therefore, an interesting research area is introduced. We developed a research model (EGAUM) under the umbrella of different scientific theories and models that are related to technology acceptance and use. We empirically tested the proposed model by analysing data collected from targeted sample. The results of the this study revealed that PB, AW, PE and RP found to be significant factors that influence the adoption and utilisation of e-Government services provided to business sector in Saudi Arabia. Although there are some limitations in this study such as the majority of male participants and the majority of large participating business firms, this study provides useful insights into the motivations underlying the intentions to adopt and use e-Government services from the business sector perspective in developing countries like Saudi Arabia.

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Mobile computation offloading architecture for mobile augmented reality, case study: Visualization of cetacean skeleton

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Abstract—Augmented Reality applications can serve as teaching tools in different contexts of use. Augmented reality application on mobile devices can help to provide tourist information on cities or to give information on visits to museums. For example, during visits to museums of natural history, applications of augmented reality on mobile devices can be used by some visitors to interact with the skeleton of a whale. However, making rendering heavy models can be computationally infeasible on devices with limited resources such as smart phones or tablets. One solution to this problem is to use techniques to Mobile Computation Offloading. This work proposes a mobile computation offloading architecture for mobile augmented reality. This solution would allow users to interact with a whale skeleton through an augmented reality application on mobile devices. Finally testing to assess the optimization of the resources of the mobile device when performing heavy render tests were made.

Keywords—Mobile augmented reality, mobile devices, render, mobile computation offloading.

I. INTRODUCTION

Augmented reality (AR) complements the real world with computer-generated objects, its means AR is a window to the virtual world. The term mobile augmented reality (MAR) arises when integrating AR applications to mobile devices [1][2][3][4]. Initially, AR systems were very expensive and only used by small groups of researchers, such as the US Air Force to improve the efficiency of the tasks performed in flight simulation [5][6]. However, applications with AR have had great technological development and are already within reach of most people. This becomes more noticeable with the popularization of smart mobile devices, such as smartphone and tablets. The AR may serve as a teaching tool for different applications, as they can allow user interaction with 3D models or objects. Examples of these tools are medical applications for virtual surgery, or tourism information applications, among others.

The AR has been adopted in various fields and new forms of interaction between people with virtual environments are sought. The MAR is the result of the search for new forms of interaction. Visiting museums, such as the natural history may benefit from the use of mobile applications [7][8][9]. Where the use of a mobile device can be used by some museum

visitors to interact with the skeleton of a whale and learn more about how the skeletal system is made of this mammal.

Interact with the skeleton of a whale using AR on a mobile device, it may be useful for visitors to natural history museums. The MAR would allow the visualization of a cetacean. This visualization can be the complete skeleton, region, or bone by bone. So, MAR technology will be highly advantage in this type of applications. Mobile devices such as smart phones, tablets, and digital cameras, among others, have had a major technological advances and they are a standard tool of communication for its portability. However, they are devices with limited resources. For this reason, the task of rendering within a mobile device may be impractical when the number of tiles to model graph is very large.

Rendering task inside a mobile device causes a high battery consumption, because the render takes a long time processing and also use the frame buffer actively. Some studies suggest that it is one of the main causes of battery consumption in smartphone [10]. Mobile Computation Offloading (MCO) it is technique to reduce the energy and time consumption for task in the mobile devices. The task of rendering within the mobile device depends on the complexity of the 3D object to be modeled (skeletal system of the whale or whale), this complexity is measured by the number of tiles that has the 3D object. In the image 1 we can see the 3D model of the whale. Image A shows the 3D object and the image B shows the wireframe or tiles of 3D model cetacean.

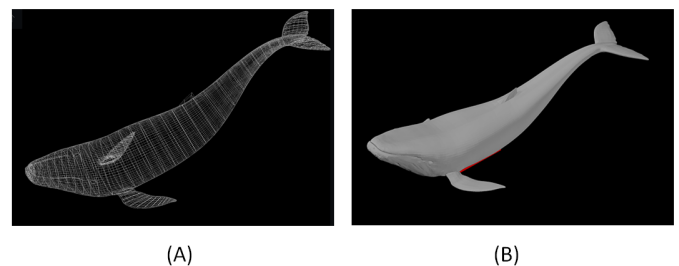


Fig. 1: 3D mesh model of the whale

The interactive visualization of the whale makes the MAR

application is computationally complex. Also, the application must be efficient, in the sense that it must provide results in very short time for the user to maintain interaction with the application. The whale has about 172 to 176 bones [11] [12]. The amount of tiles for models of bones varies between 38 and 701440 tiles. This amount of tiles for the entire skeleton model generates several problems on the mobile device: power consumption, response times not suitable for an interactive system, and using a lot of memory.

This work proposes a mobile computation offloading architecture for mobile augmented reality. Also, we perform tests to find out where it is appropriate to local or remote rendering. These tests are: energy, time and memory by applying techniques MCO seeks to have an optimized use of resources of the mobile device to the tasks of the MAR application. This solution would allow users to interact with a whale skeleton through an augmented reality application on mobile devices. Also, this solution considers the time constraints required in HCI for proper interaction between the system and the user.

The paper is organized as follows. First, a brief discussion of related work of MCO applied to MAR applications is presented. Next, the system analysis is presented. Subsequently, the proposed architecture and its justification is shown. Later, we present a section of testing and result and finally the conclusions are presented.

II. RELATED WORK

There are many mobile applications that use AR, such as "Augmented Reality", "Wikitud", "Human Pacman", among many others [13]. There are also MAR applications for museums, such as reported in [14][15][16][17]. In addition, various virtual reality applications have been developed for virtual tours in museums.

Some studies, such as [18][19][20], point to the need for MAR applications must adapt to the constraints of mobile platforms on which to run. This concept is not new, since previous work treated this adaptation as plasticity and is widely used in the context of development of type Rich-Client Applications [21].

The MCO is emerging as an alternative to optimize the use of limited resources of mobile devices. Some studies mentioned that it is important to study how to optimize the performance of mobile devices that are diverse in energy characterizing and modeling workloads to solve the weaknesses of MCO such as computational capacity, performance and energy consumption[10][22][23]. However the tests are performed for social networks, signal processing, games that require artificial intelligence module as chess or bingo, and has hardly been used for rendering.

Other studies mention that problems like energy consumption resulting from the use of graphics on the mobile device can be solved using techniques Mobile Cloud Computing (MCC), positive results in these tests are presented [24]. However quality connection Internet [25] is required, namely that it is stable, continuous, and good bandwidth and other features.

Make a combination of AR and render with MCO has several advantages as the optimization of workloads for mobile

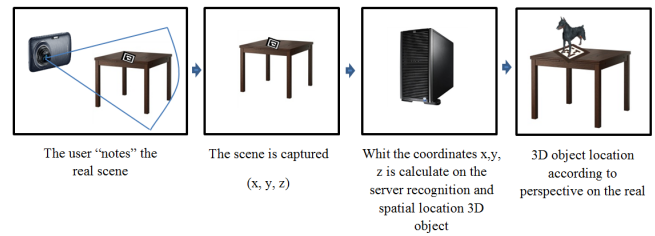


Fig. 2: Interaction system

devices, and therefore, the appropriate use of device resources. For this reason, it should be an analysis of the requirements and resources they consume more battery as a mobile device [26], the requirements that a user waiting for an RA application workloads of application RA, the bandwidth required between the mobile device and a server to render the MCO, and the dimensions of time will render the server to perform the task.

III. SYSTEM ANALISYS

On the mobile device is going to implement an AR system based on markers, as shown in Figure 2. The use of markers is widely used in mobile devices for applications with AR technique. For this system the use of the marker will allow mobile camera calculate the distance and send a reference to the server.

To be able to optimize the resources of the mobile device such as memory and processing power will be applied MCO techniques, to determine whether the rendering of the 3D object to be performed locally, ie; within the mobile device, as shown in Figure 3, or remotely (outside the mobile device). The condition to be considered may depend on the management of resources on the mobile device, amount of computation, memory, bandwidth or time when the battery runs out. MCO techniques are to be employed in order to improve performance of the mobile device. Some conditions are:

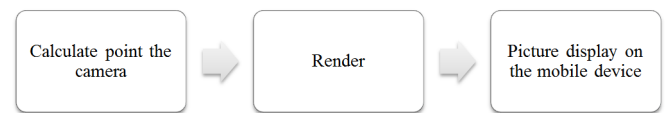


Fig. 3: System without MCO

- 1) Compute the point of the camera: The camera of the mobile device help you to calculate the benchmark image. Considering three coordinates X, Y and Z in order to display the image in the mobile device.
- 2) Render: With the data obtained from the reference point is to run the rendering process to generate an image. If the image to be displayed contains few tiles rendering to be performed locally, ie within the mobile device, otherwise MCO applied.
- 3) Display the image on the mobile device: 3D object can be displayed on the mobile device.

The number of tiles that has the model is critical to whether to apply the techniques of offloading. For this case we can apply criteria of time, power consumption or memory usage.

In the case of time and energy consumption, the program can be divided in two parts. Part inherently running on the mobile device (as it is the GUI). And the other part that may be offloading. For the case where the processing time is important, offloading is applied when the following inequality is true:

$$t_m > t_T + t_s.$$

Where t_m is the time it takes to run in the region that can be potentially for Offloading on the mobile device, t_T is assigned to the transmission time of the data required for the calculation is run out of the mobile device, and t_s is the runtime the offloading of the server. That is, when the execution time is higher in the mobile device run time out of the device.

Usually, energy consumption is related to the execution time. The energy consumed by a process es the execution time multiplied by the power of the hardware components used to perform their computations. Thus, if the energy is the criterion for offloading, then the MCO applies if the following inequality holds:

$$t_m P_m > t_T P_c + t_s P_i.$$

Where P_m is the power of the mobile processor. The values P_i and P_c are the power required to send and receive data between the mobile device and server across the network. P_c is the power required by the wireless and P_i is the power that use the network interface to wait response from server. Thus, offloading can prolong the lifetime of battery consumption migrating computational heavy tasks to servers.

Where the criterion for memory usage is important, it is necessary to consider the processes running on mobile devices may have restrictions in their address space. For example, the processes running on Dalvick in Android, can use up to 54MB [27]. In addition, many mobile devices are restricted from RAM and not use virtual memory techniques.

The strategy of MCO that we use must be calculated experimentally. This is because the task of rendering is likely to halt, and there is no direct relationship between task execution times and energy consumption multithreading [28]. It totally depends on the hardware architecture of multicore processors. The complexity of this decision increases when considering the use of graphics acceleration units such as GPUs. For our case study, the complexity depends on calculating the image and this can be translated as the computational complexity.

The user may make use of your mobile device, whether smartphone or tablet to display the 3D object of the whale (blue whale) mainly bone and skin. It is proposed that the 3D model of the whale is in the server. This will allow much of the render process is within the server if the render process is very heavy, this decision will save the energy consumption of the mobile device. Then will have a client-server operation, where the client is the mobile device, as shown in Figure 4.

- 1) Compute the point of the camera: The camera of the mobile device help you to calculate the benchmark and it will send to the server. The computer that was used to send the point of the camera requires three

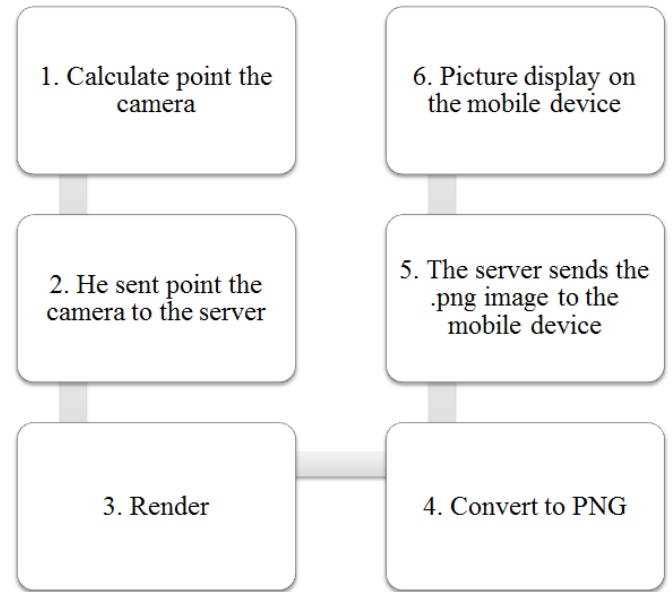


Fig. 4: System with MCO

coordinates X, Y and Z , we know that if these points are floating guy is talking about 12 bytes sent for him.

- 2) Send point the camera to the server: He sent will be made by WIFI connection type. The WIFI connection must have the following characteristics: at least 802.11a technology with a transmission speed of 54 Mbps and frequency band of 5 GHz .. However, this type of communication is not symmetric. That is, is not the same information as rising slope information.
- 3) Render: With the data obtained from the reference point of the chamber (12 bytes) is to run the rendering process to generate an image, which will be converted to PNG and then be sent to the mobile device.
- 4) Convert to PNG: The resulting image will be converted to PNG format so that the load is low and the obtained image data will not be lost at the time of conversion, the size of the image in PNG format it will be up to 3 megabytes.
- 5) He sent from server to mobile device: Once converted PNG image will be sent to the mobile device for display and interaction within the mobile device.
- 6) Mobile Device: Depending on what type of display you have the mobile device will allow determine which resolution you want the image to be received in this case is to consider the size of full definition that is 1080×1920 as main parameter. Table I shows the different types of display resolutions for tablets [29].

To describe of the sequence analysis (time constraints and data size to be transmitted. The user will send an action or event to a server using a mobile device, this action will be to send a reference point with a weight of 12 bytes. The file transmission will last approximately 0.0001 sec. Then the rendering process, which is performed; bytes are sent to an application that is within the server. The application makes

Tablet	Resolution	Size	PPI
Google Nexus 10 by Samsung	2560x1600	10.1in	300
iPad Air	2048x1536	9.7in	264
Asus Transformer	1200x1920	10.1in	224
Microsoft Surface Pro	1080x1920	10.1in	208
Samsung Galaxy Tab2 10.1	800x1280	10.1in	149
Google Nexus 7 by Asus	600x1400	7.0in	216
Samsung Galaxy Tab 2	600x1400	7.7in	170
BlackBerry PlayBack	600x1400	7.0in	169
Nook Tablet	600x800	7.0in	167

TABLE I: Tablet displays comparison

the rendering process where an image can be tif, bitmap or any other format is generated, subsequently calculated once obtained where the format is converted to PNG format. The PNG format is used because it is a lightweight format. It is noteworthy that the range obtained for the tempo of response should not exceed 0.2 seconds [30]. This time limit will be considered taking into account response time according to human perception, see Figure 5.

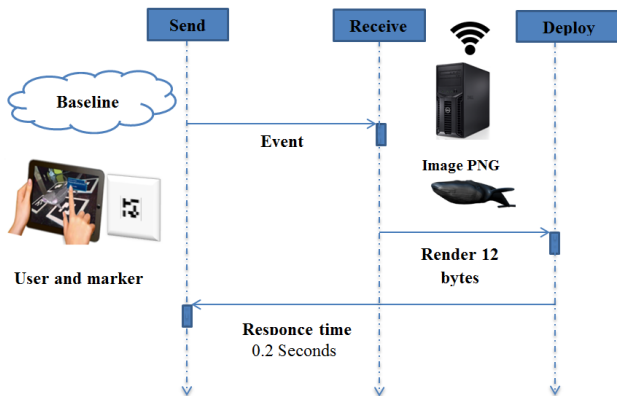


Fig. 5: Sequence diagram MCO

The problem to determine what kind of architecture for interaction is to be used is the level of detail that has the 3D object that can be rendered within or outside the mobile device, because the more higher the level of detail heavier the 3D object is made.

IV. PROPOSED ARCHITECTURE

Once that is done the analysis of system requirements is proposed that the type of connection for communication a WIFI connection with 802.11a, a transmission rate of 54 Mbps and frequency band of 5 GHz, in order to meet the benchmark 3mb camera. With this kind of connection can be up to 15 connected users, it is noteworthy that the connection speed will depend largely on the process of loading images.

It is proposed to use as Unity 3D rendering engine, this render engine is used primarily for video games. However, Unity 3D has the following characteristics: cross-platform software, creating interactive 2D and 3D content, collaborative software, among others. As the above characteristics will be of great help to the rendering process.

The minimum characteristics of the mobile device are: 7" display, dual-core processor at 1GHz, android OS, resolution of 600×1024 , 1G RAM and 3 megapixel camera.

A performance level for the display process of a whale (whale blue) may be performed as follows: When the user uses the camera of the mobile device is calculating the benchmark that will be sent to the server. This point is denoted by X, Y and Z . When the coordinate arrive at the server process may generate more two processes, which are also known as threads, the first thread stored in cache memory if the output images of the coordinate point and once was calculated, in order to be reused subsequently said images. However save images in cache may not be possible. The other thread is going to take to make the render and convert images to PNG. The thread that finishes first performing the process will be to send, see figure 6.

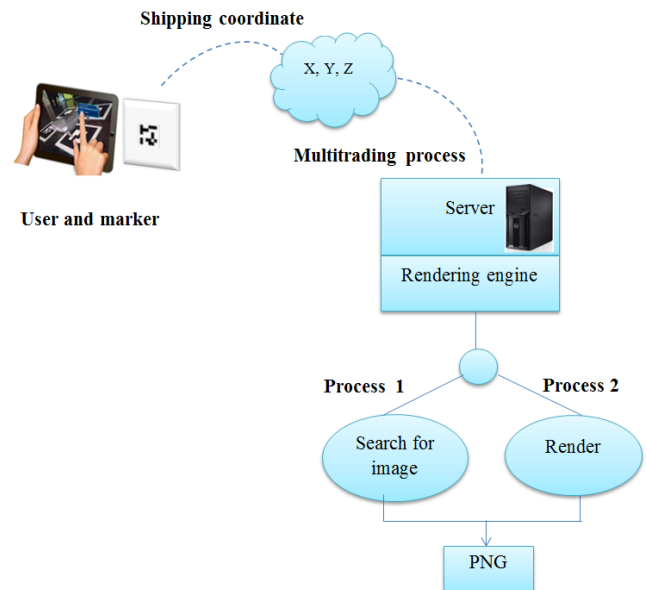


Fig. 6: Process Performance Level

The server has to be in a Red Hat Linux type either Fedora or CentOS, Filesystem minimum has to be a journalist type 3 Extension by the speed of access to the hard disk. The access must be direct route and there will be BD. Omission of a database is because it is very time consuming and would have to make a centered scheme documents for quick access, the same name of the database and distributing the directories. In one type 1.3 dual core arm cortex processor would take about 3.3 seconds while a k-5000 Nvidia 0.33 sec. A picture of 59104 tiles on a processor of these features can be 3 seconds. However if you switch to a server would be reduced to 0.33 seconds.

V. TESTS AND RESULTS

Various 3D objects are tested and compared. Every 3D object has different features among which the number of tiles and if the 3D object has with / without texture. The tests were

Name	Total 3D models	Number of tiles	Object weight	Weight assembly
Cervical	8	198	150 KB	1250 KB
Thoracic	15	230	170 KB	3010 KB
Lumbar	11	205	160 KB	1800 KB
Tail vertebrae	23	48	70 KB	2760 KB
Ribs	30	59104	160 KB	11327 KB
Escapula	2	38	739 KB	28915 KB
Humerus	2	97	210 KB	2756 KB
Radio	4	75	168 KB	1807 KB
Ulna	40	56	77 KB	3900 KB
Head	1	701440	28254 KB	28254 KB
Total	136	-	-	-
Skin	1	14960	983 KB	983 KB

TABLE II: Table of characteristics of 3D models - Blue whale.

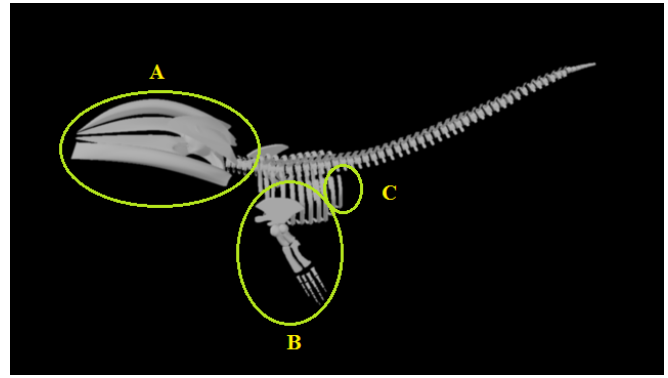


Fig. 7: 3D model of the skeletal system of the cetacean

carried out in two ways, the first is to make render within the mobile device and the second is the implementation of MCO in order to assess memory consumption, render time and energy consumption.

A. Features 3D model

3D models of the whale are developed using CAD tools and techniques of 3D modeling apply by mesh and 3D modeling by Solid. Once you have 3D models and objects are classified into two groups, the first consisting of 3D models with texture and the second non-textured 3D models.

Texture plays an important role. It is applied on the surface or 3D object model that is responsible for giving a better appearance and realism by allowing 3D models take the appearance of something real image. However, applying texture 3D models increases the complexity of the task of rendering.

As mentioned before, the skeletal system of the whale has 172 bones. However, the 3D model of the whale that was developed has 136 3D objects as some bones were simplified within the 3D object named "Skull". We can see from the table II characteristics of the models and 3D objects of cetacean "Blue whale". Render tests will be carried out taking into account the number of tiles shown in the table.

B. Render tests

For testing render 3D models are to take into account the bones are mentioned below: Fin, skull, ribs and skin. Figure 7 present the 3D model of the skeleton of the whale. The part A is the 3D model of the skull which is considered to work as a heavy 3D object, B corresponds to the assembly of the 3D model of the fin which will be considered a small ensemble. And C is the 3D model of the rib which will be a lightweight 3D object.

C. Results

We are interested in analyzing the behavior of memory, performance and energy consumptions when the render is done. Figure 8 shows memory usage for cases that we selected in this test. Figure 9 shows the energy consumption involved in rendering. And finally, Figure 10 shows the runtime to perform the render.

Figure 8 present the behavior of memory usage when the render is local and remote. In this case the memory usage

is largest than local because the PNG image size is bigger than the model 3D stored in the device. This behavior is very noticeable in the case of whale skin with and without texture. However, the PNG image size produce by the remote render does not exceed the memory available in the mobile device.

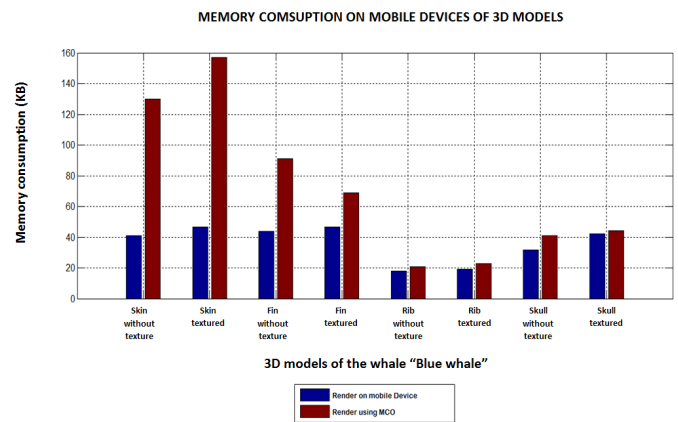


Fig. 8: Results of energy, time and memory without texture 3D models - Test your mobile device.

The energy consumption behavior is shown in Figure 9. We can notice that in all cases, remote render saving energy between 30 and 60 percent than local render.

The render time is shown in Figure 10. For this case, we are interested in the response time does not exceed 0.2 seconds, because is the HCI upper limit to respond to the user for a good experience. We can notice that local render have 6 tests that exceed this limit. For the remote render, only two cases exceed this limit. This cases are the skull with and without texture because this model is heavy to render.

In this work the case of the entire skeleton is not presented, because it takes too long to render local and presents problems of memory usage.

VI. CONCLUSION

MAR is an area that is being worked widely and consumes many computational resources. Most works that combine MCO

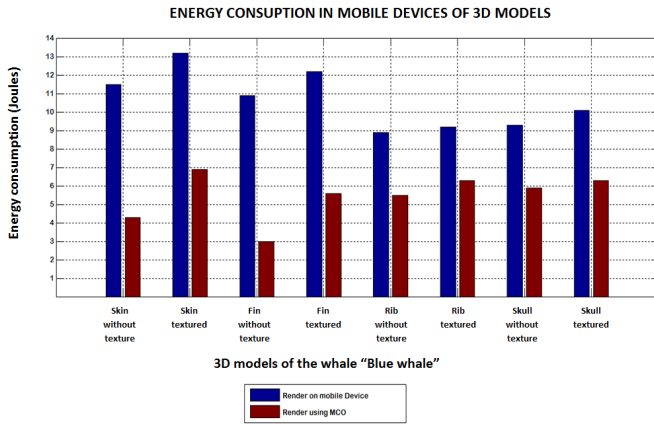


Fig. 9: Results of energy, time and memory without texture 3D models - Testing remote server.

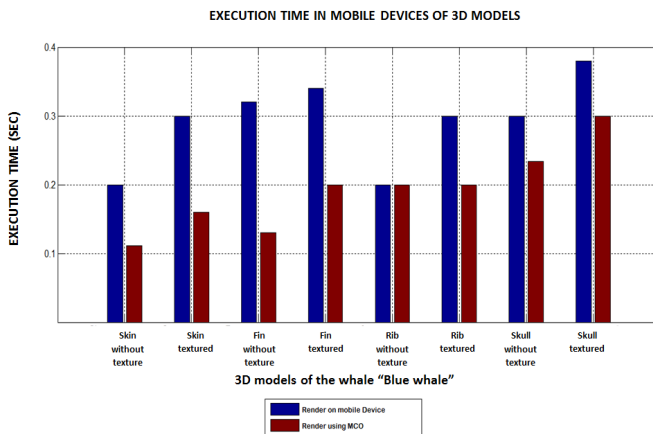


Fig. 10: Results of energy, time and memory textured 3D models - Tests on the mobile device.

and MAR have an approach to MCC. This approach generates a lack of control over resources as having: bandwidth, response times or server dependence, so if MCC is misconfigured there is no control of the internal parts of the server and you can get more expensive data transmission, in tasks where response time is critical. For our case study, it is more feasible to have the server in the museum and provide appropriate communication infrastructure. Mobile devices like tablets and phones have limited so heavy would render very time resources, battery and memory if done on the mobile device, so you should use MCO techniques. Developers have to consider mobile device restrictions such as cpu, battery and memory. Also, have to consider the constraints of HCI for interactive applications to be successful where the main constraint is time. A hardware level can consider using graphics accelerators on mobile devices. Using graphics accelerators low complexity of the task but does not know the rate of power consumption compared to a sequential program. Due to the scarcity of mathematical models to relate the behavior of time, energy, memory and updates that are in the hardware of mobile devices, the MCO decision factors would have to be done experimentally or

perform correlation models. It can be considered to generate image retrieval techniques previously processed through shortcuts to the file system on the server. Another consideration is the use of streaming technology in skeletal sent, for viewing on the MAR application as streaming to employ more bandwidth is required to optimize unloading. So the model depends more bandwidth than the service capabilities to perform the render task. Communication is the main limitation to the number of users since the server can support at least 16 concurrent users if render is done in parallel but if sent concurrently is able to have up to 64 demands render. When working with mobile devices have to be connected antenna, it is essential to know what the right size of tiles to be considered under the proposed architecture, we mention that the complexity of the 3D model can increase if we add texture. MCO techniques can save energy and improve performance on mobile systems, this technique provides a solution to the demand for limited resources, ask users when interacting with games, videos or graphics. Which has been calculated to be part the infrastructure for communications based on the number of tiles and the resulting size in megabytes of the image, has been considered a standard format Full-HD yields on average 3 megabytes. We also study the different types of algorithms for partition and download information to improve performance or energy savings. MCO is a solution to increase the limited resources of mobile devices for a single user using immediate migration of information servers. Apart from the results obtained can be performed much future work as an important part observed in the test part is that the results render the 3D model change when it is zoomed in or out. Another factor that influences the quality of the .PNG image as the image is to be rendered.

From the test of local and remote render, we can observe that the implementation of MCO could be applied using the next algorithm:

```
Render(Model3D aModel, 3DPoint cameraPoint)
BEGIN
  IF (memorySize(aModel) < LIM_APP_MEMORY AND
      responseTimeOfLocalRender < HCI_ThresholdTime AND
      energyForLocalRender < energyForMCORender)
  THEN
    localRender (Model3D aModel, 3DPoint cameraPoint)
  ELSE
    mcoRender(Model3D aModel, 3DPoint cameraPoint)
  ENDIF
END
```

Where aModel is a 3d model object, cameraPoint is the camera point for render, LIM_APP_MEMORY is the upper limit of memory for a application in the mobile device. If the memory size of the model is less than the limit of the memory of the application and the response time of the local render is less than the expected response time of HCI which is 0.2 s and the energy from the local render is less than the power remote render then it will make the 3D model rendering locally. In other case we chose the remote render.

Clearly, it is not difficult to apply the MCO because the indicators used in this work vary depending on the device. However, it gives an idea of what could be the general behavior of these applications. Contrary to what you may think, we note that there is no direct provides between time and energy consumption, as shown in the case of skin render whale with and without texture. Perhaps this change in behavior is that the

image of beauty skin framebuffer can use more than the other models. However, we note that in most cases, it's applying the MCO to do local rendering.

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Resolution Method in Linguistic Propositional Logic

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Abstract—In the present paper, the resolution method for a linguistic propositional logic with truth value in a logical algebra - refined hedge algebra, is focused. The preliminaries of refined hedge algebra are given first. Then the syntax and semantic of linguistic propositional are defined. Finally, a resolution method which based on resolution principle in two-valued logic is established. Accordingly, the research in this paper will be helpful support for the application of intelligent reasoning system based on linguistic-valued logic which includes incomparable information.

Keywords—Resolution; Linguistic Truth Value; Linguistic Propositional Logic; Hedge Algebra. .

I. INTRODUCTION

Resolution-based automated reasoning theory is an important and active research field in artificial intelligence. Since 1965, automated reasoning based on Robinson's resolution rule [1] has been extensively studied in the context of finding natural and efficient proof systems to support computational tasks. However, the resolution rule only applied in two-valued logic, so it cannot handle uncertain information, especially linguistic information. Many researchers have attent to find resolution principles in non-classical logics as effective as the resolution principles in two-valued logic.

The resolution rule, initially designed for two-valued logics by Robinson in 1965 [1], is the heart of many kinds of deductive systems such as theorem proving and logic programming. But resolution two-valued logics suffers from a drawback, it cannot handle uncertain or fuzzy information. Zadeh [2] introduced the fuzzy set theory and fuzzy logic to deal with uncertain reasoning. In fuzzy logic, a truth value domain is not the classical set {False, True} or {0, 1}, but a set of linguistic truth values [3] or the whole unit interval [0,1]. Moreover, in fuzzy logic, linguistic hedges play an essential role in the generation of the values of a linguistic variable and in the modification of fuzzy predicates [4]. Substantial works have been done for finding resolution methods in fuzzy logic as effective as the resolution principle in two-valued logic [5], [6],[7],[8],[9].

The theory of hedge algebras (HAs), introduced in Nguyen and Wechler [10], forms an algebraic approach to a natural qualitative semantics of linguistic terms in a term domain. Hedge algebra is an algebraic approach to linguistic hedges in Zadeh's fuzzy logic [11], [12]. The hedge-algebra-based semantics of linguistic terms is qualitative, relative and dependent on the order-based structure of the term domain. HAs have been shown to have a rich algebraic structure to represent linguistic domains [11], and the theory can be

effectively applied to problems such as linguistic reasoning [11] and fuzzy control [13]. Le et al. [14] introduced the fuzzy linguistic logic programming whose truth value domain is based on monotone symmetrical finite hedge algebras, then gave a procedural semantics based on many-valued modus ponens. Nguyen et al. [17],[18] presented linguistic logics with truth-valued domain based on linear symmetrical hedge algebra. Lai and Xu [15] presented a linguistic truth-valued lattice-valued propositional logic system, called $IP(X)P(X)$, whose truth value domain is a lattice implication algebra. Liu et al. [16] proposed an automated reasoning algorithm based on the linguistic valued Lukasiewicz propositional logic with truth-value in Łukasiewicz linguistic valued algebras.

Hedge algebra structure is a complete lattice but not distributive [10], and hence composed linguistic terms are not able to be expressed in the disjunction and conjunction normal forms. In addition, the structure of such algebras is rather rough. For example, let us consider the set of all possible truth values $T = \{\text{true, false, very true, very false, approximately true, possibly true, approximately true or possibly true, approximately true and possibly true, etc.}\}$. It can be shown from [19] that the linguistic term "approximately true OR possibly true" will be expressed by "approximately true" \cup "possibly true" and it equals to "true" in the structure of extended hedge algebra of the linguistic truth variable, where \cup is the join operation of this algebra. This is clearly unsuitable in nature. To overcome this problem, the so-called refined hedge algebras have been developed, based on an extension of the axiomatic system of the hedge algebras. The results in [12] showed that refined hedge algebra (RHA) of a linguistic variable with a chain of the primary terms is a distributive lattice. The lattice operations join and meet can model the semantics of the logical disjunction and conjunction. RHAs with two antonymous primary terms are called symmetrical refined hedge algebras have a rich enough algebraic structure in order to be used as a truth value domain for logical systems. Starting from this observation, we have studied a linguistic proposition logic whose the truth value domain is generated by symmetrical refined hedge algebras.

In this work, we integrate resolution rule and RHAs to construct a logical system that facilitates the representation and reasoning on knowledge expressed in natural languages. In our logical system, the set of truth values is that of linguistic ones taken from an RHA of a linguistic truth variable. Furthermore, people use finitely many degrees of quality or quantity to describe real world applications which are granulated [20], so we consider only finitely many truth values. Since the truth value domain of the logic is generated by symmetrical refined hedge algebra we can express every logical formula in the disjunction

and conjunction normal forms [12]. A resolution procedure is presented, simultaneously its soundness and completeness are proved. In addition, we introduce the notion of reliability to capture the approximate nature of resolution inference. A reliability of the conclusion of a resolution inference is a semantics value that is not greater than the reliability of premises of the resolution inference.

The paper is structured as follows: the next section gives some concepts and results about refined hedge algebra. Section III presents syntax and semantics of linguistic-valued propositional logic. Section IV presents the resolution rule and resolution procedure along with soundness and completeness results. The paper is concluded in Section V.

II. PRELIMINARIES

In this section, we will present some elementary concepts, the details can be found in the Ref [10] and [12].

Let $AX = (X, G, LH, \leq)$ be an abstract algebra where X is the term set, G is the set of generators, H is the set of hedges or modifiers, and \leq is partial order on X .

We assume that the set LH is decomposed into two subsets LH^+ , LH^- such that $LH^+ + I$ and $LH^- + I$ are finite lattices with zero-element I . Then X and LH are said to be *semantically consistent* [12] if the following conditions hold:

1. X is generated from the generators in G by means of hedges in LH , i.e. elements of X are of the form $h_n \dots h_1 a$ for $h_i \in LH$, $i = 1, \dots, n$, and $a \in G$.
2. For any $h, k \in LH^+ + I$ (respectively $LH^- + I$, $h < k$ in $LH^+ + I$ (respectively $LH^- + I$) iff $\forall x \in X((hx > x$ or $kx > x$ implies $hx < kx$) and $(hx < x$ or $kx < x$ implies $hx > kx$). And h, k are incomparable in $LH^+ + I$ (respectively $LH^- + I$) iff $\forall x \in X)(hx \neq x$ or $kx \neq x$ implies hx and kx are incomparable.

Example II.1. Let $AX = (X, G, LH, \leq)$ be an abstract algebra where $G = \{\text{True}, \text{False}\}$, $H = \{\text{Very}, \text{More}, \text{Approximately}, \text{Possibly}, \text{Less}, \text{Little}\}$ is a set of linguistic hedges. Then, the poset of values of the linguistic variable Truth X is represented in Fig. 1. Intuitively, it can be seen that $H^+ = \{\text{Very}, \text{More}\}$ and $H^- = \{\text{Less}, \text{Approximate}, \text{Possible}, \text{Little}\}$; $H^+ + I$ and $H^- + I$ are lattices given in Fig. 2. The result of applying any operation h to an element x can be understood as follows: $h\text{True}$ and $h\text{False}$ are defined to be the elements given in Fig. 1; and $khx=hx$; for all $h, k \in H$ and $x \in X$. It can easily be seen that X and H are semantically consistent.

AX is called a *refined hedge algebra* (RHA) [12], if X and LH are semantically consistent and the following conditions hold (where $h, k \in LH$):

- (1) Every operation in LH^- is converse to each operation in LH^+ .
- (2) The unit operation V of $H^+ + I$ is either positive or negative w.r.t. any operation in H . In addition, H should satisfy the PN-homogeneous property.
- (3) (Semantic independent property) If u and v are independent, i.e. $u \notin LH(v)$ and $v \notin LH(u)$, then $x \notin LH(v)$ for any $x \in LH(u)$ and vice versa. If $x \neq hx$ then

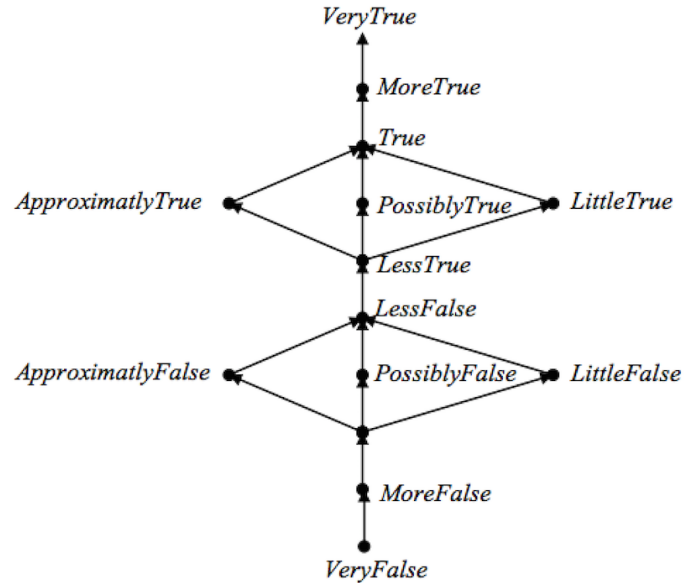


Fig. 1: A poset of values of the linguistic variable Truth.

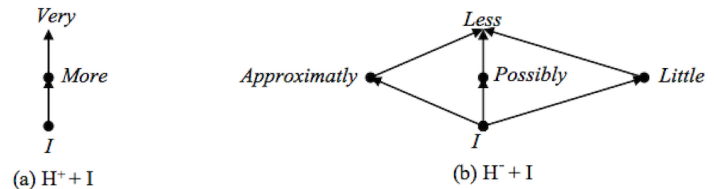


Fig. 2: Lattices of hedges.

$x \notin LH(hx)$. Further, if $hx \neq kx$ then kx and hx are independent.

- (4) (Semantic inheritance) If hx and kx are incomparable, then same for elements $u \in LH(hx)$ and $v \in LH(kx)$. Especially, if $a, b \in G$ and $a < b$ then $LH(a) < LH(b)$. And if $hx < kx$ then
 - i) In the case that $h, k \in LH_i^c$, for some $i \in SI^c$ the following statements hold:
 - $\delta hx < \delta kx$, for any $\delta \in LH^*$.
 - δhx and y are incomparable, for any $\forall y \in LH(kx)$ such that $y \not\leq \delta kx$.
 - δkx and z are incomparable, for any $\forall y \in LH(hx)$ such that $z \not\leq \delta hx$.
 - ii) Otherwise, $h, k \notin LH_i^c$, then $h'hx \leq k'kx$, for every $h', k' \in UOS$.
- (5) (Linear order between the graded classes) Assume that $u \in LH(x)$ and $u \notin LH(LH_i^c[x])$, for any $i \in I^c$. If exist $v \in LH(hx)$, for $h \in LH_i^c$ such that $u \geq v$ (or $u \leq v$), then $u \geq h'v$ (or $u \leq h'v$) for any $h' \in UOS$.

In [12] show that if $AX = (X, G, LH, \leq)$ is an RHA where G is a totally ordered set, then AX is a distributive lattice. The condition that G is a totally ordered set is reasonable because on linguistic domain the sets of generators are often

simple sets where all element are comparable (e.g. G includes two elements True, False). Operations of meet and join on AX lattice are determined recursively, based on the corresponding operations on hedge lattice LH .

Denote $LH[x] = \{hx : h \in LH + I\}$, where $x \in X$. It is proved that $LH[x]$ is a distributive sub-lattice of AX and operations of meet and join on it is defined as follows:

$$hx \vee kx = \begin{cases} (h \cup k)x & \text{if } hx \geq x, \\ (h \cap k)x & \text{if } hx \leq x \end{cases}$$

$$\text{and } hx \wedge kx = \begin{cases} (h \cap k)x & \text{if } hx \geq x, \\ (h \cup k)x & \text{if } hx \leq x \end{cases}$$

where \cup, \cap are the join and meet operators of LH , and \vee, \wedge are the join and meet operators of X .

In natural languages, there are many linguistic variables, which have only two distinct primary terms. These terms have intuitive contradictory meaning such as “true” and “false”, “old” and “young”, “tall” and “short”, etc. Therefore, we consider hedge algebras have exactly two primary generators, one of which is called positive generator, and the other is called negative generator. It seems reasonable to consider “true”, “old”, and “tall” as positive generators and “false”, “young”, and “short” as negative ones. Note that the set of generators may contain special constants such as \perp, \top , and W which are different from the primary terms and understood as “absolutely false”, “absolutely true” and the “neutral”, respectively. These constants can be characterized by the conditions that $hc = c$ for all $h \in LH, c \in \{\perp, W, \top\}$ and $\perp < W < \top$. In this paper, we shall be working with RHA $AX = \{X, G, LH, \leq\}$, where $G = \{\perp, \text{True}, W, \text{False}, \top\}$.

Let x be an element of the RHA AX , $x = h_n \dots h_1 a$ is called the canonical representation of x where $a \in \{\text{True}, \text{False}\}$. The contradictory element of x , denoted by \bar{x} , is an element y such that $y = h_n \dots h_1 a'$ where $a' \in \{\text{True}, \text{False}\}$ and $a' \neq a$. The contradictory element of \top is \perp and, conversely, the contradictory element of \perp is \top . In the case where $x = W$, we define the contradictory element of W to be just itself. For example, $\bar{x} = \text{“VeryVeryFalse”}$ is a contradictory element of $x = \text{“VeryVeryTrue”}$; $\bar{y} = \text{“VeryLittleBad”}$ is a contradictory element of $y = \text{“VeryLittleGood”}$. By the definition, it is obvious that the positive generator is a contradictory element of the negative one and vice versa and if y is a contradictory element of x then x is a contradictory element of y .

Let $AX = (X, G, LH, \leq)$ be an RHA where $G = \{\perp, \text{False}, W, \text{True}, \top\}$. Then, AX is said to be a *symmetrical RHA* provided every element $x \in X$ has a unique contradictory element in X .

It is useful to limit the set of values X only consists of finite length elements. This is entirely suitable with the practical application in natural language, which does not consider an infinite number of hedge of string. Denote $LH_p[G] = \{h_n \dots h_1 a : h_i \in LH + I, a \in G, n \leq p\}$. A symmetrical RHA $AX = (LH_p[G], G, LH, \leq)$ is a complete distributive lattice.

From now on, we consider a symmetrical RHA $AX = (LH_p[G], G, LH, \leq, \neg, \vee, \wedge, \rightarrow)$, where $G = \{\perp, \text{False}, W, \text{True}, \top\}$, \perp, \top are fixed points, W is the neutral

element, and $\perp < \text{False} < W < \text{True} < \top$.

Let x and y be two elements of the symmetrical RHA AX , then

- the negation operator is an unary operator, which is defined by $\neg x = \bar{x}$, where \bar{x} is the contradictory element of x ,
- the implication operator is a binary operator, which is defined through negative and join operators: $x \rightarrow y = \neg x \vee y$.

Theorem II.1. [12] Let $AX = (LH_p[G], G, LH, \leq)$ be a symmetrical RHA. Then, for all $x, y \in LH_p[G]$, for all $h, k \in LH$, we have:

- 1) $\neg(hx) = h\neg x$.
- 2) $\neg(\neg x) = x$.
- 3) $\neg(x \vee y) = \neg x \wedge \neg y$ and $\neg(x \wedge y) = \neg x \vee \neg y$.
- 4) $x \wedge \neg x \leq y \vee \neg y$.
- 5) $x \wedge \neg x \leq W \leq x \vee \neg x$.
- 6) $\neg \top = \perp, \neg \perp = \top$ and $\neg W = W$.
- 7) $x > y$ iff $\neg x < \neg y$.
- 8) $x \rightarrow y = \neg y \rightarrow \neg x$.
- 9) $x \rightarrow (y \rightarrow z) = y \rightarrow (x \rightarrow z)$.
- 10) $x \rightarrow y \leq x' \rightarrow y'$ if $x \leq x'$ and $y \geq y'$.
- 11) $x \rightarrow y = \top$ iff $x = \perp$ or $y = \top$.
- 12) $\top \rightarrow x = x$ and $x \rightarrow \top = \top$; $\perp \rightarrow x = \top$ and $x \rightarrow \perp = \neg x$.
- 13) $x \rightarrow y \geq W$ iff $x \leq W$ or $y \geq W$, and $x \rightarrow y \leq W$ iff $x \geq W$ or $y \leq W$.

III. LINGUISTIC PROPOSITIONAL LOGIC

In this section, we shall define the syntax and semantics of the linguistic propositional logic.

Definition III.1. An alphabet of linguistic propositional logic consists of:

- constant symbols: $\perp, \top, \text{MoreTrue}, \text{VeryFalse}, \dots$,
- propositional variables: A, B, C, \dots ,
- logical connectives: $\vee, \wedge, \rightarrow, \neg, \equiv$, and
- auxiliary symbols: $\square, (,), \dots$

Definition III.2. An atom is either a propositional variable or a constant symbol.

Definition III.3. Let A be an atom and α be a constant symbol. Then A^α is called a literal.

Definition III.4. Formulas are defined recursively as follows:

- either a literal or a constant symbol is a formula,
- if P is a formula, then $\neg P$ is a formula, and
- if P, Q are formulas, then $P \vee Q, P \wedge Q$ and $P \rightarrow Q$ are formulas.

Definition III.5. A clause is a finite disjunction of literals, which is written as $l_1 \vee l_2 \vee \dots \vee l_n$, where the symbol l_i is a literal. The empty clause is denoted by \square .

Definition III.6. A formula F is said to be in conjunctive normal form if it is a conjunction of clauses.

Definition III.7. An interpretation consists of the followings:

- a linguistic truth domain, which is a symmetrical RHA,
- for each constant in the alphabet, the assignment of an element in $LH_p[G]$,
- for each formula, the assignment of a mapping from $LH_p[G]^n$ to $LH_p[G]$.

Definition III.8. Let I be an interpretation and A be an atom such that $I(A) = \alpha_1$. Then the truth value of a literal A^α under the interpretation I is determined uniquely as follows:

- $I(A^{\alpha_2}) = \alpha_1 \wedge \alpha_2$ if $\alpha_1, \alpha_2 > W$,
- $I(A^{\alpha_2}) = \neg(\alpha_1 \vee \alpha_2)$ if $\alpha_1, \alpha_2 \leq W$,
- $I(A^{\alpha_2}) = (\neg\alpha_1) \vee \alpha_2$ if $\alpha_1 > W, \alpha_2 \leq W$, and
- $I(A^{\alpha_2}) = \alpha_1 \vee (\neg\alpha_2)$ if $\alpha_1 \leq W, \alpha_2 > W$.

Definition III.9. The truth value of formulas under an interpretation is determined recursively as follows:

- $I(P \vee Q) = I(P) \vee I(Q)$,
- $I(P \wedge Q) = I(P) \wedge I(Q)$,
- $I(\neg P) = \neg I(P)$,
- $I(P \rightarrow Q) = I(P) \rightarrow I(Q)$

The following result follows from the properties of the \wedge and \vee operators.

Theorem III.1. Let A, B and C are formulas, and I be an arbitrary interpretation. Then,

- Commutative:
 - $I(A \vee B) = I(B \vee A)$
 - $I(A \wedge B) = I(B \wedge A)$
- Associative:
 - $I((A \vee B) \vee C) = I(A \vee (B \vee C))$
 - $I((A \wedge B) \wedge C) = I(A \wedge (B \wedge C))$
- Distributive:
 - $I(A \vee (B \wedge C)) = I((A \wedge B) \vee (A \wedge C))$
 - $I(A \wedge (B \vee C)) = I((A \vee B) \wedge (A \vee C))$

Proof: The proof of the Theorem is straightforward. ■

Definition III.10. Let F be a formula and I be an interpretation. Then

- F is said to be true under interpretation I iff $I(F) > W$, I is also said to satisfy formula F , F is said to be satisfiable iff there is an interpretation I such that I satisfies F , F is said to be tautology iff it is satisfied by all interpretations;

- F is said to be false under interpretation I iff $I(F) < W$, I is also said to falsify formula F , F is said to be unsatisfiable iff it is falsified by all interpretations.

According to the definition, if $I(F) = W$ then I both satisfies and falsifies F . Further, *not satisfying* is different from *falsifying* and *not falsifying* is different from *satisfying*.

Definition III.11. Formula B is said to be a logical consequence of formula A , denoted by $A \models B$, if for all interpretation I , $I(A) > W$ implies that $I(B) > W$.

Theorem III.2. Let A and B be formulas. Then, $A \models B$ iff $\models (A \rightarrow B)$.

Proof: Assume that $A \models B$, for any interpretation I , then if $I(A) < W$, $I(\neg A) > W$; otherwise if $I(A) > W$, we recall that $A \models B$, so $I(B) > W$. Hence, $I(A \rightarrow B) = \neg I(A) \vee I(B) > W$. In other words, $\models (A \rightarrow B)$. Conversely, by a similar way, we can also show that $\models (A \rightarrow B)$ implies $A \models B$. ■

Definition III.12. Two formulas A and B are logically equivalent, denoted by $A \equiv B$, if and only if $A \models B$ and $B \models A$.

The following theorem follows from the properties of the \wedge and \vee operators and Definition III.12.

Theorem III.3. Let A, B and C be formulas. Then the following properties hold

1) Idempotency:

- $A \vee A \equiv A$
- $A \wedge A \equiv A$

2) Implication:

- $A \rightarrow B \equiv (\neg A) \vee B$
- $(A \equiv B) \equiv (A \rightarrow B) \wedge (B \rightarrow A)$

3) Double negation:

- $\neg\neg A \equiv A$

4) De Morgan:

- $\neg(A \vee B) \equiv (\neg A) \wedge (\neg B)$
- $\neg(A \wedge B) \equiv (\neg A) \vee (\neg B)$

5) Commutativity:

- $A \vee B \equiv B \vee A$
- $A \wedge B \equiv B \wedge A$

6) Associativity:

- $A \vee (B \vee C) \equiv (A \vee B) \vee C$
- $A \wedge (B \wedge C) \equiv (A \wedge B) \wedge C$

7) Distributivity:

- $A \vee (B \wedge C) \equiv (A \vee B) \wedge (A \vee C)$
- $A \wedge (B \vee C) \equiv (A \wedge B) \vee (A \wedge C)$

Proof: The proof is straightforward. ■

As mentioned previously, we will be working with resolution as the inference system of our logic. Therefore formulas need to be converted into conjunctive normal form. The following theorem, which follows from the equivalence properties in Theorem III.3, ensures that the transformation is always feasible.

Theorem III.4. *Let F be a formula of arbitrary form. Then F can be converted into an equivalent formula in conjunctive normal form.*

Proof: It is easy to prove this theorem based on the properties in the Theorem III.3 ■

Let S be a set containing exactly n atoms A_1, A_2, \dots, A_n . A *semantic tree* of S is an n -level complete binary tree, each level corresponds to an atom. The left edge of each node at the level i is assigned the label $A_i \leq W$, and the right edge of each node at the level i is assigned the label $A_i > W$ (cf. Fig 3).

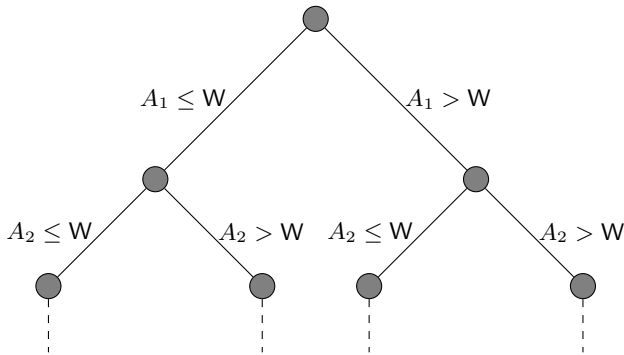


Fig. 3: Semantic tree

A set of clauses S is *failed at the node t* of a semantic tree T iff there exist an interpretation I corresponding to a branch in T which contains t , such that S is false under I . A node t is called a *failure node* of S iff S fails at t and does not fail at any node above t . A node t in a semantic tree T is called an *inference node* iff both successor nodes of t are failure nodes. If there are failure nodes for S on every branch of the corresponding semantic tree T , removing all child nodes of each failure node, we receive a *failure tree* FT .

It is important to underline that if a set of clauses S is unsatisfiable then S has a corresponding failure tree.

Lemma III.1. *There always exists an inference node on the failure tree.*

Proof: Assume that we have a failure tree FT . Because FT has a finite level, so there exists one (or more) leaf node on FT at the highest level, let say this node is called j . Let i be the parent node of j . By definition of failure tree, i cannot be failure node. Therefore, i has another child node, named k (Figure 4). If k is a failure node then i is inference node, the theorem is proved. If k is not a failure node then it has two child nodes: l, m . Clearly l, m are at higher level than j . This contradicts with the assumption that j is at the highest level.

Therefore, k is a failure node and i is an inference node. This completes the proof of the lemma. ■

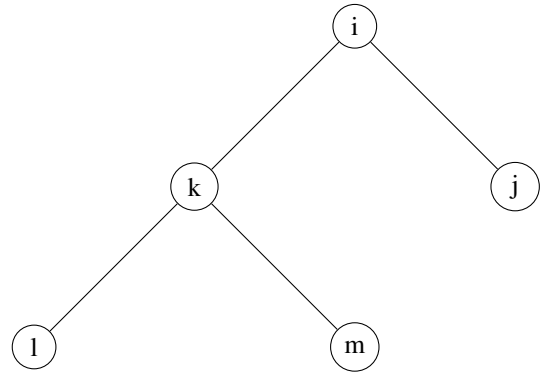


Fig. 4: Inference node on failure tree

Let FT_1 (respectively FT_2) be a failure tree of the set of clauses S_1 (respectively S_2). We denote $FT_1 \supset FT_2$ iff there exists an inference node i of FT_1 such that removing two successor nodes of i on FT_1 we receive FT_2 .

IV. RESOLUTION METHOD

We shall introduce resolution rule for automated reasoning in our linguistic propositional logic. Firstly, we recall basic notions inference rules.

An *inference* has the form:

$$\frac{C_1, C_2, \dots, C_n}{C}$$

where the clauses C_1, C_2, \dots, C_n are the premises and C is the conclusion.

In two-valued logic, when we have a set of formulas $\{A, \neg A\}$ (written as $\{A^{\text{True}}, A^{\text{False}}\}$ in our logic) then the set is said to be contradictory. However, in our logic, the degree of contradiction can vary because the truth domain contains more than two elements. Let us consider the two sets of formulas $\{A^{\text{VeryTrue}}, A^{\text{VeryFalse}}\}$ and $\{A^{\text{LittleTrue}}, A^{\text{LittleFalse}}\}$. It is obvious that the first set of formulas is “more contradictory” than the second one. The notion of reliability is hence introduced to capture the fuzziness of resolution inferences.

Let α be an element of X such that $\alpha > W$, and C be a clause. The a clause C with reliability α is the pair (C, α) . The same clauses with different reliabilities are called variants. That is (C, α) and (C, α') are called a variant of each other.

If $S = \{C_1, C_2, \dots, C_n\}$ is a set of clauses, then the reliability α of S is defined as: $\alpha = \alpha_1 \wedge \alpha_2 \wedge \dots \wedge \alpha_n$, where α_i is the reliability of the clause C_i (for $i = 1, 2, \dots, n$).

An inference rule R that works with clauses with reliability is represented as:

$$\frac{(C_1, \alpha_1), (C_2, \alpha_2), \dots, (C_n, \alpha_n)}{(C, \alpha)}$$

We call α the reliability of R , provided that $\alpha \leq \alpha_i$ for $i = 1..n$.

Definition IV.1. Define the fuzzy linguistic resolution rule as follows:

$$\frac{(A^a \vee B^{b_1}, \alpha_1), (B^{b_2} \vee C^c, \alpha_2)}{(A^a \vee C^c, \alpha_3)}$$

where b_1, b_2 and α_3 satisfy the following conditions:

$$\begin{cases} b_1 \wedge b_2 \leq W, \\ b_1 \vee b_2 > W, \\ \alpha_3 = f(\alpha_1, \alpha_2, b_1, b_2) \end{cases}$$

with f is a function ensuring that $\alpha_3 \leq \alpha_1$ and $\alpha_3 \leq \alpha_2$.

In Definition IV.1 α_3 is defined so as to be smaller or equal to both α_1 and α_2 . This makes sure that the more inferences we need to deduce a clause the less reliable the obtained clause is. There are different ways to define α_3 . Below we define α_3 in three ways based on \wedge and \vee operators.

The reliability of the resolution rule IV.1 based on \wedge operator is given by:

$$\alpha_3 = f(\alpha_1, \alpha_2, b_1, b_2) = \alpha_1 \wedge \alpha_2 \wedge \neg(b_1 \wedge b_2) \quad (1)$$

The reliability of the resolution rule IV.1 based on \vee operator is given by:

$$\alpha_3 = f(\alpha_1, \alpha_2, b_1, b_2) = \alpha_1 \wedge \alpha_2 \wedge (b_1 \vee b_2) \quad (2)$$

The reliability of the resolution rule IV.1 based on the combination of \wedge and \vee operators is given by:

$$\alpha_3 = f(\alpha_1, \alpha_2, b_1, b_2) = \alpha_1 \wedge \alpha_2 \wedge (\neg(b_1 \wedge b_2)) \wedge (b_1 \vee b_2) \quad (3)$$

Proposition IV.1. The reliability based on \wedge operator (respectively \vee operator or the combination of \wedge and \vee operators) satisfies the conditions on α_3 in Definition IV.1.

Proof: It is clear that that $\alpha_3 \leq \alpha_1$ and $\alpha_3 \leq \alpha_2$, as well as α_3 depends on b_1, b_2 . Additionally, it is clear that $\alpha_1, \alpha_2 \in X^+$. Moreover, $b_1 \wedge b_2 \leq W$ implies $\neg(b_1 \wedge b_2) > W$. Then, by Formula (1), we have $\alpha_3 > W$.

The proof for \vee operator is similar.

For the combination of \wedge and \vee operators, we have

$$\begin{aligned} \alpha_3 &= \alpha_1 \wedge \alpha_2 \wedge (\neg(b_1 \wedge b_2)) \wedge (b_1 \vee b_2) \\ &= (\alpha_1 \wedge \alpha_2 \wedge \neg(b_1 \wedge b_2)) \wedge (\alpha_1 \wedge \alpha_2 \wedge (b_1 \vee b_2)) \end{aligned}$$

Then applying the results of \wedge operator and \vee operator we have $\alpha_3 > W$. ■

Theorem IV.1. The fuzzy linguistic resolution rule IV.1 is sound.

Proof: We need to prove that for any interpretation I , if $I((A^a \vee B^{b_1}) \wedge (B^{b_2} \vee C^c)) > W$ then $I(A^a \vee C^c) > W$. We have that

$$\begin{aligned} &I((A^a \vee B^{b_1}) \wedge (B^{b_2} \vee C^c)) \\ &= I((A^a \wedge B^{b_2}) \vee (A^a \wedge C^c) \vee (B^{b_1} \wedge B^{b_2}) \vee (B^{b_1} \wedge C^c)) \\ &= I(A^a \wedge B^{b_2}) \vee I(A^a \wedge C^c) \vee I(B^{b_1} \wedge B^{b_2}) \vee I(B^{b_1} \wedge C^c) \end{aligned}$$

It is easy to show that:

- $I(A^a \wedge B^{b_2}) \leq I(A^a) \leq I(A^a \vee C^c)$,
- $I(A^a \wedge C^c) \leq I(A^a \vee C^c)$,
- $I(B^{b_1} \wedge C^c) \leq I(C^c) \leq I(A^a \vee C^c)$, and
- $I(B^{b_1} \wedge B^{b_2}) \leq W$

So, if $I(A^a \vee C^c) \leq W$ then we must have that

$$I(A^a \wedge B^{b_2}) \vee I(A^a \wedge C^c) \vee I(B^{b_1} \wedge B^{b_2}) \vee I(B^{b_1} \wedge C^c) \leq W$$

which contradicts with the initial assumption. This completes the proof of the theorem. ■

Definition IV.2. A linguistic resolution derivation as a sequence of the form S_0, \dots, S_i, \dots , where

- each S_i is a set of clauses with reliability, and
- S_{i+1} is obtained by adding the conclusion of a fuzzy linguistic resolution inference with premises from S_i , that is $S_{i+1} = S_i \cup \{(C, \alpha)\}$, where (C, α) is the conclusion of the fuzzy linguistic resolution

$$\frac{(C_1, \alpha_1), (C_2, \alpha_2)}{(C, \alpha)}$$

and $(C_1, \alpha_1), (C_2, \alpha_2) \in S_i$.

The following theorem shows that the resolution procedure in the Def.IV.2 is sound and complete.

Theorem IV.2. Let S_0, \dots, S_i, \dots be a fuzzy linguistic resolution derivation. S_n contains the empty clause (for some $n = 0, 1, \dots$) iff S_0 is unsatisfiable.

Proof: (\Rightarrow) If S_n contains the empty clause, then S_n is false under any interpretation. By Theorem IV.1, we have S_{n-1} is false under any interpretation, too. Similarly, S_{n-1}, \dots, S_1, S_0 are also false under any interpretation.

(\Leftarrow) Because S_0 is unsatisfiable, there is a corresponding failure tree FT . By Lemma III.1, there exists an inference node i on FT with two child nodes j, k . Assuming that the label of edge $i - j$ is $A \leq W$ and the label of edge $i - k$ is $A > W$. The interpretation corresponding to the branch contains the edge $i - j$ and terminating at j makes S_0 satisfiable. So, there is at least one clause in S_0 containing the literal A^{α_1} where $\alpha_1 \leq W$, let say C_1 . Similarly, there exists at least one clause in S_0 containing the literal A^{α_2} where $\alpha_2 > W$, we name it C_2 . Applying the resolution rule in the Def. IV.1:

$$\frac{C_1, C_2}{C_3}$$

C_3 does not contain atom A , so that C_3 is false under all interpretations containing A . Thus, failure tree FT_1 of the clause set $S_1 = S_0 \cup C_3$ does not contain node j, k , this means $FT \supset FT_1$.

By applying resolution procedure, there exist failure trees FT_2, FT_3, \dots of the sets of clauses S_2, S_3, \dots such that $FT \supset FT_1 \supset FT_2 \supset FT_3 \supset \dots$. Because there are only a finite number of nodes in FT , then exists some n satisfying: $FT_n = \square$ (i.e. $FT \supset FT_1 \supset FT_2 \supset FT_3 \supset \dots \supset FT_n = \square$). Only the empty clause is false under the empty interpretation. This means that the set of clauses S_n (S_n corresponds to FT_n) contains the empty clause. ■

A resolution proof is of a clause C from a set of clauses S consists of repeated application of the resolution rule to derive the clause C from the set S . If C is the empty clause then the proof is called a *resolution refutation*. We will represent resolution proofs as *resolution trees*. Each tree node is labeled with a clause. There must be a single node that has no child node, labeled with the conclusion clause, we call it the root node. All nodes with no parent node are labeled with clauses from the initial set S . All other nodes must have two parents and are labeled with a clause C such that

$$\frac{C_1, C_2}{C}$$

C where C_1, C_2 are the labels of the two parent nodes. If RT is a resolution tree representing the proof of a clause with reliability (C, α) , then we say that RT has the reliability α .

Example IV.1. Let $AX = (LH_p[G], G, LH, \leq, \neg, \vee, \wedge, \rightarrow)$ be a symmetrical RHA, where

- $LH^+ = \{\text{Very, More}\},$
- $LH^- = \{\text{Approximately, Little}\},$
- $G = \{\perp, \text{False}, W, \text{True}, \top\},$
- \perp, \top are the smallest and biggest elements, W is the neutral element, and $\perp < \text{False} < W < \text{True} < \top$.

Assume that we have the set of clauses:

- 1) $A^{\text{ApproximatelyTrue}} \vee B^{\text{MoreTrue}}$
- 2) A^{False}
- 3) $B^{\text{VeryFalse}} \vee C^{\text{True}}$
- 4) $C^{\text{LittleFalse}}$

At the beginning, we shall assign each clause to the highest reliability \top . We consider the reliability based on \vee operator. The resolution refutation is found as following:

$$\frac{\frac{(A^{\text{ApproximatelyTrue}} \vee B^{\text{MoreTrue}}, \top) (A^{\text{False}}, \top)}{(B^{\text{VeryTrue}}, \text{ApproximatelyTrue})} \quad \frac{(B^{\text{VeryFalse}} \vee C^{\text{True}}, \top)}{(C^{\text{True}}, \text{ApproximatelyTrue})}}{\frac{(C^{\text{True}}, \text{ApproximatelyTrue}) (C^{\text{LittleFalse}}, \top)}{(C^{\text{True}}, \text{ApproximatelyTrue})}} (\square, \text{ApproximatelyTrue})$$

V. CONCLUSION

We have presented the resolution method in linguistic truth-valued propositional logic based on linguistic truth-valued symmetrical RHA. The syntax and semantics of our logic are defined. The resolution rule with the reliability is given. The soundness and completeness are proved. The presented work provided a key theoretical support for a resolution-based automated reasoning approaches in linguistic truth-valued logic based on RHA. Accordingly, this research work supports linguistic information processing which can directly focus on linguistic values. Further research can be carried out on the practical implementation of the proposed resolution-based automated reasoning procedure as well as its application in intelligent reasoning and decisions making.

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Single-Handed Cursor Control Technique Optimized for Rear Touch Operation and Its Usability

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Abstract—To improve single-handed operation of mobile devices, the use of rear touch panel has potential for user interactions. In this paper, a basic study of operational control simply achieved through drag and tap of the index finger on a rear touch panel is conducted. Since a user has to hold the handheld device firmly with the thumb and fingers, a movable range of the tip of an index finger is limited. This restriction requires a user to perform several times of dragging actions to reach a cursor to the long distance target. Considering such kinematic restriction, a technique optimized for rear operation is proposed, wherein not only the position but also the velocity of fingertip movement is regarded. Movement time, the number of dragging operation, and the throughputs of the proposed technique have been evaluated in comparison with the generic technique using Fitts's law. Experiments have been conducted to perform the target selection in the form of reciprocal 1D pointing tasks with ten participants. The combinations of two ways of holding the device (landscape and portrait) and two directions of dragging (horizontal and vertical) are considered. As a result, the proposed technique achieved the improvements of from 5 to 13% shorter movement time, from 20 to 40% higher throughputs and no deterioration of the number of dragging even for the longer distance targets. In addition, the further analysis addressed that there exists the advantageous combinations of the way of holding and the direction of dragging, which would be beneficial for better design of single-handed user interactions using rear touch.

Keywords—Rear touch; cursor control; mobile device; single-handed; Fitts's law

I. INTRODUCTION

Many types of mobile devices have appeared on the market in recent years. Such devices satisfy various inherent needs and diverse lifestyles. Mobile game consoles, which have a longer history than cell phones, are also enlivening the market. In general, two hands are required to use such devices; the device is held in one hand and operated with the other. However, sometimes only one hand is free to operate the device. In addition, as display sizes increase, single-handed thumb-touch operations become more difficult. To address these issues, we have considered the use of rear touch and proposed the technique to fully utilize the limited movable range of the tip of an index finger. Using a portable gaming device with a rear touch panel, we demonstrate a practical single-handed input technique for mobile devices. The purpose of this study is not to compare the proposed technique to existing bimanual methods, such as generic front touch operations. The benefit of the use of rear touch is that it works sufficiently even when

the generic modes of operation are not stable or available. Another benefit is no blockage of the display. In generic front touch operation, occlusion and selection point ambiguity are problematic, and the use of rear touch has potential to be the solution.

The remainder of this paper is organized as follows. Related work is presented in Section II. A design for rear touch operation, which addresses the generic technique and the proposed rear touch oriented technique, is described in Section III. Methods of experiments are presented in Section IV, and results of performance evaluation are shown in Section V. Moreover, discussions are presented in Section VI, and conclusions and future works are given in Section VII.

II. RELATED WORK

A. Rear Touch

In recent touchscreen-based mobile devices, there is a challenging trade-off between visual expressivity and ease of interaction. Information-rich interfaces on a small display result in a target size that requires thumb and finger operations. Occlusion and selection point ambiguity are problematic, and a number of studies have addressed these problems [1], [2], [3], [4]. On the other hand, rear touch technology is a recent development, and such technology would greatly benefit tablet-based devices where the front touch panel occupies most of the surface area and leaves little space for other input methods. Practical research for this technology is limited [5], [6], [7]. We find enhanced studies for the simultaneous use of the front and the back of the device [8], [9], [10], and the text input [11], [12]. Reference [13] conducted the performance analysis of the rear touch and found that the index finger performed well on the small-sized display device. Recently, [14], [15] explored the relationship between hand grip from the back of the device and user interactions. With regard to single-handed rear touch panel operation, different ways to hold hand-held devices have been investigated, and five spontaneous ways to hold tablets have been described [16] and analyzed using a kinematic chain model [17]. A kinematic chain approach is appropriate for considering holding methods that involve frame, support, and interact in the kinematic chain. However, in this study, we do not discuss different ways to hold mobile devices in detail. Rather, we evaluate two cursor control techniques in relation to particular holding methods, landscape and portrait orientations. At present, most hand-held tablets are thin rectangular-shaped

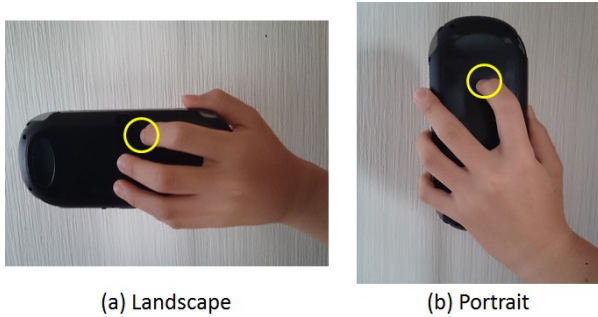


Fig. 1: Ways of holding the device

devices weighing 250-600 g with a 7-10 inch display. Smart phones, which are also rectangular, typically weigh 100-200 g with a 4-5.5 inch display. The trend is toward larger displays and reduced weight. In general, devices that weigh less are easier to hold for a long time with one hand and can be held more easily in various ways. We assume that hand-held devices that weigh up to 200 g will not result in user stress. However, this assumption should be carefully examined from various perspectives, and will be considered in future studies. In this study, we define two particular holding methods and evaluate two cursor control techniques using a hand-held gaming device that weighs 219 g.

B. Target Selection

In this study, the performance of two cursor control techniques are evaluated using Fitts's classic 1D reciprocal-pointing paradigm [18] as modified by MacKenzie [19]. Although it dealt with traditional pointing devices, such as a mouse and a stylus, we applied this evaluation norm for these rear touch techniques. Moreover, it does not fully indicate the usability or the practicability of systems; nevertheless, we applied Fitts's law to assess the fundamental performance. Fitts's law states that target acquisition time, or movement time (MT), in graphical user interfaces is almost entirely determined by the ratio of target distance D and target width W as follows:

$$MT = a + b \log_2(D/W + 1), \quad (1)$$

where a and b are constants determined through linear regression. The expression $\log_2(D/W + 1)$ defines what Fitts referred to as the index of difficulty (ID) and carries the unit "bits." If MT is measured in seconds, then the unit for a is "s" and the unit for b is "s/bit." The reciprocal of b defines what Fitts termed as the index of performance (IP) in "bits/s." IP is the rate of human information processing for the movement task under investigation and is referred to as bandwidth or throughput.

III. DESIGN FOR REAR TOUCH OPERATION

Two ways of holding tablets are considered. Fig. 1 shows the ways of holding the device in landscape and portrait orientation, respectively. The yellow circles in Fig. 1 indicate movable ranges of the tip of an index finger. In both orientations, they are limited in the size of approximately from 10 to 15 mm (around 0.5 inch) in diameter. On the other hand,

the display size of hand-held devices is fairly large in size of from 4 to 10 inch. Therefore, an input within a limited range has to control a cursor which covers from 8 to 20 times wider range on the display.

A. Generic Technique

In this study, two cursor control techniques using rear touch are evaluated. One is the generic technique known as the standard touch pad control, wherein touch movement is linearly applied for cursor movement. The mapping algorithm from the touch position to the cursor position is expressed as follows:

$$q(k) - q(0) = M_p(p(k) - p(0)), \quad (2)$$

where $k > 0$ is the sampling number (sampling frequency was 60 Hz in the implementation), $p(k) \in R^{2 \times 1}$ is the current touch position, $p(0)$ is the start position of dragging, $q(k) \in R^{2 \times 1}$ is the current cursor position, $q(0)$ is the initial cursor position associated with $p(0)$, and $M_p \in R^{2 \times 2}$ is the appropriate constant diagonal matrix. Note that $p(k)$ exists in a rear touch pad coordinate system, and $q(k)$ exists in a display coordinate system. The action of repeating dragging achieves the long distance movement of a cursor. In the implementation design, we used

$$M_p = \begin{pmatrix} 6 & 0 \\ 0 & 6 \end{pmatrix} \quad (3)$$

in consideration with a balance of the gain of cursor movement and the cursor positioning accuracy.

B. Rear Touch Oriented Technique

Considering the kinematic restriction of finger movement on the rear face, a technique optimized for rear operation is proposed. The proposed technique achieves long distance cursor movement by small distance fingertip movement. Here, a velocity of fingertip movement is considered so that the faster dragging leads to the longer distance movement of a cursor. The proposed mapping algorithm from the touch position to the cursor position is expressed as follows:

$$q(k) - q(0) = M_p(p(k) - p(0)) + M_v(p(k) - p(k-1))^2, \quad (4)$$

where $M_v \in R^{2 \times 2}$ is the appropriate constant diagonal matrix. In the implementation design, we used

$$M_v = \begin{pmatrix} 1/288 & 0 \\ 0 & 1/288 \end{pmatrix}. \quad (5)$$

In addition, we applied the limiter for $p(k) - p(k-1)$ to avoid the excessively fast movement of a cursor, which was set as approximately 0.53 mm/sample in the implementation.

IV. METHOD

A. Apparatus

Implementation and evaluation were conducted on a PS Vita. The PS Vita has a front 5 inch organic light-emitting diode touch panel display with a resolution of 960×544 pixels, which corresponds to a single pixel size of approximately 0.115 mm. On the back, the PS Vita has a capacitive-sensing touch panel with a maximum of six touch points. The experimental application was run at the sampling rate of 60 frames/s.

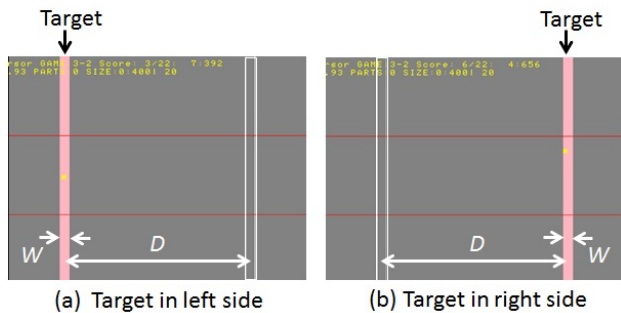


Fig. 2: Target selection task ($D = 400$, $W = 20$)

B. Participants

Ten volunteers (4 female, 6 male; 14-50 years) participated. All were right-handed, and held the device and operated rear touch with their left hand. All participants regularly use smartphones and/or tablet devices.

C. Procedure

The performance of the target selection operation was evaluated using Fitts's law 1D reciprocal-pointing paradigm. The 1D pointing task required participants to select two fixed-sized targets repeatedly in succession (Fig. 2). The targets were rendered as solid rectangles equidistant from the center of the display in opposite directions along the horizontal or vertical axis. The targets to be selected were colored red, and they appeared one after another on opposite sides of the display. In addition, the target color changed to light pink when the cursor was on the target. When participants correctly selected a target, the current target briefly became white and disappeared; then, the next target would appear on the opposite side of the display. Participants moved a cursor by performing drag operation on a rear touch panel (repeatedly, if necessary), and tapped to accomplish a selection. A task was continued until participants completed a selection successfully, and therefore, a faulty selection did not occur (no errors). Instead of error rates, we counted and evaluated the number of dragging actions per task.

D. Independent Variables

The performance of the generic technique and the proposed technique was evaluated. We specifically focused on the way of holding the handheld device and the direction to move the tip of an index finger. Therefore, the independent variables were the cursor control technique *TEC*, the way of holding the device *HLD*, the direction of dragging *DIR*, the target distance D , and the target width W .

E. Dependent Variables

The main dependent variable is *MT*, which is defined as the time taken to move to and select the active red target. The other main dependent variable is the number of dragging operation *NUM*.

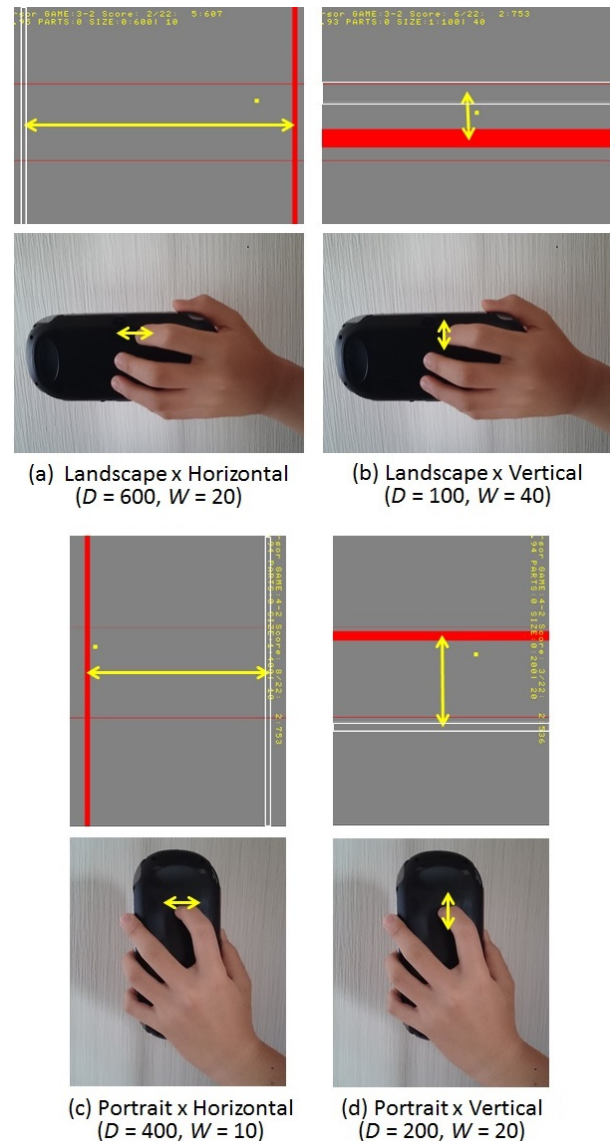


Fig. 3: The directions of fingertip movement and associated cursor movement for four models ($HLD \times DIR$)

F. Design

The independent variables were the cursor control technique *TEC* ("G"eneric technique, "A"ccelerated technique), the way of holding *HLD* ("L"andscape, "P"ortrait), the direction of dragging *DIR* ("H"orizontal, "V"ertical), the target distance D (100, 200, 400, 600 pixels), and the target width W (10, 20, 40 pixels). The target distance $D = 600$ was available only for the models of LH ($HLD=L$, $DIR=H$) and PV ($HLD=P$, $DIR=V$) due to the display size limitation. Here, we express each factor for *TEC*, *HLD* and *DIR* as the first character of each word, hence, $HLD=L$ means $HLD=L$ andscape. A fully crossed design resulted in 84 combinations of *TEC*, *HLD*, *DIR*, D , and W . Fig. 3 shows images of four models LH, LV, PH and PV ($HLD \times DIR$). The lower photographs show the way of holding the device in each orientation, wherein the directions of movement of the tip

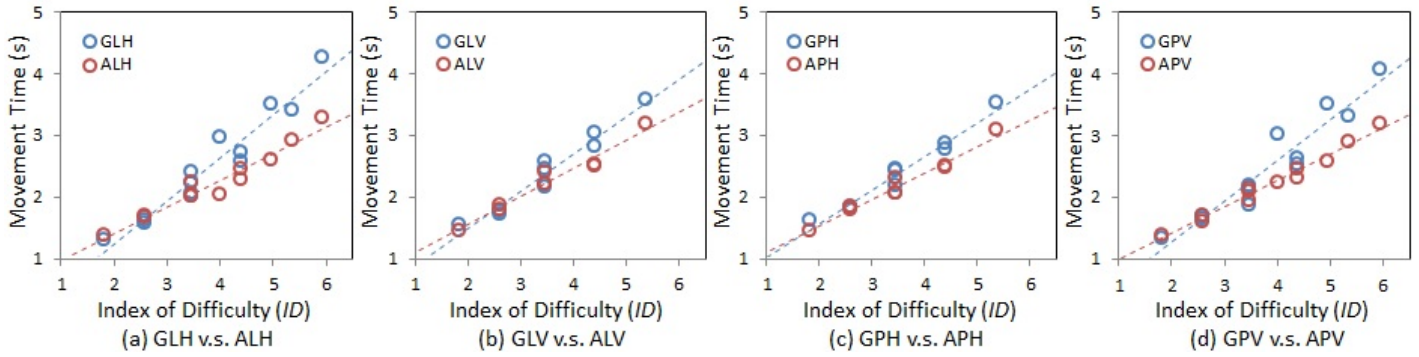


Fig. 4: Movement time by ID for eight models

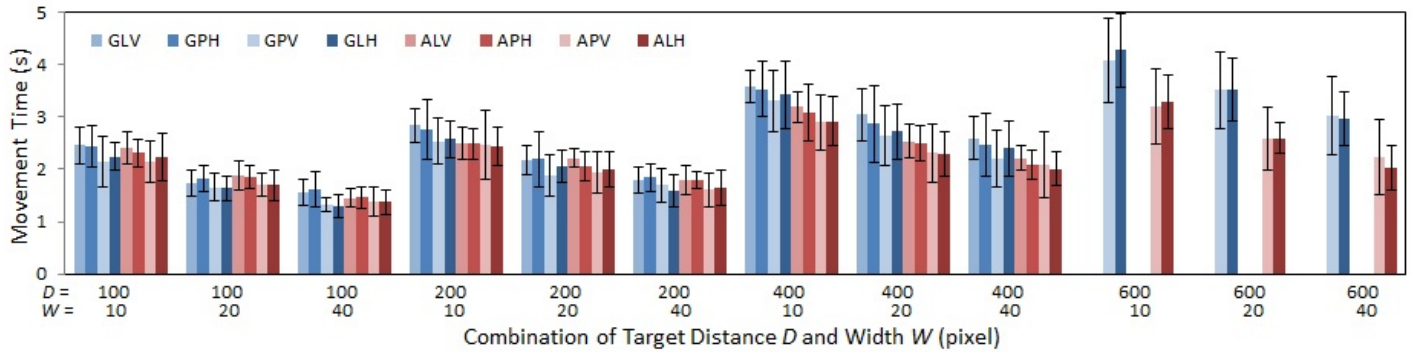


Fig. 5: Movement time by W and D for eight models

of an index finger are indicated as yellow arrows. The upper images show the display image of the experimental application, wherein the targets are colored red and the directions of cursor movement are indicated as yellow arrows.

The experiment was organized into four sessions for eight models MDL (GLH, GLV, ALH, ALV, GPH, GPV, APH, APV). The participants were separated into two teams (five participants for each). For each team, sessions were organized as follows:

	Team 1	Team 2
Session 1:	GLH→ALH	GPV→APV
Session 2:	ALV→GLV	APH→GPH
Session 3:	GPV→APV	GLH→ALH
Session 4:	APH→GPH	ALV→GLV

Here, GLH→ALH means that participants attempt all trials of GLH first and those of ALH second in a session. Each participant performed the trials in a single session that lasted approximately 60 min. Prior to starting the session, the participants were allowed 5 min of practice. The session was broken up by MDL , with nine or twelve trial sets ($D \times W$) completed for each MDL . The nine or twelve trial sets were separated into three blocks to counterbalance order, fatigue, and the effects of practice.

- Block 1: $D = 100, W = 10 / D = 200, W = 20 / D = 400, W = 40 / D = 400, W = 10$
- Block 2: $D = 100, W = 20 / D = 200, W = 40 / D = 400, W = 10 / D = 400, W = 20$
- Block 3: $D = 100, W = 40 / D = 200, W = 10 / D = 400, W = 20 / D = 400, W = 40$

TABLE I: Regression Coefficients for Eight Models

Model	r^2	Regression Coefficients		
		Intercept a (s)	Slope b (s/bit)	IP (bits/s)
GLH	0.9397	0.156	0.703	1.4
GLV	0.9550	0.312	0.602	1.7
GPH	0.9648	0.484	0.546	1.8
GPV	0.9147	0.060	0.663	1.5
ALH	0.9453	0.550	0.433	2.3
ALV	0.9603	0.654	0.454	2.2
APH	0.9636	0.684	0.429	2.3
APV	0.9777	0.557	0.429	2.3

Here, the trial sets of $D = 600$ were conducted only for the four models GLH, GPV, ALH and APV. In each block, the participants completed trial sets for three or four combinations of D and W that were presented randomly. A 5 min break was taken between each block; therefore, there were six blocks of trials and five breaks. A trial set considered 12 selection tasks, i.e., 11 reciprocal movements between 12 targets. The data of the first two selection tasks were discarded.

V. RESULTS

MT and NUM for the eight models were evaluated. As $D = 600$ was available only for the four models, we define two categories where the combinations of MDL , D and W

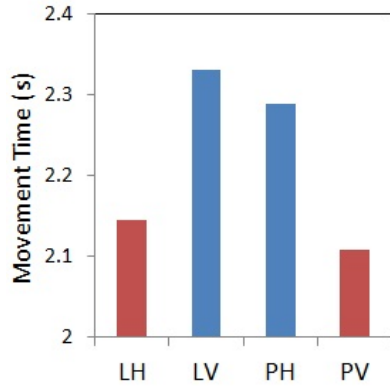


Fig. 6: MT by $HLD \times DIR$

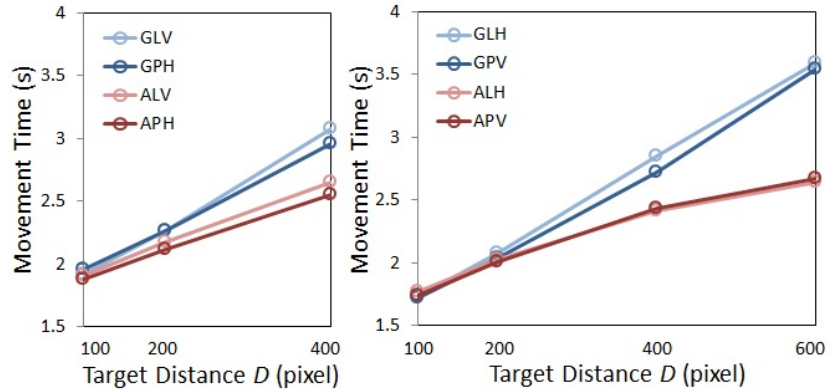


Fig. 7: MT by D for eight models

are defined as follows:

Category 1 ($CAT1$):

$MDL = GLH, GLV, GPH, GPV, ALH, ALV, APH, APV$

$D = 100, 200, 400$

$W = 10, 20, 40$

Category 2 ($CAT2$):

$MDL = GLH, GPV, ALH, APV$

$D = 100, 200, 400, 600$

$W = 10, 20, 40$

A. Movement Time

Fig. 4 plots MT of the eight models as a function of ID . The results of linear regression analysis for each plot are given in Table I. The fits between the Fitts's law predictions and the data collected in the rear-touch-based cursor control operations yielded r^2 values above 0.91. Moreover, from 20 to 40% improvements are found in throughputs (IP) by using the proposed technique. The mean MT in $CAT1$ were 2.220 s for GLH, 2.419 s for GLV, 2.394 s for GPH, 2.151 s for GPV, 2.081 s for ALH, 2.244 s for ALV, 2.184 s for APH, and 2.061 s for APV. The mean MT in $CAT2$ were 2.563 s for GLH, 2.503 s for GPV, 2.221 s for ALH, and 2.214 s for APV. Fig. 5 shows all MT plots by D and W for the eight models, wherein the models are roughly arranged in descending order for each TEC . Repeated measure analysis of variance in $CAT1$ showed a significant main effect for TEC ($F_{1,9} = 49.62, p < 0.0001$), D ($F_{2,18} = 958.90, p < 0.0001$), and W ($F_{2,18} = 436.29, p < 0.0001$). The following interaction effects were observed: $HLD \times DIR$ ($F_{1,9} = 39.48, p < 0.0001$). Fig. 6 shows MT for LH, LV, PH and PV ($HLD \times DIR$). Apparently, MT of LH and PV is shorter than that of LV and PH. This result implies that fingertip movement of the longitudinal direction would be easier than that of the transverse direction. Repeated measure analysis of variance in $CAT2$ showed a significant main effect for TEC ($F_{1,9} = 280.72, p < 0.0001$), D ($F_{3,27} = 373.00, p < 0.0001$), and W ($F_{2,18} = 284.19, p < 0.0001$). The following interaction effects were observed: $TEC \times D$ ($F_{3,27} = 63.52, p < 0.0001$) and $D \times W$ ($F_{6,54} = 8.53, p < 0.0001$). Fig. 7 shows MT by D for the eight models. Evidently, MT was improved by the proposed technique. As a result, the mean MT by the proposed technique was from 5 to 9% shorter in $CAT1$ and approximately

13% shorter in $CAT2$ than that of the generic technique. This tendency is particularly noticeable at $D = 600$, so that more than 25% of improvements are observed.

B. The Number of Dragging

The mean NUM in $CAT1$ were 1.337 for GLH, 1.380 for GLV, 1.140 for GPH, 1.091 for GPV, 1.048 for ALH, 1.084 for ALV, 1.064 for APH, and 1.042 for APV. The mean NUM in $CAT2$ were 1.563 for GLH, 1.336 for GPV, 1.063 for ALH, and 1.054 for APV. Fig. 8 shows all NUM plots by D and W for the eight models. Repeated measure analysis of variance in $CAT1$ showed a significant main effect for TEC ($F_{1,9} = 122.10, p < 0.0001$), HLD ($F_{1,9} = 150.21, p < 0.0001$), D ($F_{2,18} = 198.48, p < 0.0001$), and W ($F_{2,18} = 116.79, p < 0.0001$). The following interaction effects were observed: $TEC \times HLD$ ($F_{1,9} = 295.41, p < 0.0001$), $TEC \times D$ ($F_{2,18} = 186.05, p < 0.0001$), $HLD \times D$ ($F_{2,18} = 97.55, p < 0.0001$), $D \times W$ ($F_{4,36} = 7.93, p = 0.0001$), and $TEC \times HLD \times D$ ($F_{2,18} = 124.53, p < 0.0001$). Repeated measure analysis of variance in $CAT2$ showed a significant main effect for TEC ($F_{1,9} = 280.72, p < 0.0001$), D ($F_{3,27} = 373.00, p < 0.0001$), and W ($F_{2,18} = 284.19, p < 0.0001$). The following interaction effects were observed: $TEC \times D$ ($F_{3,27} = 63.52, p < 0.0001$), and $D \times W$ ($F_{6,54} = 8.53, p < 0.0001$). Fig. 9 shows interaction plots of $TEC \times HLD$ by D , where $GL = GLH + GLV$, $GP = GPH + GPV$, $AL = ALH + ALV$, and $AP = APH + APV$ for $D = 100, 200$ and 400 , and $GL = GLH$, $GP = GPV$, $AL = ALH$, and $AP = APV$ for $D = 600$. NUM increases for $D = 400$ and 600 in GL and GP , and on the other hand, NUM stays around 1.0 even for longer D in AL and AP . This means that the single dragging operation achieves to reach longer distance targets by the proposed technique. In addition, we observe different behaviors of GL and GP at $D = 400$ and 600 in Fig. 9. This result implies that the movable range of the tip of an index finger would be slightly different; the range of the portrait orientation grip style would be larger than that of the landscape orientation. As a result, the mean NUM of the generic technique increased, and on the other hand, that of the proposed technique was not deteriorated even for longer D .

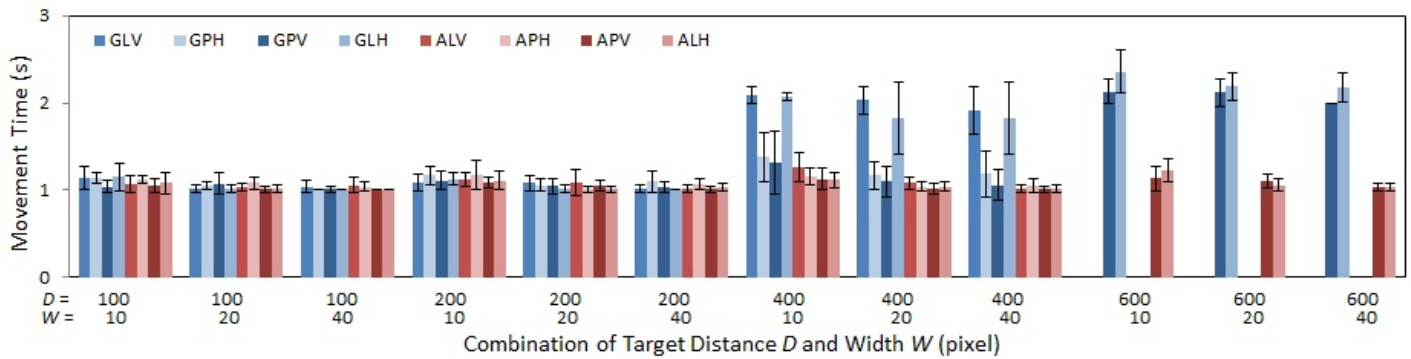


Fig. 8: The number of dragging by W and D for eight models

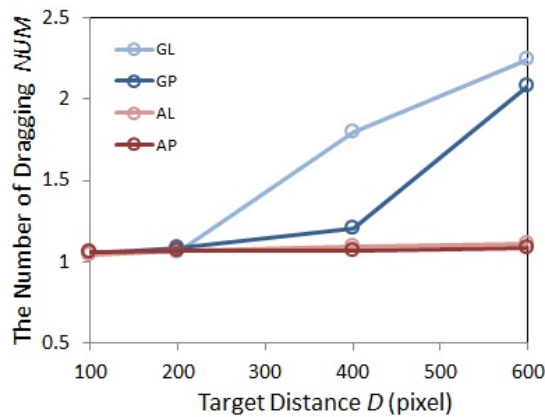


Fig. 9: NUM by D for four models ($TEC \times HLD$)

VI. DISCUSSION

Results in Fig. 4 show that plots of GLV, GPH, ALV and APH are aligned on a straight line and yielded r^2 values of above 0.95. Although step-wise plots caused by an increase of NUM at $D = 400$ and 600 are observed in GLH and GPV, we find better alignments of r^2 values of 0.945 in ALH and 0.978 in APV. The performance improvements of the proposed technique are clearly demonstrated in Fig. 7 and Fig. 9. Actually, the approach as addressed by the proposed technique is effective to utilize the limited range inputs. Results in Fig. 6 indicate that the longitudinal movement would be easier than the transverse movement of the tip of an index finger. While, results in Fig. 9 imply that the movable range of the tip of an index finger in the portrait orientation grip style would be slightly larger than that of the landscape orientation. These results suggest that APV would be the most appropriate design for single-handed user interactions using rear touch.

VII. CONCLUSION AND FUTURE WORK

We have conducted a basic study of single-handed cursor control technique optimized for rear touch operation, and showed the performance improvements in target selection tasks by the proposed technique in comparison with the generic technique. Especially, we demonstrated that the single dragging operation achieved to reach longer distance targets by the proposed technique. Since the movable area of the tip of an

index finger on a rear touch panel is limited, the technique, such that small movement of dragging accomplishes long distance cursor movement, is effective. In addition, evaluation in relation to the way of holding the handheld device and the direction of dragging operation suggested the existence of advantageous combinations for usability. These results will be the beneficial knowledge for better design of single-handed user interactions using rear touch.

Considering a practical use of handheld devices, not only cursor navigation but also command execution, such as scroll or zoom, is required. To find the ways to control commands, which have an affinity for the combined use with rear-touch-based cursor navigation system, is an important future work. Touch-gesture-based approach [20], [21], [22] with regarding touch-pressure [23], [24] has potential for solutions. The influence of the size of display and the weight of device on performance is also of significant interest. And the effects of haptic feedback are another topic of future consideration for improvements of usability.

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Traffic Sign Detection and Recognition using Features Combination and Random Forests

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Abstract—In this paper, we present a computer vision based system for fast robust Traffic Sign Detection and Recognition (TSDR), consisting of three steps. The first step consists on image enhancement and thresholding using the three components of the Hue Saturation and Value (HSV) space. Then we refer to distance to border feature and Random Forests classifier to detect circular, triangular and rectangular shapes on the segmented images. The last step consists on identifying the information included in the detected traffic signs. We compare four features descriptors which include Histogram of Oriented Gradients (HOG), Gabor, Local Binary Pattern (LBP), and Local Self-Similarity (LSS). We also compare their different combinations. For the classifiers we have carried out a comparison between Random Forests and Support Vector Machines (SVMs). The best results are given by the combination HOG with LSS together with the Random Forest classifier. The proposed method has been tested on the Swedish Traffic Signs Data set and gives satisfactory results.

Keywords—Traffic Sign Recognition (TSR); thresholding; Hue Saturation and Value (HSV); Histogram of Oriented Gradients (HOG); Gabor; Local Binary Pattern (LBP); Local Self-Similarity (LSS); Random forests

I. INTRODUCTION

Advanced driver assistance systems (ADAS) are one of the fastest-growing fields in automotive electronics. ADAS technology can be based upon vision systems [1], active sensors technology [2], car data networks [3], etc. These devices can be utilized to extract various kinds of data from the driving environments. One of the most important difficulties that ADAS face is the understanding of the environment and guidance of the vehicles in real outdoor scenes. Traffic signs are installed to guide, warn, and regulate traffic. They supply information to help drivers. In the real world, drivers may not always notice road signs. At night or in bad weather, traffic signs are harder to recognize correctly and the drivers are easily affected by headlights of oncoming vehicles. These situations may lead to traffic accidents and serious injuries. A vision-based road sign detection and recognition system is thus desirable to catch the attention of a driver to avoid traffic

hazards. These systems are important tasks not only for ADAS, but also for other real-world applications including urban scene understanding, automated driving, or even sign monitoring for maintenance. It can enhance safety by informing the drivers about the current state of traffic signs on the road and giving valuable information about precaution. However, many factors make the road sign recognition problem difficult (see Fig. 1) such as lighting condition changes, occlusion of signs due to obstacles, deformation of signs, motion blur in video images, etc.



Fig. 1: Examples for difficulties facing the traffic sign recognition (TSR) task.

A traffic sign recognition algorithm usually consists of two modules: the detection module and the classification module. The detection module receives images from the camera and finds out all the regions in the images that may contain traffic signs; then the classification module determines the category of traffic sign in each region. The information provided by the traffic signs is encoded in their visual properties: color, shape, and pictogram. Therefore, the detection and the recognition modules are based on color and shape cues of traffic signs. In this paper, we describe a fast system for vision based traffic sign detection and recognition.

The rest of the paper is organized as follows. Section 2 presents an overview of past work on traffic sign detection and recognition. Section 3 details the proposed approach to traffic sign detection and recognition. Experimental results are illustrated in Section 4. Section 5 concludes the paper.

II. OVERVIEW

Many different approaches to traffic sign recognition have been proposed and it is difficult to compare between those approaches since they are based on different data. Moreover, some articles concentrate on subclasses of signs, for example on speed limit signs and digit recognition. This section gives an overview of the techniques used in the TSR and previous works using these techniques. According to the two basic tasks in traffic sign recognition, we simply divide the overview into two categories: traffic sign detection and classification.

A. Traffic Sign Detection

The purpose of traffic sign detection is to find the locations and sizes of traffic signs in natural scene images. The well-defined colors and shapes are two main cues for traffic sign detection. Thus, we can divide the detection methods into two categories: color-based and shape-based. Color-based methods are usually fast and invariant to translation, rotation and scaling. As color can be easily affected by the lighting condition, the main difficulty of color-based methods is how to be invariant to different lighting conditions. These methods tend to follow a common scheme: the image is transformed into a color space and then thresholded. Some authors perform this thresholding directly in RGB (Red Green Blue) space, even if it is very sensitive to illumination changes. To overcome this, simple formulas relating red, green and blue components are employed. For example, Escalera et al. in [4] used different relations between the R, G and B components to segment the desired color. In [5] the difference between R and G, and the difference between R and B channels are employed to form two stable features in traffic sign detection. Ruta et al. in [6], used the color enhancement to extract red, blue and yellow blobs. This transform emphasizes the pixels where the given color channel is dominant over the other two in the RGB color space. In addition to RGB space, other color spaces such as YUV and HSI are also used. For example, The YUV system is considered in [7] to detect blue rectangular signs. In [8] a segmentation method in both La-b and HSI color spaces is used to extract candidate blobs for chromatic signs. At the same time, white signs are detected with the help of an achromatic decomposition. Then a post-processing step is performed in order to discard non-interest regions, to connect fragmented signs, and to separate signs located at the same post.

In the other hand, shape-based methods employ either Haar-like features in frameworks inspired by the popular Viola-Jones detector or the orientation and intensity of image gradients in frameworks inspired by the Generalized Hough Transform. The first sub-category comprises the works by Bahlmann et al. [9] and by Brkic et al. [10], whereas in the second we find the Regular Polygon Detector [11], the Radial Symmetry Detector [12], the Vertex Bisector Transform [13], the Bilateral Chinese Transform and, alike, the two schemes of Single Target Voting for triangles and circles proposed by Houben [14]. Many recent approaches use gradient orientation information in the detection phase, for example, in [11], Edge Orientation Histograms are computed over shape-specific sub-regions of the image. Gao et al. [15] classify the candidate traffic signs by comparing their local edge orientations at arbitrary fixation points with those of the templates. In [16],

the Regions of Interest (ROI) obtained from color-based segmentation are classified using the HOG feature. To integrate color information in the HOG descriptor, Creusen et al. [17] concatenate the HOG descriptors calculated on each of the color channels. The advantages of this feature are its scale-invariance, the local contrast normalization, the coarse spatial sampling and the fine weighted orientation binning.

B. Traffic Sign Recognition

The purpose of traffic sign recognition is to classify the detected traffic signs to their specific sub-classes. Regarding the recognition problem, it is common to use some features with machine learning algorithms. Maldonado et al. in [18] utilized different one-vs-all SVMs with Gaussian kernel for each color and shape classification to recognize signs. In [19] SVMs are used with HOG features to carry out classification on candidate regions provided by the interest region detectors. It withstand great appearance variations thanks to the robustness of local features, which typically occur in outdoor data, especially dramatic illumination and scale changes. Zaklouta [20] uses different sized HOG features, and adopts random forest based classification to achieve high detection accuracy. Tang [21] proposes an efficient method of traffic sign recognition using complementary features to reduce the computation complexity of traffic sign detection, and then uses the SVM to implement the traffic sign classification. The complementary features used in Tang [21] include HOG [22] and LBP [23]. Convolutional Neural Network (CNN) is another method used for traffic sign classification. It is proved in [24] that the performance of CNN on traffic sign classification outperforms the human performance. In [25], a CNN together with a Multi-Layer Perception (MLP), which is trained on HOG features, was used. In [26], a CNN with multi-scale features by using layer-skipping connection is presented. In [1], the authors suggest a hinge loss stochastic gradient descent method to train convolutional neural networks. The method yields to high accuracy rates. However, a high computing cost is paid to train the data when using CNNs.

In general, the quality of the results obtained by any study on TSR varies from one research group to another. It is very difficult to decide which approach gives better overall results, mainly due to the lack of a standard database of road images. It is not possible to know, for example, how well the systems respond to changes in illumination of the images since in the different studies it is usually not specified whether images with low illumination have been used in the experiments. Another disadvantage of the lack of a standardised database of road images is that some studies are based on a small set of images since the compilation of a set of road scene images is a very time-consuming task. The problem with working with such small data sets is that it is difficult to evaluate the reliability of the results.

III. PROPOSED METHOD

The proposed system consists of three stages: segmentation, shape detection and recognition. In the first stage, we aim to segment the images to extract ROIs. In the second one, we detect the desired shapes from the ROIs. In the last stage, we recognize the information included in the detected traffic signs. Fig. 2 illustrates the algorithm scheme of the proposed

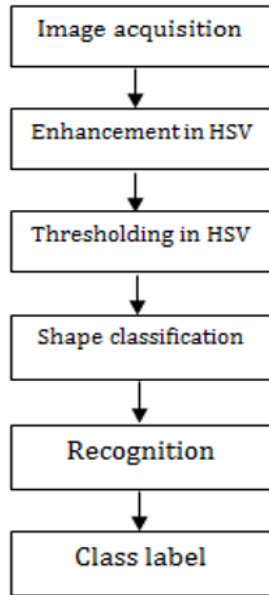


Fig. 2: Algorithm scheme.

method. In this section, we detail each step of the proposed approach.

A. Segmentation

Color segmentation algorithms are influenced by weather condition, daytime, shadows, orientation of objects in relation to the sun and many other parameters. These parameters change frequently in dense urban area scenes. In addition, there are many other objects in the street of the same color as traffic signs (red and blue). Therefore, the color information is only used to generate ROIs without performing classification.

To overcome the difficulties related to illumination changes and possible deterioration of the signs, the HSV color space is used in our system. We implement both enhancement and thresholding techniques. First, we enhance the input image in HSV color space. Then, we segment the image using fixed thresholds. These thresholds were empirically deduced using traffic sign images. The resulting binary image is then post-processed to discard insignificant ROIs and to reduce the number of ROIs to be provided to shape classification stage.

1) *Enhancement*: Approved by many experiments, HSV color space is a good choice for color image enhancement. There is only a weak correlation between HSV components, which indicates that a modification to one component will only slightly change another. Unfortunately, in some situation, the slightly change in HSV will result in great color distortion. In this paper, the hue and saturation component are kept intact and only value component of the input image is subjected for enhancement. This enhancement is done according to two steps: Luminance enhancement and Contrast enhancement.

First, The luminance enhancement is applied to the value component using the formula provided in [27]. Suppose that $V_1(x, y)$ denote the normalized V channel in HSV space and $V_2(x, y)$ be the transferred value by applying nonlinear transfer



Fig. 3: Enhancement results.

function defined below.

$$V_2 = \frac{V_1^{0.75z+0.25} + 0.4(1-z)(1-V_1) + V_1^{(2-z)}}{2} \quad (1)$$

where, z is the image dependent parameter and is defined as follows

$$z = \begin{cases} 0, & \text{for } L \leq 50 \\ \frac{L-50}{100}, & \text{for } 50 < L \leq 150 \\ 1, & \text{for } L > 150 \end{cases} \quad (2)$$

where, L is the value (V) level corresponding to the cumulative probability distribution function (CDF) of 0.1. In equation 2 the parameter z defines the shape of the transfer function or the amount of luminance enhancement for each pixel value.

The second step is the contrast enhancement. In this process, the Gaussian convolution using Gaussian function $G(x, y)$ is carried out on the original V channel of the input image in HSV space. The convolution can be expressed as:

$$V_3(x, y) = \sum_{m=0}^{M-1} \sum_{n=0}^{N-1} V(m, n)G(m+x, n+y) \quad (3)$$

V_3 in equation 3 denotes the convolution result, which contains the luminance information from the surrounding pixels. The amount of contrast enhancement of the centre pixel is now determined by comparing centre pixel value with the Gaussian convolution result. This process is described in the following equation:

$$V_4(x, y) = 255V_2(x, y)^{E(x, y)} \quad (4)$$

where

$$E(x, y) = \left[\frac{V_3(x, y)}{V(x, y)} \right]^g \quad (5)$$

where, g is the parameter determined from the original value component image in HSV space for tuning the contrast enhancement process. This parameter g is determined using following equation:

$$g = \begin{cases} 1.75, & \text{for } \sigma \leq 2 \\ \frac{27-2\sigma}{13}, & \text{for } 2 < \sigma < 10 \\ 0.5, & \text{for } \sigma \geq 10 \end{cases} \quad (6)$$

where, σ denotes the standard deviation of the individual block of the original value component image. The standard deviation is determined globally, as it was done in [27]. Fig. 3 shows an example of image before and after enhancement process.

TABLE I: Thresholds used for road sign detection.

	Red	Bleu
Hue	$0 < H < 12$ Or $300 < H < 360$	$190 < H < 260$
Saturation	$25 < S < 250$	$70 < S < 250$
Value	$30 < V < 200$	$56 < V < 128$



Fig. 4: Segmentation results.

2) *Thresholding*: After the enhancement process, we refer to thresholding to segment the image into ROIs. Each image element is classified according to its hue, saturation, and value. A pixel color is considered as red or blue using the threshold values shown in Table I. The hue obtained H is within the interval $[0, 360]$, the saturation S and intensity I is within $[0, 255]$. We further refer to the achromatic decomposition used in [18] to segment white color. This achromatic decomposition is defined as:

$$f(R, G, B) = \frac{|R - G| + |G - B| + |B - R|}{3D} \quad (7)$$

The R, G and B represent the brightness of respective color. D is degree of extracting an achromatic and it is empirically set to $D = 20$ in [18]. An achromatic color is represented by $f(R, G, B)$ of less than 1, and an $f(R, G, B)$ of greater than 1 represents chromatic colors.

After the segmentation stage, we obtain a binary image with the pixels of interest being white and others black (see Fig. 4(b)). Then, according to the size and the aspect ratio of the blobs, we eliminated noise and blobs considered as non-interest. The limits for both criteria, i.e., size and aspect ratio, were empirically derived based on road images (see Fig. 4(c)).

B. Shape Classification

In this stage, we classify the blobs that were obtained from the segmentation stage according to their shape. We only consider triangular, circular and rectangular shapes. Thus, Distance to Borders (DtBs) [18] are used as feature vectors for the inputs of a random forest classifier. DtBs are the distances from the external edge of the blob to its bounding box. These features are widely used to classify shapes, and show its performance in many traffic sign recognition works. Fig. 5 shows these distances for a triangular shape. After computing these features, a random forest classifier is used to classify the ROIs into appropriate shapes. A Random Forest is an ensemble of classification trees, where each tree contributes with a single vote for the assignment of the most frequent class to the input data. It adds an additional layer of randomness to bagging. In addition to constructing each tree using a different bootstrap sample of the data, random forests change

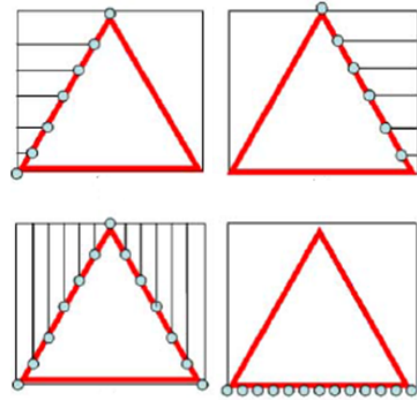


Fig. 5: DtBs for a triangular shape.

how the classification or regression trees are constructed. In standard trees, each node is split using the best split among all variables. In a random forest, each node is split using the best among a subset of predictors randomly chosen at that node. This somewhat counter intuitive strategy turns out to perform very well compared to many other classifiers, and is robust against overfitting [28]. Random Forests have received increasing interest because they can be more accurate and robust to noise than single classifiers [20] [16].

The proposed method is invariant to translation, scale and rotation. First, it is invariant to translation because it does not matter where the candidate blob is. Second, the method is invariant to scale due to the normalization of the DtB vectors to the bounding-box dimensions. Finally, the detection process is invariant to rotation because the most external pixels of each blob are detected to determine the original orientation, and after this, all blobs are oriented in a reference position. In conclusion, samples of DtB vectors show a similar evolution for each geometric shape.

C. Recognition

Once the candidate blobs are classified into a shape class, the recognition process is initiated. The main objectives of this stage to be based on a method with a high accuracy but at the same time, the memory and the complexity of the algorithm used have to be minimized.

In this work, we compare the Random Forests classifier, to the state-of-the-art SVM classifier. As we will see in section 4, random forests performance is better than SVMs in both the accuracy rate and the execution time.

For the feature extraction, inspired by the existing ones, we try to introduce new ones using different combinations. HOG, Gabor filters, LBP, and LSS are used in this work. The performance and the execution time of these features as well as the classifiers ones are shown in section 4.

1) *Features extraction*: We used in this work four kinds of features namely, HOG, Gabor, LBP, and LSS. The first feature used is HOG feature. It was proposed by Navneet Dalal and Bill Triggs [22] for pedestrian detection. The basic idea of HOG features is that the local object appearance and shape can often be characterized rather well by the distribution of the local intensity gradients or

edge directions, even without precise knowledge of the corresponding gradient or edge positions. The method is very simple and fast so the histogram can be calculated quickly.

The second one is Gabor feature. Gabor filters have been applied to many signal processing and pattern recognition problems. They are able to explore the local spectrum characteristics of image. A 2D Gabor filter is a band-pass spatial filter with selectivity to both orientation and spatial frequency [29].

The third one is LBP. It was proposed by T. Ojala [23], and it is a popular texture descriptor. The concept of LBP feature vector, in its simplest form, is similar to the HOGs. The window is divided into cells. For each pixel in a cell, we compare the center pixels value to each of its 8 neighbours, and the pixels value is set to 1 if its value is greater than the center pixels value, or set to 0 otherwise. Then compute the histogram, over the cell, of the frequency of each 'number' occurring, and normalize it to obtain histograms of all cells. This gives the features vector for the window.

The last one is LSS feature. Generally in LSS, The selected image is partitioned into smaller cells which, conveniently compared with a patch located at the image center. The resulting distance surface is normalized and projected into the space intervals partitioned by the number of angle intervals and radial intervals. The maximum value in an interval space would be considered as the value of the feature.

2) *Classifiers*: Two classifiers were used in this work: random forests and SVMs. The results of the comparison are presented in section 4.

As mentioned in III-B, Random Forests have received increasing interest because they can be more accurate and robust to noise than single classifiers. Another advantage of Random Forests is their ease of use in the sense that they have only two parameters (the number of variables in the random subset at each node and the number of trees in the forest), and is usually not very sensitive to their values. The main idea of random forests consists of an arbitrary number of simple trees, where the final predicted class for a test object is the mode of the predictions of all individual trees.

In the other hand, SVMs are used to extend our study of classifiers for TSR. The algorithm attempts to separate the positive examples of negative examples. The basic concept of SVM is to transform the input vectors to a higher dimensional space by a nonlinear transform, and then an optimal hyperplane that separates the data, can be found. This hyperplane should have the best generalization capability. In many cases, the data cannot be separated by a linear function. The use of a kernel function becomes essential in these cases. SVM is designed to solve a binary classification problem. However, for a road sign inventory problem, which is a multiple classification problem, classification is accomplished through combinations of binary classification problems. There are two ways to do that: one-vs.-one or one-vs.-all.

IV. EXPERIMENTAL RESULTS

This section presents the results obtained by the proposed approach. Evaluation of the classifiers as well as the features presented in III-C1 are presented to justify the choice of the proposed system. All the tests were performed on the public STS data set [30] using a 2.7 GHz Intel i5 processor.

A. Data Set

We implement our method on the Swedish Traffic Sign data set (STSD). It is a public data set which contains sequences videos and includes more than 20 000 images in which 20% of the images are labeled. It contains 3488 traffic signs. The images in STSD are obtained from highways and cities record from more than 350 km of Swedish roads Fig. 6.



Fig. 6: Examples of STS Data Set images.

B. Traffic Sign Detection

The evaluation of the detection stage is performed based on precision-recall curve, where the recall and precision values are computed as follows:

$$recall = \frac{\text{Number of correctly detected signs}}{\text{Number of true signs}} \times 100 \quad (8)$$

$$precision = \frac{\text{Number of correctly detected signs}}{\text{Number of detected signs}} \times 100 \quad (9)$$

The precision-recall curves of the proposed method when applied to STS data set are depicted in Fig. 7. The best trade-off between the recall and precision values as well as the Area Under Curve (AUC) of the detection module are listed in Table II. It can be seen that the method yields the best results with recall of 93.41% at a precision of 95.12%. The AUC of the precision-recall curve is 94.50%.

Fig. 8(a) shows an example among images used to test the proposed detection approach. The corresponding segmentation

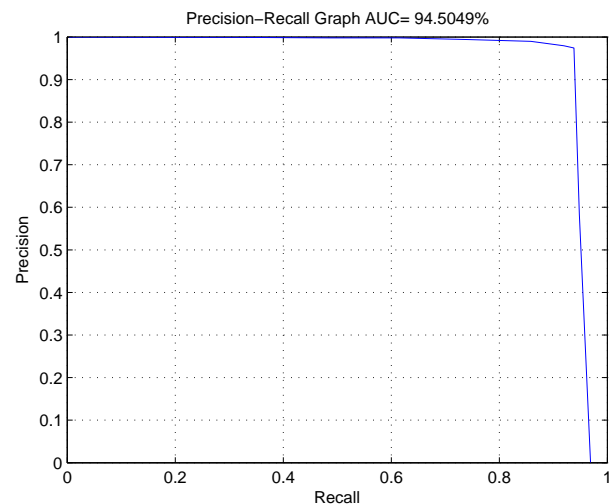


Fig. 7: Precision-recall curve of the proposed detection approach when applied STS data sets.

TABLE II: The best trade-off between the recall and precision values as well as the AUC obtained by the detection method on STS data set in %.

	Recall	Precision	AUC
On the STS Data set	93.41	95.12	94.50

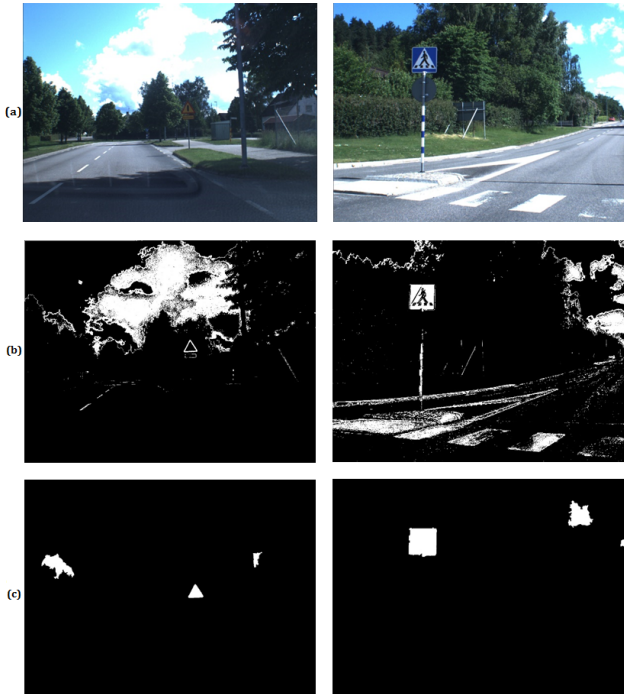


Fig. 8: Example of test images (a) and (b), (c) segmentation phase's results.



Fig. 9: Final detection results by the proposed detection method.

results with and without using size and aspect ratio constraints are illustrated in Fig. 8(b). Referred to these figures, some regions are discarded as non-interest objects according to their size and aspect ratio. Therefore, the detection process can be reduced as the number of ROIs is reduced. The segmentation method succeeds to detect the road sign present in Fig. 8(a) among the extracted ROIs in Fig. 8(b). However, some ROIs have been detected even they do not represent road signs. The shape classification method has been applied to the ROIs in Fig. 8(c). The DtBs of the extracted ROIs have been computed and feed to random forest classifier. Fig. 9 shows the final detection results by the proposed detection method. Red bounding box represents detected region of traffic sign.

C. Traffic Sign Recognition

To evaluate the recognition stage, a comparison between features and classifiers used in the system is performed. To obtain optimal design parameters of each descriptor, we run some cross-validation experiments on the training dataset: divide the training images into a basic training set and a validation set. By training classifier on the basic training set and evaluate on the validation set, we selected the setting of maximum validation accuracy and a setting of lower-dimensionality. After that, the classifier is re-trained on the whole training set with selected feature extraction settings.

To compute the HOG feature vector, we normalize the window detected in the previous stage to 40×40 , and the normalized image is divided into 8×8 overlapping blocks, which gives us a total number of 49 blocks. Each one of these blocks is divided to 2×2 cells, and each cell contains a 5×5 pixels. In each cell, we obtain a gradient histogram of 9 bins. For the Gabor feature, we used tow scales and eight orientations. The window is partitioned into 16×16 blocks and sampling interval varies according to the block sizes. For the LBP feature: we employ a basic LBP descriptor to compute the LBP features. The normalized window is partitioned into 6×6 non-overlapping blocks. Using the uniform patterns method, we extract 59 features per block, and finally form the LBP feature vector. The last feature is LSS. It has four primary parameters: the size of image, the radius of window, the interval radius of image patches and angle interval. These parameters are closely associated with each other. In our implementation, we used 3×3 patches, correlated against a surrounding window with radius equal to 10. Our log-polar coordinates was partitioned into 80 bins (20 angles and 4 radial intervals).

After calculating the four different features individually, we concatenate them to form new features. In Table III, different compound features are listed to compare their performance on the STS data set. As we can see from Table III among the four single feature descriptors, the HOG feature has a Correct Classification Rate (CCR) of 95.38%, higher than the Gabor, LBP, and LSS features. According to the results, combining two different features can improve classification accuracy evidently. Particularly, the combination of HOG and LSS features gives a CCR of 96.13%, significantly better than the best single feature HOG or LSS. Each combination of two features outperforms its constituent single features. This confirms that the different features are complementary.

Table III gives also a comparison between the state-of-the-art SVMs with radial basis function (RBF) kernel, $C = 7$ and $G = 0.09$ and Random Forest with 600 trees and 100 variables, in the terms of CCR and running time. It is obvious from the table that the Random Forest classifier provides accurate results with less running time when compared to the SVM classifier. Thus, we have adopted in the proposed recognition method the Random Forest classifier together with the HOG+LSS features.

Figs. 10, 11 and 12 illustrate examples of recognition results when the proposed approach is applied to images of various traffic environments. In Fig. 10, the traffic signs contained in the images have been successfully detected and

TABLE III: The CCR and the average running time of the classifiers and features used in this work.

Feature	Random Forests		SVMs	
	CCR(%)	Execution Time(ms)	CRR(%)	Execution Time(ms)
HOG	95.38	20.82	94.52	37.36
Gabor	94.89	31.21	94.23	41.67
LBP	94.63	12.30	94.02	18.45
LSS	94.24	10.72	3.84	13.38
HOG+Gabor	95.59	38.51	95.15	58.96
HOG+LBP	95.94	25.23	95.46	43.48
HOG+LSS	96.13	22.18	95.68	42.81
Gabor+LBP	95.39	33.81	94.79	49.34
Gabor+LSS	95.42	33.47	94.47	49.52
LBP+LSS	95.29	13.90	94.44	20.91



Fig. 10: Detection and recognition results.



Fig. 11: Examples of recognition with misdetection.



Fig. 12: Examples of recognition with confused classification.

recognized. In Fig. 11, the system was not able to detect traffic signs. Consequently, the ROIs corresponding to the signs were not feed to the recognition stage. In Fig. 12, the traffic signs contained in the images have been successfully detected. However, the system could not recognize them due to the motion blur in the signs.

V. CONCLUSION

In this paper, a fast system for Traffic Sign Detection and Recognition was described. In the first stage, we refer to color segmentation to reduce the search space. We used an enhancement then a thresholding on the HSV color space. In the second stage, the circular, rectangular and triangular signs are

detected using the Distance to Border feature and a Random Forest classifier. The detected candidates are identified using The Random forests classifier with a combination of HOG and LSS features. The system achieves correct classification rate of over 96% at a processing rate of 8–10 frames/s. In the future work, we can use adaptive thresholds to overcome the color segmentation problems. Temporal information could also be integrated to track the detected traffic signs and reinforce the decision making process. This would also allow us to restrict the search space in the current image considering previous detections information, which can accelerate the candidate detection.

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Risk Propagation Analysis and Visualization using Percolation Theory

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Abstract—This article presents a percolation-based approach for the analysis of risk propagation, using malware spreading as a showcase example. Conventional risk management is often driven by human (subjective) assessment of how one risk influences the other, respectively, how security incidents can affect subsequent problems in interconnected (sub)systems of an infrastructure. Using percolation theory, a well-established methodology in the fields of epidemiology and disease spreading, a simple simulation-based method is described to assess risk propagation systematically. This simulation is formally analyzed using percolation theory, to obtain closed form criteria that help predicting a pandemic incident propagation (or a propagation with average-case bounded implications). The method is designed as a security decision support tool, e.g., to be used in security operation centers. For that matter, a flexible visualization technique is devised, which is naturally induced by the percolation model and the simulation algorithm that derives from it. The main output of the model is a graphical visualization of the infrastructure (physical or logical topology). This representation uses color codes to indicate the likelihood of problems to arise from a security incident that initially occurs at a given point in the system. Large likelihoods for problems thus indicate “hotspots”, where additional action should be taken.

Keywords—security operation center; malware infection; percolation; BYOD; risk propagation; visualization

I. INTRODUCTION

Risk is a notoriously fuzzy term that describes the possibility of suffering damage, based on expected occurrences of certain incidents. Security risks add to the complexity of (general) risks an element of rationality, as security commonly assumes the existence of some hostile actor in the system. This takes a security risk analysis beyond the scope of pure probabilistic modeling, since events occur no longer at random, but may partially follow unknown rationales and evolve over time. Advanced persistent threats are one prominent example of this.

A typical security assessment is composed from many reports, traffic light scales or other rating systems or visualizations. All of these ideas have a common ground, and aggregate information for the sake of a simple or complex visualization. Often, the effect is either information flooding or information loss for the decision maker. Independently of which risk assessment method or model is used, the results should thus not be over-condensed into a single indicator (like

a “security traffic light” that turns red in case of trouble, yellow to indicate problems ahead, and green if everything is okay), and support a decision maker by providing an adjustable level of granularity. Ideally, the representation allows to “zoom-in” or “zoom-out” of the risk picture to get a solid understanding of the situation, and to derive the proper actions thereof. This visualization aspect is discussed further in section III.

In the following, let us consider an infrastructure as a system of interconnected components, which in a simplified view can be described as a graph (nodes being components, edges being their connections). Most practical systems are heterogeneous, in the sense of interconnecting many subsystems of quite different nature. For example, utility networks are usually a compound construct of a physical utility supply system (e.g., water, electricity, etc.), which is controlled by an upper layer for supervisory control and data acquisition (SCADA) systems. Likewise, complex enterprise infrastructures may be clustered into local area networks (LANs) that are themselves interconnected by a wide area network (WAN) layer, which may or may not be under the control of the enterprise (e.g., the physical communication services could be outsourced to some external party).

Incident propagation in such a heterogeneous environment is generally difficult to analyze, since an incident occurring at one point may have (in)direct implications that depend on the system dynamics, but also on how the problem’s origin node is “connected” to other parts of the system. For example, if a malware-infected sensor delivers incorrect measurements, the resulting incorrect information may cause subsequent problems in other parts of the system. However, the malware itself may not be able to spread over the same link, if the sensor connection is only for signalling and not for data transmission. So, a problem may propagate differently over different links, and the physical data item that causes the problems may change along its way through the system.

A further example of exactly this kind of information change is the distribution of information between agents in social and market systems (cf. [1], [2]). This is yet another important aspect of risk management, especially when it comes to a company’s reputation and public image. Spreading rumours or negative information can have serious consequences for an enterprise, and risk management must consider this. Although these parts of risk management relate to psychology and

sociology rather than technology, the techniques and results presented in this work will (after obvious adaptations) apply to such matters equally well.

Basically, incident propagation can be treated like an *epidemic outbreak*, where the infection depends on how the individuals are connected to each other. In the technical context here, possible connections could be for data transmissions, while others may be only signalling, but some may also be social, when people exchange information by email or orally. Problems arising at one place may then propagate through various kinds of connection and infect large parts of the relevant infrastructure; and it is the risk management's duty to assess the potential impacts thereof. While the relevant standards (cf. [3]–[5] among others) explicitly prescribe to identify and analyze the system characteristics for an informed risk assessment, none of these standards provides an explicit method to do this. This article shall help in this regard.

In fact, while epidemics spreading has received much attention (see below), its use to analyze risk management is so far still limited. In practice, most risk assessments are heuristic and based on expertise and experience, which makes them inevitably subjective. This work is intended to aid risk managers by providing simulation models and tools to easily analyse and assess incident effects.

Before coming to the details, section I-A and section I-B both discuss related work on epidemic spreading, in order to illustrate the improvements/benefits offered by the model that is described in section II.

A. Epidemic Models – A Critical Look

Roughly speaking, the vast amount of epidemic models available in the literature (see, e.g., [6]–[14]) can be divided into deterministic and stochastic approaches, which are discussed separately below.

A popular representative of the deterministic class is the SIR model (see [15] for example), whose name encodes three functions of time t , which are $S(t)$ = number susceptible, $I(t)$ = number infectious, and $R(t)$ = number recovered. These functions are then described by differential equations, whose numeric or algebraic solution can be used to predict and control the outbreak of an infection. Depending on the assumptions made (e.g., a uniform and constant rate of infection, recovery with immunity, etc.), various other models can be obtained. The SIS model is one such particular instance, and assumes recovery of individuals without immunity. Such individuals can model technical components on which an infection cannot ultimately be banned by applying a patch. More importantly, any kind of immunity by security boundaries (e.g., firewall, physical protection, etc.) can be circumvented if users are allowed to connect their own devices that may be infected with malware. In light of such BYOD (bring your own device) events, which are inherently random, a deterministic epidemic outbreak regime seems too restrictive. In turn, the stochastic element induced by BYOD may also invalidate some assumptions underneath deterministic models, such as constant rates of infection.

Consequently, stochastic models of epidemic spreading appear to be an attractive alternative. Nevertheless, they also ship

with difficulties: essentially, statistical models are built from (massive lots of) data (observations), which may be difficult to obtain or even be unavailable at all. Especially in a security context, companies are quite reluctant in releasing information about incidents in order to not endanger their reputation. Thus, a statistical approach should work with as little data as is there, and should avoid further loss of information by aggregation (as is common in risk assessment, say by taking the overall risk as the maximum risk across all system components).

B. Malware Infections and Percolation Theory

The stochastic model of choice in this work uses percolation theory [6], [16]–[19] to assess the cascading effects of problems hitting some defined point in the system. Based on simulations on which parts of the system (may) become infected, the likelihood for an infection of a particular node can then be expressed by color-encoding (heat map) of the infrastructure graph. This creates an easily understandable overview for a decision maker, which visualizes the current situation and helps to decide on proper actions. Combining the so-obtained incident indications (likelihoods) with standard risk aggregation (say, by the aforementioned maximum principle [20]) helps to simplify or detail the infrastructure picture, depending on what is relevant for the decision making. This is the twofold benefit over most related work in this area, as neither an oversimplification of results nor an over-complication of the underlying model can render the risk assessment useless.

Starting from a simple and intuitive simulation approach to malware spreading (e.g., the aforementioned BYOD scenario), percolations are a natural way of describing such simulations formally. Percolation theory then delivers even closed form criteria for whether or not an epidemic can (will) grow into a pandemic. The particular the closed-form criterion that is discussed in section II-C generalizes the work of [9], [21], who assume a “homogenous outbreak”, in the sense that an infection is equiprobable for any pair of entities in contact, and in particular happens irrespectively of the nature of the two entities. This is similar to the assumption of a constant infection rate as found in many deterministic models. As shown below, this restriction can be dropped by using different infection rates for different kinds of connections (e.g., email contact, wireless layer 2 connection, etc.).

BYOD provides a particularly illustrative example: while a virus can enter the system on a USB stick (“BYOD connection”), it may spread by itself within the locally connected network (email connection), and may later penetrate another physical separation by another BYOD incident. One prominent such case was reported for the Iranian nuclear power plants, which got infected by the Stuxnet worm [22] via BYOD. Although BYOD and protection against it has received considerable attention in the literature [23]–[26], an epidemics-like treatment of malware infections (e.g., caused by BYOD) for risk management is not available so far. Existing proposals in this area are usually restricted to specific topologies [6], [18], [19] or are focused on factors that are specific for human disease spreading [10], and thus do not directly apply to the purely technical network infection scenario.

II. THE SIMULATION AND PERCOLATION MODEL

Let the network infrastructure be modeled as a directed graph $G = (V, E)$. The set V contains all nodes, with their interconnections being edges in $E \subseteq V \times V$. Note that w.l.o.g., G can be assumed as directed, since bidirectional links are easily emulated by adding arrows in both directions. Assume that all edges in E fall into different non-overlapping “classes”, where each class has distinct characteristics in how a problem propagates over the respective edges. Examples could be email-communication (class 1), direct network links (class 2), BYOD links (class 3), and so on. Formally, E is partitioned into m subsets of edge classes, each of which has different properties related to incident propagation, and one edge is assigned to exactly one edge class. Specifically, let each edge class k be associated with a likelihood p_k to transport the problem. The way how to define and obtain p_k is discussed later in section II-B.

Note that a distinction between different connection types is indeed common practice in critical infrastructure risk analysis [27]–[32], so it is reasonable to assume a concrete such partitioning available (e.g., based on any of the cited precursor references). The issue is revisited in section III, where hints on how to identify the edge classes are given.

A. Simulation of Infections

Let $\lambda > 0$ denote a general infection rate parameter, which equals the average number of events per time unit. For example, if the daily email traffic in a network is known to be (on average) N_1 emails per hour, then λ is obtained by scaling N_1 down (or up) to the time unit used for the simulation. Likewise, if the traffic on links of edge class 2 is N_2 packets per time unit, then $\lambda_2 = N_2$, etc. Let one such rate parameter per edge class be given, calling them $\lambda_1, \dots, \lambda_m$. That is, a class-specific (and thus not constant) infection rate is assumed. Estimates for each rate parameter can be obtained from empirical analysis of the traffic in the infrastructure, e.g., by using packet sniffing, counting emails, asking people how often they plug in USB devices on average, and so forth. For the i -th edge class, the number of infection attempts (once a problem occurred at some point), is *Poisson*(λ_i)-distributed. The total number of infection attempts is therefore also Poisson distributed with rate parameter $\lambda = \lambda_1 + \lambda_2 + \dots + \lambda_m$, and the pause between any two events is exponentially distributed with parameter $1/\lambda$.

The outbreak of an infection, i.e., the risk propagation, will be simulated on the given directed network graph $G = (V, E)$, where each edge $i \rightarrow j$ is assigned a likelihood p_k specific for each edge class and describing the probability of the problem to jump from node i to node j in the infrastructure. The simulation of an outbreak then boils down to graph coloring at random. That is, let all nodes in G be colored green initially (indicating that they are “healthy”). As input for the simulation, take any node $v_0 \in V$ (the infection’s starting point), and color v_0 in red (indicating that it is “infected”). Then, the basic simulation runs as follows (in pseudo-code):

```
1: while simulation is not finished
2:   for each red node in  $V$ , set  $N(v) \leftarrow \{u \in V : (v, u) \in E\}$ 
3:   for each neighboring node  $u \in N(v)$ 
4:     let  $k$  be the class in which the edge  $v \rightarrow u$  falls into,
```

```
5:     with likelihood  $p_k$ , color  $u$  in red
6:   endfor
7: endfor
8: endwhile
```

Now, to become specific on when to finish the simulation, let a finite time horizon T be given. In each step of the innermost for-loop, the algorithm can simply draw a random quantity of exponential distribution (with parameter $1/\lambda$ as defined above), and advance the current time until it exceeds T . Thus, the full simulation algorithm is only a slight extension of its previous version:

```
1:  $t \leftarrow 0$ 
2: while  $t < T$ 
3:   for each red node in  $V$ , set  $N(v) \leftarrow \{u \in V : (v, u) \in E\}$ 
4:   for each neighboring node  $u \in N(v)$ 
5:     let  $k$  be the class in which the edge  $v \rightarrow u$  falls into,
6:     with likelihood  $p_k$ , color  $u$  in red,
7:     draw an exponentially random  $\Delta t \sim Exp(1/\lambda)$ ,
8:      $t \leftarrow t + \Delta t$ .
9:   endfor
10: endfor
11: endwhile
```

Generating exponentially distributed random numbers in line 7 is a easy by using the inversion method; that is, given a uniformly random value $r \in [0, 1]$, an *Exp*($1/\lambda$)-distributed variate is obtained as $\Delta t = -\lambda \cdot \log r$.

The output of the algorithm is a partially red-colored graph, in which all red nodes are considered as “infected”. Note that nothing is assumed about disinfections or healing of nodes, so as to simulate a worst-case scenario (without any repair attempts). The final rate of infection is simply the fraction of red nodes relative to all nodes in G . In detail, calling $|V|$ the total number of nodes in the network and N the random number of infected nodes, the result is the degree of infection (measurable in %),

$$\text{degree of infection} = \frac{N}{|V|}.$$

Extending the simulation by assuming nodes to recover at random then amounts to adding another random number M to the output N , where M is the number of healed nodes, the degree of infection changes into $(N - M)/|V|$.

As demonstrated later, at least the distribution of N can be found with the help of generating functions depending on the class of edge that failed. The distribution of M cannot be determined in general, as it depends on the particular recovery processes (which are up to organizational regulations).

B. Dealing with Uncertainty

The quantity p_k in the algorithm (used in line 5 of the basic, and line 6 of the full simulation) is so far assumed as an exact figure. Practically, experts may provide only vague and possibly disagreeing estimates on the exact magnitude of the likelihood for a problem to infect a related system. A standard method to deal with this uncertainty is aggregation (e.g., taking the maximum over all different estimates of the likelihoods, or similar). Except for its simplicity, such an approach is

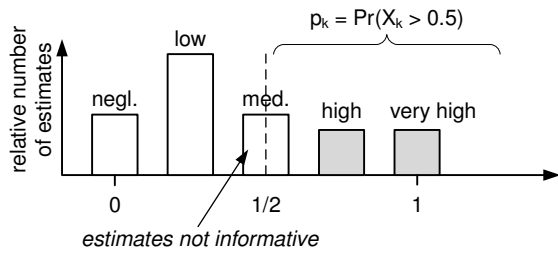


Fig. 1: Treatment of Uncertain/Ambiguous Expert Assessments

quite lossy in terms of information, so the proposed method works somewhat different: let $p_{k,1}, \dots, p_{k,n_k}$ be a set of n_k different estimates for the likelihood p_k . If these estimates are “qualitative”, in the sense that they are given on an ordinal scale (e.g., “low”, “medium” or “high”), then assign arbitrary (but for the application meaningful) values in $[0, 1]$ that preserve the ranks. For example, the scale (low < medium < high) would be mapped to the representatives $(\frac{1}{6} < \frac{1}{2} < \frac{5}{6})$. The overall probability p_k for the problem to jump over an edge of class k is then no longer a fixed number but actually a random variable X_k that is distributed according to the expert’s opinions about edges of class k .

However, the appeal of the simple simulation approach is that the necessary change to incorporate this uncertainty in full power is simply done by taking the decision in line 6 with the likelihood

$$p_k := \Pr[X_k > 0.5]. \quad (1)$$

Under this heuristic, the likelihood for the problem to jump over an edge of class k is determined by how much mass the experts put on the event of an infection. Continuing the above example with the qualitative scale (low < medium < high) for the likelihoods, this would mean that, e.g., a “medium” assessment contributes no information, as it assigns mass to the exact center, or equivalently, puts equal mass on both events, to infect or not infect a node. This is (at least intuitively) meaningful, as infections are more frequent in the simulation if more experts consider an edge as being a likely infection path. Conversely, if experts rate the chance of an infection over an edge of class k rather low, then the simulation adheres to this expected behavior. Figure 1 shows an abstract example of a distribution that is constructed from several experts rating an edge of class k on a five part scale “negligible/low/medium/high/major”. The values of the grayish highlighted bars are merely summed up to give the sought likelihood p_k .

The likelihood used for the simulation according to the above formula is the mass located to the right of the center. This rule offers the additional appeal of being somewhat robust against outliers, which in case of risk expertise elicitation would correspond to extreme risk aversion or extreme willingness to take risks. Experts that fall into either of these classes would not affect the 50%-quantile (that p_k according to equation (1) basically is).

An alternative yet most likely computationally infeasible approach would be averaging over all different configurations of edge likelihoods, which are available from the total set of expert opinions. It is easy to imagine that – if a discrete scale

is used – the number of simulations grows like $\Omega(n^m)$, where m is the number of edge classes, and n is the minimal number of different estimates for p_k per class k . Aggregating all these simulation outputs into a single weighted sum (with weights determined by the relative fraction of how often a particular configuration appears in the overall set) is also only feasible under discrete estimates. If the assessment is on a continuous scale (unlikely but possible in practice), then things become even more involved and unclear in how to define them properly.

C. A Pandemic Criterion from Percolation Theory

The question of primary interest is whether or not an infection stays local or evolves unboundedly across the network, such that every part of the system is ultimately reached. Running the simulation algorithm from section II-A multiple times to see where the infection is leading to, takes a lot of time and effort. Thus, a more direct answer to the question is required.

Alas, a naive use of a shortest-path algorithm to compute the most likely infection path from v_0 to all other nodes in V , is flawed, since this would return *only one* among (in general exponentially) *many* possible ways over which an infection can hit a node. Thus, a more sophisticated model is needed, which percolation theory provides.

Since the formal proofs are messy and involved (see [33] for full details), the following description stays at an intuitive and high level.

Existing percolation models for spread of failures or epidemics hardly take into account diversity among different connections but assume uniform probability of failure [34] (an approach that distinguishes at least directed from undirected connections can be found in [8]). Based on the model as described in section II, especially the different probabilities p_i for each edge class i , the structure of the *infected* part of the network can be determined using probability generating functions (see e.g. [35]). Intuitively, these help answering the following question:

How big is the impact of an error, i.e., what is the expected number of infected nodes due to failure of a component?

The “size” of the impact is herein taken as the average number of infected nodes (taken as the limit over a hypothetical infinitude of independent simulations). A pandemic outbreak is then nothing else than an unbounded average number of infected nodes. The subsequently stated criterion to predict this pandemic is based on this understanding:

Definition 2.1 (Pandemic Outbreak): A pandemic in a graph G occurs if the expected number of affected nodes is unbounded.

For both failure of a node as well as of an edge of class i , a linear equation system in the expected number of infected nodes can be obtained. The coefficient matrix is therein fully determined by the graph topology (specifically the probability q_k for an edge of type k to exist in the network), and the probability p_k for an edge of type k to transport the problem. The scope of the outbreak is governed by these probabilities,

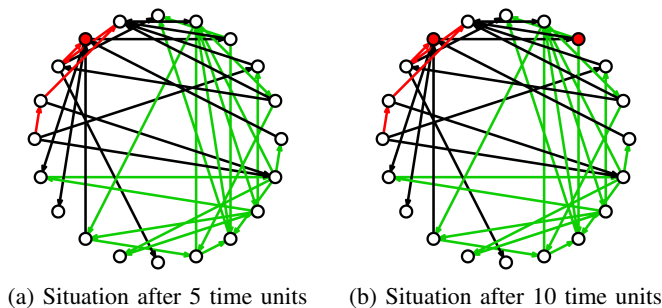


Fig. 2: Bounded Epidemic (Condition (2) satisfied)

and whether or not a pandemic occurs can be decided upon knowing these quantities.

If the network is described through an Erdős-Rényi model [36], the equation system is directly solvable and yields a particularly simple relation between the network properties and a possible pandemic:

Theorem 2.1: Let a network of n nodes with m classes of edges be described through a Erdős-Rényi model, in which an edge of class i exists with probability q_i . Furthermore, let each class i have probability p_i to let an error propagate over an edge of this kind. If

$$1 - np_1q_1 - \dots - np_mq_m > 0, \quad (2)$$

then *no pandemic* (according to def. 2.1) will occur.

The theorem can be illustrated using the simulation algorithm of section II-A: Assume the network to consist of 20 nodes with 3 different classes of connections, shown in red, green and black (cf. Fig. 2 and Fig. 3). Let the probabilities for an edge to exist in each class be $q = (0.1, 0.1, 0.1)$ and let the error propagate over each class with likelihoods $p = (0.1, 0.2, 0.05)$. Then, the hypothesis of 2.1 is satisfied and a rather limited fraction of nodes is expected to fail. Indeed, the simulation results shown in Figure 2 display only one infection (node colored red) having occurred after ten time units after the initial incident. On the contrary, if the probabilities for existence of a link are $q = (0.1, 0.4, 0.25)$ and those for an error propagation are $p = (0.1, 0.3, 0.25)$, then the “anti-pandemic” condition in theorem 2.1 is violated, and a pandemic is expected. The simulation indeed confirms this behavior, as Figure 3 shows only 3 of the 20 nodes remaining healthy 10 time steps after the outbreak (the other 17 nodes are all red).

III. VISUALIZATION BY HEAT-MAPS AND GEO-REFERENCES

In its plain form, the simulation delivers a concrete scenario of infections, and the percolation epidemics criterion judges the (un)boundedness of the average number of infections. Both pieces of information can be made much more comprehensible by a visualization technique, which adds the geographic location to each node, and – using color codes – indicates the likelihood of a node becoming infected as the relative frequency of its red-coloring over many simulations of the outbreak. Speaking percolation language, a pandemic outbreak

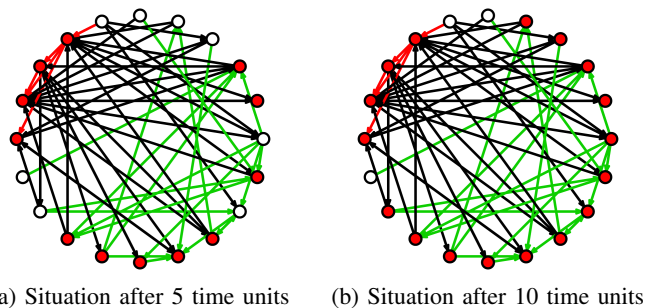


Fig. 3: Pandemic Outbreak (Condition (2) violated)

will then become visible by a giant red component in the infrastructure, while a locally bounded epidemic will appear as a smaller bounded cluster in the network infrastructure.

For the general problem of incident propagation, it is useful to consider the way in which infrastructures can be modeled beyond pure physical connections. This is a standard step to gain an understanding of the infrastructure, and it can have a second use in identifying the proper edge classes to set up the percolation model as described in section II. Usually, such modeling is aided by a good visualization, which also can have a double use, namely, to set up the model and to visualize the results.

To get started with the model, [32] proposes a modularized approach that can be adopted here too: This process starts with an identification of physical components and applications, and interdependencies therein. Figure 4a displays an example of such a dependency model, in which an application A depends on several subcomponents, e.g., an application server (AS1), a data warehouse (DW), which itself relies on (two) databases DB1, DB2, and components are virtualized (represented by virtual machine nodes VM1, ..., VM5, which in turn run on physical servers, etc.

Figure 4b represents a higher-level view that is restricted basically to interdependencies between applications. Thus, in distinguishing dependencies between components from those between applications, two edge classes for the percolation analysis have already been identified. The resulting infrastructure model is nothing else than a weighted directed graph that captures dependencies of physical components and/or applications on one another. More precisely, nodes are components or applications and edges are relationships between components, each of which falls into a specific edge class (determined by the type of dependency), and has a specific behavior in error propagation. In this logical view, model components-to-components relations, as well as components-to-application and applications-to-application-relationships, by defining the respective edge classes. The graph representation is in no way restricted in its visual elements, so the model itself can use boxes and circles to distinguish components from applications, while both are being abstracted to simple “nodes” in the graph model G for the percolation.

On the so-obtained graphical dependency model, the pandemic criterion (theorem 2.1) can be invoked to obtain a first (initial) risk estimate. For a more detailed picture, the proposed

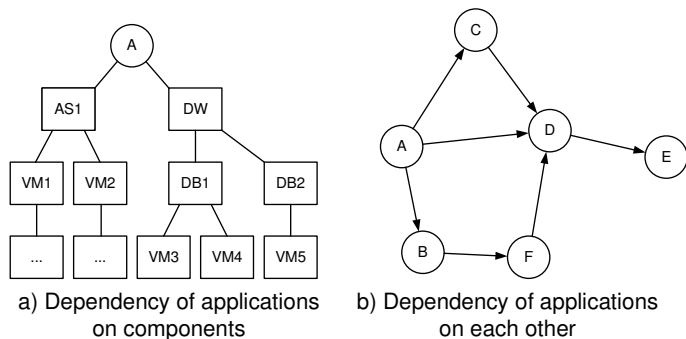


Fig. 4: Dependency Model – Example

simulation can be executed, with its results fed back into the visualization technique of [32], to get the sought “zoomable” information and visualization system. The visualization is nothing else than the initial dependency model, only being augmented by color codes to indicate neuralgic spots. A decision maker can then coarsen or refine the graph by clustering nodes or dissolving cluster-nodes to display more or less details. For example, one can (un)hide the physical dependencies below application A from the higher-level application dependency graph in Figure 4b. Also it is easy to augment nodes with meta-information, such as the node type, the geographical location, the likelihood of an infection, and whatever else is relevant. This is the typical case for visualizations as being used in a security operation center, as they are used by governments or agencies to supervise the security situation in IT infrastructures (e.g., a local intranet, a critical infrastructure, etc.).

An important part of the meta-information is directly obtained from the percolation simulation, and meta-information about the infrastructure as a whole can be gathered from the pandemic criterion as given in section II-C. For example, imagine that a number of independent simulations are done, then the outcome of each one (specific infection or error propagation scenario) may be different per run. Simply counting the relative number of times that application A has been infected by the simulated incident, approximates the likelihood for A to be eventually in trouble. This likelihood can be displayed using a color indicator, as shown in Figure 5. That is, instead of coloring a node green or red as in the simulation, the analysis would now assign any color between purple (high likelihood of a problem) until green (low likelihood of a problem), to graphically visualize which parts of an infrastructure are critical, vulnerable, etc.

The resulting picture is called a heat-map, and could look like figure 5. In this image, the circles represent applications (only), and the color codes indicate the vulnerability of an application to running into trouble. In the simplified picture shown in figure 5, application A as being infected by malware may also infect C with some likelihood, but D remains somewhat secure, although E is affected with high probability again. This simplified view does not explain this effect, which, however, may be due to the error propagating from A to E over some edges that belong to a class that is hidden in this view.

In general, by displaying different edge classes, different pictures arise, and explanations for vulnerabilities can be found from the graphical model almost directly.

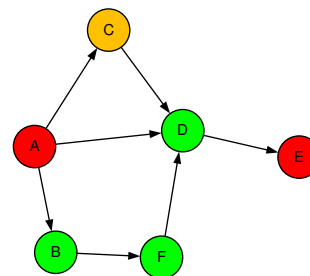


Fig. 5: Heatmap example

In a second step of the visualization, the risk manager can look at the meta-information attached to the nodes in the heatmap, so as to change the heatmap according to what the decision maker are currently interested in. For example:

- Layers of security: these determine the depth of information details. In the first layer, applications and their relations are presented (as one edge class). In a second layer, components that applications depend on can be represented (another edge class), and so on. Thus, the security layers may directly correspond to edge classes. Depending on what edge classes are being displayed, the picture may change (see figure 6).

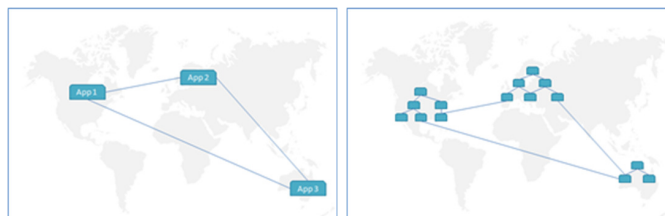


Fig. 6: Showing different layers

- Protective target: the simulation/percolation may refer to different security targets, which amounts to assigning different probabilities to the existing edge classes. For example, if the security goal is integrity, then inconsistencies in the data delivered by application A may cause subsequent errors in application B as well (as A delivers data to B). If the protective target is availability, then an outage of application A may cause much less troubles with application B. The respective heatmaps may look very different, and – combined with geographical data – directly point to the most likely problems (see figure 7).
- Different filters can be used to omit parts/details from the picture if they are not relevant. These filters could concern the infection grade (to show only highly vulnerable spots) or to restrict the view on a specific country or subsidiary of an enterprise. For example, an important filter concerns the connectivity of a node, to see which nodes are easy to isolate or repair. Such filters are useful to define remediation plans, if an epidemic (or pandemic) is indicated.



Fig. 7: Showing different protective targets

IV. CONCLUSION

Percolation offers a simple, intuitive and powerful model of epidemics spreading in networks, which is particularly applicable to risk propagation. A current limitation of this model is its ignorance of recovery or healing effects, which may take place in long run simulations or even when advanced persistent threats (APTs) are being mounted (when malware that has not yet become active is removed by coincidence or upon detection).

For a decision maker and security expert, both not concerned with the details of the theory but how to apply its results in practice, the method offers several appealing features: first, it can easily deal with uncertainty and ambiguity in the model parameters. Although the model's outcome is only as good as its input, the full variety of methods to deal with uncertainty is available to aggregate diverging opinions into a justified model parameter. In this context, equation (1) is robust against outliers, i.e., unusual/uninformed risk assessments, but a Bayesian estimate would be equally possible. Second, the theory as such provides a direct answer to a frequent direct question, i.e., "whether or not the system is going to be in trouble". If the answer to the latter is positive, then the visualization technique (as sketched in section III) allows to dig into the simulation data (as obtained by the simple algorithm in section II-A), to let a decision maker "zoom-into" the picture to see different scenarios of incident propagation, to manually and informedly refine her/his opinion about what needs to be done.

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MAI and Noise Constrained LMS Algorithm for MIMO CDMA Linear Equalizer

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Abstract—This paper presents a constrained least mean squared (LMS) algorithm for MIMO CDMA linear equalizer is presented, which is constrained on spreading sequence length, number of subscribers, variances of the Gaussian noise and the multiple access interference (MAI) plus the additive noise (introduced as a new constraint). The novelty of the proposed algorithm is that MAI and MAI plus noise variance has never been used as a constraint in MIMO CDMA systems. Convergence analysis is performed for the proposed algorithm in case when statistics of MAI and MAI plus noise are available. Simulation results are presented to compare the performance of the proposed constrained algorithm with other constrained algorithms and it is proved that the new algorithm has outperformed the existing constrained algorithms.

Index Terms—Least mean squared (LMS), multiple input, multiple output (MIMO), linear equalizer, multiple access interference (MAI), Variance, AWGN, adaptive algorithm

I. INTRODUCTION

It is shown in the literature that performance of an adaptive algorithm may be enhanced if partial knowledge of a particular channel is blended in the algorithm design [1], [2]. Based on this idea, [1] presented an algorithm (Noise-constrained LMS) for identification and tracking of finite impulse response (FIR) channels using the variance of the receiver noise. An advantage of the noise constrained least mean squared (NCLMS) is that it outperforms the LMS algorithm while keeping the same computational complexity. A complementary pair LMS (CP-LMS) [3] was initiated by using constrained optimization technique named augmented Lagrangian (AL) which can be utilized to solve the problem of selecting an appropriate update step-size in LMS algorithm. This technique was utilized in [4], in which the knowledge of the variance of MAI plus noise was incorporated to develop what is called the constrained LMS algorithm (MNCLMS) for CDMA systems.

Since MAI and the additive white Gaussian noise (AWGN) effect performance of CDMA multi user, multi-antenna environment, it is imperative to design a receiver architecture which will negate the effect of MAI and additive noise. This requires a MIMO implementation of the MNCLMS algorithm presented in [4]. The proposed constrained algorithm is developed by incorporating MIMO MAI and noise variances thus resulting in a generalized MIMO MAI plus noise constrained LMS (MIMO-MNCLMS) adaptive algorithm. As it is

generalized, we can deduce MAI constrained algorithm, noise constrained algorithm and zero constrained noise algorithm as special cases.

This paper is organized as follows. After introductory remarks, section 2 presents the motivation for the algorithm development. Algorithm development is presented in section 3 whereas section 4 deals with the convergence analysis. Computational complexity of the algorithm is given in section 6. In order to find the theoretical findings, simulation results are presented in Section 7. Concluding remarks are shown in section 8.

II. MOTIVATION

Adaptive algorithms such as LMS and RLS do not use models for channel coefficients and/or noise, whereas model based algorithms utilize various types of models such as random walk, auto-regressive etc. for coefficients and AWGN. Model parameters are either known or jointly estimated with a channel. Adaptive algorithms can be inferred to as model based algorithms with model parameters' choice dependent on data [5]. It has been reported in the literature that practically it is possible to enhance the performance of an adaptive algorithm if partial knowledge of the channel is available provided that the computational cost of an algorithm is not increased tremendously. According to the noise constrained LMS algorithm [6]

$$\mathbf{w}_{n+1} = \mathbf{w}_n + \mu_n^l e_n \mathbf{X}_n \quad (1)$$

$$\mu_{n+1} = 2\mu_n^l (1 + \gamma \lambda_n) \quad (2)$$

$$\lambda_{n+1} = \lambda_n + \beta \left[\left(\frac{1}{2} e_n^2 - \sigma_{v_n}^2 \right) - \lambda_n \right] \quad (3)$$

Where λ , α and β are positive constants. This is a variable step size (LMS) algorithm because step size rule is applicable due to the constraint on the noise variance. The computational cost of the aforementioned algorithm is the same as of LMS but the convergence rate of the noise constrained LMS algorithm is much faster than the LMS due to its three independent parameters.

MAI is the major limiting factor in the system performance of a multiuser environment, it is required to design a multi receiver scheme which will negate the effect of MAI and the additive noise. Previous research work treated MAI as a part of interfering noise. This assumption is not practically correct which led this work to use MAI as an additional constraint by using structured information contained in it. We also believe that by using the combined information of MAI and the interfering noise to form into a single constraint would result in an algorithm which would outperform the noise only constrained algorithm. It is worth mentioning here that by using MAI alone as a constraint is not a viable choice since noise is an undeniable physical constraint and may not be ignored while developing such algorithms.

As NCLMS algorithm is noise constrained only, a new constrained algorithm is established, by incorporating MAI variance as well as the noise variance, thus resulting in a generalized MAI plus noise constrained LMS (MNCLMS) adaptive algorithm. Since this proposed algorithm is generalized, this algorithm is able to deduce MAI constrained, noise constrained and zero constrained noise algorithms as special cases.

III. ALGORITHM DEVELOPMENT

In MIMO-CDMA system with N transmitting and M receiving antennas, output of m^{th} matched-filter, matched to intended subscriber's (subscriber 1) spreading sequence consists of the desired user's component, b_m^l , MIMO-MAI, U_m^l and the white Gaussian noise, v_m^l as

$$y_m^l = b_m^l + U_m^l + v_m^l \quad (4)$$

$$= b_m^l + Z_m^l$$

where y_m^l is the output of the matched filter and Z_m^l is the MIMO MAI and noise at the m^{th} receiving antenna with variance $\sigma_{Z_m^l}^2$. The statistical characterization (variance) of MIMO CDMA MAI and noise is given in [7]. The MIMO CDMA MAI is given by

$$Z_m^l = \sum_{n=1}^N \sum_{k=2}^K A^k b_n^{l,k} \rho^{K,1} h_{mn}^l + v_m^l \quad (5)$$

$$= U_m^l + v_m^l$$

In 5, K is the number of subscribers, whereas $s_n^{l,k}(t)$ represents the rectangular signature waveform with a random signature sequence of k^{th} subscriber defined as $(l-1)T_b \leq t \leq lT_b$, whereas T_b and T_c represent bit period and the chip intervals respectively related by $N_c = T_b/T_c$. $b_n^{l,k}$ is the input bit stream of the k^{th} subscriber, h_{mn}^l is the l^{th} channel tap between the n^{th} transmitter and the m^{th} receiver, A^k is the transmitted amplitude of the k^{th} subscriber and v_m is the AWG noise having zero mean and variance σ_v^2 at the m^{th} receiver. The cross-correlation between the signature sequences of subscriber j and k for the l^{th} symbol is given as

$$\rho_l^{k,j} = \int_{(l-1)T_b}^{lT_b} s_n^k(t) s_n^j(t) dt \quad (6)$$

$$= \sum_{i=1}^{N_c} c_{l,i}^k c_{l,i}^j$$

where $c_{l,i}^k$ is the normalized spreading sequence of the subscriber k for the l^{th} symbol.

The MIMO CDMA MAI consists of two random variables given as

$$U_m^l = \sum_{n=1}^N \sum_{k=2}^K A^k b_n^{l,k} \rho^{K,1} h_{mn}^l \quad (7)$$

$$= \sum_{n=1}^N \mathcal{C} h_{mn}^l,$$

where \mathcal{C} is MAI in AWGN environment [8]. The desired subscriber's component is written as

$$b_m^l = \left[\left(\mathbf{h}_{m1}^l \right)^T \left(\mathbf{h}_{m2}^l \right)^T \dots \left(\mathbf{h}_{mn}^l \right)^T \right] \times$$

$$\left[\left(\mathbf{x}_1^l \right)^T \left(\mathbf{x}_2^l \right)^T \dots \left(\mathbf{x}_N^l \right)^T \right]^T \quad (8)$$

In (8), $\mathbf{h}_{m1}^l = [h_{m1}^l h_{m1}^{l-1} \dots h_{m1}^{l-L+1}]^T$ is the time varying impulse response (TVIR) of the channels and is an $L \times 1$ vector. $\mathbf{x}_n^l = [x_n^l x_n^{l-1} \dots x_n^{l-L+1}]^T$ is an $L \times 1$ vector. The filter impulse response (FIR) of the LE which consists of an FFF is given by

$$\mathbf{w}_n^l = \left[\left(F_n^l \right)^T \right] \quad (9)$$

In (9), F_n^l is the n^{th} multiple input, single output (MISO) FFF with dimension of $ML \times 1$, where L is the taps of FFF and M is the number of receivers.

The mean-squared error (MSE) to be minimized is

$$J(\mathbf{w}_n^l) = [e_n^l]^2 \quad (10)$$

where e_n^l is the error between output of a matched filter and an adaptive filter and is shown to be

$$e_n^l = \hat{x}_n - \mathbf{w}_n^l D_n^l \quad (11)$$

here, D_n^l is the combined input to the LE and is given by

$$D_n^l = \left[\left(\mathbf{y}_n^l \right)^T \right] \quad (12)$$

and is of the order of $(ML \times 1)$.

$$\mathbf{y}_n^l = \left[\left(\mathbf{y}_1^l \right)^T \left(\mathbf{y}_2^l \right)^T \dots \left(\mathbf{y}_M^l \right)^T \right]^T \quad (13)$$

is the input to the FFF of dimension $ML \times 1$ and is a collection of vectors consisting of y_m^l given by

$$\mathbf{y}_m^l = \left[y_m^l y_m^{l-1} \dots y_m^{l-L+1} \right]^T \quad (14)$$

and

$$\hat{\mathbf{x}}_n = \mathbf{w}_o^T D_n^l \quad (15)$$

$$= \mathbf{x}_n^l + \bar{v}_n^l$$

or

$$\mathbf{x}_n^l = \mathbf{w}_o^T D_n^l - \bar{v}_n^l \quad (16)$$

\bar{v}_n^l is the filtered noise which passes through FFF and is composed of MAI and noise.

Minimizing cost function in (10) over \mathbf{w}_n^l will give an appropriate value at time l . In other words, $\mathbf{w}_{opt} = H_{mn}^l$ (of size $1 \times NL$ matrix) with $\hat{f}(opt) = \sigma_{\bar{v}_n^l}^2$ represents MSE. It is known that knowledge of $\sigma_{\bar{v}_n^l}^2$ [6] is helpful in selecting the search direction for an adaptive algorithm in multisubscriber case which is quite similar to the MNCLMS algorithm in single subscriber environment.

To minimize $J(\mathbf{w}_n^l)$ over \mathbf{w}_n^l subject to the constraint $J(\mathbf{w}_n^l) = \sigma_{\bar{v}_n^l}^2$, the Lagrangian of this is given by

$$J_1(\mathbf{w}_n^l, \lambda_n^l) = J(\mathbf{w}_n^l) + \lambda_n^l [J(\mathbf{w}_n^l) - \sigma_{\bar{v}_n^l}^2] \quad (17)$$

Since critical values of λ_n^l are not unique in this case so we are using an augmented Lagrangian to get rid of this issue by defining the under mentioned cost function

$$J_2(\mathbf{w}_n^l, \lambda_n^l) = J(\mathbf{w}_n^l) + \gamma \lambda_n^l [J(\mathbf{w}_n^l) - \sigma_{\bar{v}_n^l}^2] - \gamma_n \lambda_n^2 \quad (18)$$

Solution to (18) is given by using the method used in [6] and is shown to be

$$\mathbf{w}_n^{l+1} = \mathbf{w}_n^l + \mu_n^l e_n^l D_n^l \quad (19)$$

where μ_n^l is the positive step size and is given by

$$\mu_n^l = \mu_n \left(1 + \gamma_n \lambda_n^l \right), \quad m = 1, 2, \dots, M \quad (20)$$

$$\lambda_n^{l+1} = \lambda_n + \beta_n \left[\frac{1}{2} (e_n^{l2} - \sigma_{\bar{v}_n^l}^2) - \lambda_n^l \right], \quad m = 1, 2, \dots, M \quad (21)$$

Since channel taps are independent of spreading sequence as well as data sequence, the interferer's components are also independent of one another having zero mean.

The variance of MAI for an equal power can be written as

$$U_m^l = A^2 \left(\frac{K-1}{N_C} \right) \sum_{n=1}^N E [h_{mn}^2] \quad (22)$$

whereas the variance of MAI in the unequal transmitted power is given by

$$U_m^l = \frac{\sum_{k=2}^k (A^k)^2}{N_C} \sum_{n=1}^N E [h_{mn}^2] \quad (23)$$

In (22) and (23), $E [h_{mn}^2]$ is the second moment of $E [h_{mn}]$.

IV. CONVERGENCE ANALYSIS

In this section, convergence behavior of our proposed algorithm in the presence of MAI and noise for an LE receiver is analyzed. Following independent assumptions have been used while performing the convergence analysis [9].

Assumption 1: The input process $\{\mathbf{x}_n^l\}$ is an independent and identically distributed (i.i.d) random variable with auto correlation matrix of $\mathbf{R}_{xx} = E [\mathbf{x}_n^T \mathbf{x}_n]$.

Assumption 2: The noise sequence is a zero mean i.i.d. sequence, Gaussian random variable having variance $\sigma_{\bar{v}_n}^2$. This sequence is independent of the input process.

Assumption 3: MAI in AWGN environment represented by U_m^l is zero mean Gaussian random variable with variance $\sigma_{U_m}^2$. It is independent of the input process as well as the noise.

Assumption 4: For any fixed time. say l , step size μ_n^l and weight vector \mathbf{v}_n^l (defined later) are statistically independent of each another.

The above mentioned assumptions are justified as follows

Although assumption 1 is not true in reality but is commonly used in literature and it has been shown in the literature that simulation results under this condition closely match with the analytical results.

Assumption 2 termed as independent assumption, is also commonly used in the literature and could be justified in many cases.

Gaussian approximation of MAI in AWGN is extensively used in the adaptive filtering literature [10]. Since MAI is independent of the noise process, assumption 3 can be validated. In this way, MAI and noise is independent of the input process as well.

Generally μ_n^l and \mathbf{v}_n^l are dependent but if the parameters are chosen in such a way that steady-state variance of μ_n^l and or \mathbf{v}_n^l is small, assumption 4 can be justified.

\mathbf{x}_k and \mathbf{x}_n are also independent as both are uncorrelated for $n \neq k$. Thus the input vector \mathbf{x}_n and the weight error vector (defined later) \mathbf{v}_n^l are also independent [11].

Weight update equation of the proposed algorithm is written as

$$\mathbf{w}_n^{l+1} = \mathbf{w}_n^l + \mu_n^l e_n^l D_n^l \quad (24)$$

Where $n = 1, 2, \dots, N$ represents the n^{th} stream.

If $\mathbf{w}_{n(opt)}$ is said to be the optimum value of weight according to the Wiener solution then weight error vector \mathbf{v}_n^l can be defined as

$$\mathbf{v}_n^l = \mathbf{w}_{n(opt)} - \mathbf{w}_n^l \quad (25)$$

Subtracting $\mathbf{w}_{n(opt)}$ from both sides of (19) results in

$$\mathbf{v}_n^{l+1} + \mathbf{w}_{n(opt)} = -\mathbf{v}_n^l + \mathbf{w}_{n(opt)} - \mu_n^l e_n^l D_n^l$$

$$\mathbf{v}_n^{l+1} = -\mathbf{v}_n^l - \mu_n^l e_n^l D_n^l \quad (26)$$

The desired response of the decision device ($\tilde{\mathbf{x}}_n^l$) can be expressed as

$$\tilde{\mathbf{x}}_n^l = \mathbf{w}_{(opt)}^T D_n^l \quad (27)$$

By using 27, it can be shown in case of system identification scenario, e_n^l could be setup as below:

$$e_n^l = \mathbf{w}_{(opt)}^T - (\mathbf{w}_n^l)^T \quad (28)$$

For $n = 1, 2, \dots, N$.

(28) can be compactly written as

$$e_n^l = \left(\mathbf{v}_n^l \right)^T D_n^l = \left(D_n^l \right)^T \mathbf{v}_n^l \quad (29)$$

Now the recursion of weight error vector can be shown as

$$\mathbf{v}_n^{l+1} = \mathbf{v}_n^l - \mu_n^l D_n^l (D_n^l)^T \mathbf{v}_n^l \quad (30)$$

$$\mathbf{v}_n^{l+1} = \{\mathbf{I} - \mu_n^l D_n^l (D_n^l)^T\} \mathbf{v}_n^l \quad (31)$$

Taking expectation on both sides of (31) with the assumptions made earlier, will yield

$$\bar{\mathbf{v}}_n^l = \left[\mathbf{I} - \bar{\mu}_n^l E \left\{ D_n^l (D_n^l)^T \right\} \right] \bar{\mathbf{v}}_n^l \quad (32)$$

$\bar{\mu}_n^l = E[\mu_n^l]$ is the mean step size and $\bar{\mathbf{v}}_n^l = E[\mathbf{v}_n^l]$ is the weight error vector in the equation above. Whereas

$$E \left[D_n^l (D_n^l)^T \right] = E \left[(\mathbf{y}_n^l)^T \right] \left[(\mathbf{y}_n^l)^T \right]^T$$

or

$$\tilde{R} = E \left[(\mathbf{y}_n^l)^T \right]^T \left[(\mathbf{y}_n^l)^T \right] \quad (33)$$

or

$$\tilde{R} = E \left[D_n^l (D_n^l)^T \right] \quad (34)$$

where \tilde{R} is the correlation matrix of input process

A. Auto-correlation Structure of MIMO-CDMA Linear Equalizer (LE)

The correlation matrix \tilde{R} is given by

$$\tilde{R} = R_{D_n^l D_n^l} \quad (35)$$

B. Eigenvalues of Linear Equalizer (LE)

If $\gamma_1, \gamma_2, \gamma_3 \dots \gamma_N$ represent the eigenvalues of \tilde{R} , then the an important condition for the convergence in the mean is given as

$$\left| 1 - \bar{\mu}_n^l \gamma_N \right| < 1,$$

The value of $\bar{\mu}_n^l$ is bounded in the range

$$0 < \bar{\mu}_n^l < \frac{2}{\gamma_{max}} \quad (36)$$

where γ_{max} is the maximum eigenvalue of \tilde{R} . By using a strong and a much simpler condition for convergence of mean weight error vector [12] can be given as

$$\mu_{max} < \frac{2}{\gamma_{max}} \quad (37)$$

V. TRANSIENT ANALYSIS OF THE PROPOSED ALGORITHM

Transient analysis of an adaptive algorithm is very important to observe the convergence behavior of an adaptive algorithm and to derive steady-state expressions for different error performance measures, such as EMSE and mean-square deviation (MSD). Basically, transient analysis is the observation of the time-evolution of the adaptive algorithms when there are variations in the signal statistics; or in other words to study the learning mechanism of an adaptive algorithm. Energy conservation method is used to carry out the transient analysis [9].

A. Error Measures

Transient analysis of an adaptive algorithm deals with the time evaluation of $E \left[|e_n^l|^2 \right]$ and $E \left[\|\mathbf{v}_n^l\|^2 \right]$, where \mathbf{v}_n^l is the weight error vector and is given by

$$\mathbf{v}_n^l = \mathbf{w}_n^l (opt) - \mathbf{w}_n^l \quad (38)$$

For some symmetric positive definite weighting matrix Ω , We define, weighted a priori and a posteriori estimation errors as

$$e_{an}^\Omega = \left(\mathbf{v}_n^l \right)^T \Omega D_n^l \quad (39)$$

and

$$e_{pn}^\Omega = \left(\mathbf{v}_n^{l+1} \right)^T \Omega D_n^l \quad (40)$$

For the case when, $\Omega = I$, the weighted a priori and a posteriori estimation errors defined above will be reduced to a standard a priori and a posteriori estimation errors, respectively, i.e

$$e_{an}^l = \left(D_n^l \right)^T \mathbf{v}_n^l \quad (41)$$

and

$$e_{pn}^l = \left(D_n^l \right)^T \mathbf{v}_n^{l+1} \quad (42)$$

It will be elaborated later in that different choices of Ω will yield different performance measures for the evaluation of an adaptive filter.

Since,

$$\begin{aligned} e_n^l &= \mathbf{x}_n^l - \left(\mathbf{w}_n^l \right)^T D_n^l \\ &= \left(\mathbf{w}_n^{(opt)} \right)^T - \bar{\mathbf{v}}_n^l - \left(\mathbf{w}_n^l \right)^T D_n^l \\ e_n^l &= \left(\mathbf{v}_n^l \right)^T D_n^l - \mathbf{v}_n^l \\ &= e_{an}^l - \bar{\mathbf{v}}_n^l \end{aligned} \quad (43)$$

Also, by using (40) and (26) it can be shown that

$$e_{pn}^{\Omega,l} = e_{an}^{\Omega,l} - \mu_n^l \|D_n^l\|_\Omega^2 e_n^l \quad (44)$$

B. Performance Measures

The EMSE of the proposed algorithm is given by

$$EMSE = E \left[|e_{an}^l|^2 \right] \quad (45)$$

and steady state EMSE is

$$E \left[|e_{an}^l|^2 \right] = \xi_\infty \quad (46)$$

C. Fundamental Weighted Energy Relation

The fundamental weighted-energy conservation relation [9] is used in this section to develop the framework for the transient analysis of the proposed MNCLMS algorithm. (44) can be expressed as

$$e_n^{\Omega,l} = \frac{e_n^{\Omega,n} - e_{pn}^{\Omega,l}}{\mu_n^l \|D_n^l\|_{\Omega}^2} \quad (47)$$

By using (26) and (47), the following equation can be established

$$\bar{\mathbf{v}}_n^{l+1} = \mathbf{v}_n^l - \frac{D_n^l}{\|D_n^l\|_{\Omega}^2} \left[e_{an}^{\Omega,l} - e_n^{\Omega,l} \right] \quad (48)$$

The fundamental weighted-energy conservation relation can be shown as

$$\mathbf{v}_n^{l+1} + \frac{1}{\|D_n^l\|_{\Omega}^2} \left| e_{an}^{\Omega} \right|^2 = \|\mathbf{v}_n^l\|_{\Omega}^2 + \frac{1}{\|D_n^l\|_{\Omega}^2} \left| e_{pn}^{\Omega} \right|^2 \quad (49)$$

49 describes, how the weighted energies of the error quantities evolve with time. Different choices of Ω will yield different performance measures for the evaluation of an adaptive filter[9].

D. Time Evolution of the Weighted Variance

This section deals with derivation of time evolution of the weighted variance for the proposed MNCLMS algorithm using the fundamental weighted-energy conservation relation equation (49). Substituting (44) into (49) and taking expectation on both sides will result in

$$E \left[\|\mathbf{v}_n^{l+1}\|_{\Omega}^2 \right] = E \left[\|\mathbf{v}_n^l\|_{\Omega}^2 \right] - 2\bar{\mu}_n^l E \left[e_{an}^{\Omega} e_n \right] + \overline{(\mu_n^l)^2} E \left[\|D_n^l\|_{\Omega}^2 e_n^2 \right] \quad (50)$$

where $\overline{(\mu_n^l)^2}$ is $E[(\mu_n^l)^2]$. Next, the expectations in the second and third terms on the right hand side of equation (50) is evaluated by using the following assumption

A5): For any constant matrix Ω and for all l , e_{an}^{Ω} and e_{an}^{Ω} are jointly Gaussian.

This assumption seems reasonable for longer filters using the central limit theorem [13], [14], [4]. So, $E \left[e_{an}^{\Omega} e_n \right]$ can be simplified as

$$\begin{aligned} E \left[e_{an}^{\Omega} e_n \right] &= E \left[(D_n^l)^T \Omega \mathbf{v}_n^l (D_n^l)^T I \mathbf{v}_n^l \right] = \\ &E \left[(\mathbf{v}_n^l)^T \left(\Omega (D_n^l) (D_n^l)^T I \right) \mathbf{v}_n^l \right] \\ &= E \left[\|\mathbf{v}_n^l\|_{\Omega E[(D_n^l)(D_n^l)^T]}^2 \right] = E \left[\|\mathbf{v}_n^l\|_{\Omega \bar{R}}^2 \right] \end{aligned} \quad (51)$$

Now, $E \left[\|D_n^l\|_{\Omega}^2 e_n^2 \right]$ is being solved by using the following assumption

A6): The adaptive filter is long enough so that $\|D_n^l\|_{\Omega}^2$ and e_n^2 are uncorrelated [9].

This assumption is more realistic when the filter length gets longer[9]. As MAI plus noise is independent of D_n^l , expectation $E \left[\|D_n^l\|_{\Omega}^2 e_n^2 \right]$ can be simplified as

$$E \left[\|D_n^l\|_{\Omega}^2 e_n^2 \right] = E \left[\|D_n^l\|_{\Omega}^2 \right] E \left[e_n^2 \right] = E \left[\|D_n^l\|_{\Omega}^2 \right] \left(E \left[(e_{an}^l)^2 \right] - \sigma_{v_n^l}^2 \right) \quad (52)$$

Now using (51) and (52) and, (50) can be written as

$$E \left[\|\mathbf{v}_n^{l+1}\|_{\Omega}^2 \right] = E \left[\|\mathbf{v}_n^l\|_{\Omega}^2 \right] - 2\bar{\mu}_n^l E \left[\|\mathbf{v}_n^l\|_{\Omega \bar{R}}^2 \right] + \overline{(\mu_n^l)^2} E \left[\|D_n^l\|_{\Omega}^2 \right] \left(\xi_n^l - \sigma_{v_n^l}^2 \right) \quad (53)$$

(53) shows the time evaluation or the transient behavior of the weighted variance $E \left[\|\mathbf{v}_n^l\|_{\Omega}^2 \right]$ for any constant weight matrix Ω . Different performance measures can be achieved by the proper choice of the weight matrix Ω .

E. Constructing the Learning Curves for the Excess Mean Square Error (EMSE)

Learning curves for the EMSE can be constructed by using $EMSE = E \left[(e_{an}^l)^2 \right] = E \left[\|\mathbf{v}_n^l\|_{\bar{R}}^2 \right]$. If $\Omega = I, \bar{R}, \dots, \bar{R}^{N-1}$, a set of relations can be obtained. Now by using the Cayley-Hamilton theorem, following can be established

$$\Omega = -p_0 I - p_1 \bar{R} - \dots - p_{N-1} \bar{R}^{N-1} \quad (54)$$

where

$$p(x) \triangleq \det(xI - \bar{R}) = p_0 + p_1 x + \dots + p_{N-1} x^{N-1} + x^N \quad (55)$$

is the characteristic polynomial of \bar{R} . Consequently,

$$\begin{aligned} E \left[\|\mathbf{v}_n^{l+1}\|_{\bar{R}^{N-1}}^2 \right] &= E \left[\|\mathbf{v}_n^l\|_{\bar{R}^{N-1}}^2 \right] + 2\bar{\mu}_n^l \\ &\left(p_0 E \left[\|\mathbf{v}_n^l\|_I^2 \right] + p_1 E \left[\|\mathbf{v}_n^l\|_{\bar{R}}^2 \right] + \dots \right. \\ &\left. + p_{N-1} E \left[\|\mathbf{v}_n^l\|_{\bar{R}^{N-1}}^2 \right] \right) \\ &+ \overline{(\mu_n^l)^2} E \left[\|D_n^l\|_{\bar{R}^{N-1}}^2 \right] \left(\xi_n^l + \sigma_{v_n^l}^2 \right) \end{aligned} \quad (56)$$

So,

$$\Upsilon_n^l = \begin{bmatrix} 1 & -2\bar{\mu}_n^l & 0 & \dots & 0 & 0 \\ 0 & 1 & -2\bar{\mu}_n^l & 0 & \dots & 0 \\ \vdots & 0 & 1 & -2\bar{\mu}_n^l & \dots & \vdots \\ 0 & 0 & 0 & 1 & -2\bar{\mu}_n^l & 0 \\ 0 & 0 & \dots & 0 & 1 & -2\bar{\mu}_n^l \\ 2\bar{\mu}_n^l p_0 & 2\bar{\mu}_n^l p_1 & 2\bar{\mu}_n^l p_2 & \dots & 2\bar{\mu}_n^l p_{N-2} & 1 + 2\bar{\mu}_n^l p_{N-1} \end{bmatrix} \quad (57)$$

And

$$\bar{\omega}_n^l = \begin{bmatrix} E \left[\|\mathbf{v}_n^l\|_I^2 \right] \\ E \left[\|\mathbf{v}_n^l\|_{\bar{R}}^2 \right] \\ \vdots \\ E \left[\|\mathbf{v}_n^l\|_{\bar{R}^{N-1}}^2 \right] \end{bmatrix} \quad (58)$$

Similarity

$$a_n^l = \begin{bmatrix} E \left[\|D_n^l\|_I^2 \right] \\ E \left[\|D_n^l\|_{\tilde{R}}^2 \right] \\ \vdots \\ E \left[\|D_n^l\|^2 \right] \end{bmatrix} \quad (59)$$

By combining 57, 58 and 59,

$$\bar{\omega}_n^{l+1} = \Upsilon_n^l \bar{\omega}_n^l + \overline{(\mu_n^l)^2} (\xi_n^l + \sigma_{v_n^l}^2) a_n^l \quad (60)$$

VI. STEADY-STATE ANALYSIS OF THE MNCLMS ALGORITHMS

Steady-state analysis of an adaptive filter is performed to study the behavior of steady-state EMSE and MSD. Steady-state performance measures is also obtained by analyzing (50) when $l \rightarrow \infty$ which is presented in the next section i.e

$$\lim_{l \rightarrow \infty} E \left[\|\mathbf{v}_n^{l+1}\|_{\Omega}^2 \right] = \lim_{l \rightarrow \infty} E \left[\|\mathbf{v}_n^l\|_{\Omega}^2 \right] \quad (61)$$

So at steady-state (53) will become

$$2\bar{\mu}_n^{\infty} \lim_{l \rightarrow \infty} E \left[\|\mathbf{v}_n^l\|_{\Omega \tilde{R}}^2 \right] = \overline{(\mu_n^{\infty})^2} \lim_{l \rightarrow \infty} E \left[\|D_n^l\|_{\Omega}^2 \right] (\xi_n^{\infty} + \sigma_{v_n^l}^2) \quad (62)$$

Where $\bar{\mu}_n^{\infty}$, $\overline{(\mu_n^{\infty})^2}$ and ξ_n^{∞} are steady-state values of $\bar{\mu}_n^l$, $(\mu_n^l)^2$ and ξ_n^l respectively. Now using (20) and (21) it can be shown that

$$\left(\bar{\mu}_n^l\right)^2 = \overline{(\mu_n^l)^2} \left(2\gamma\lambda_n^l + \gamma^2 (\bar{\lambda}_n^l)^2\right) \quad (63)$$

$$\begin{aligned} (\lambda_n^{l+1})^2 &= (1-\beta)^2 (\lambda_n^l)^2 + \beta(1-\beta) \bar{\lambda}_n^l \xi_n^l + \\ &\quad \frac{\beta^2}{4} \left[E \left[|e_n^l|^4 \right] \right. \\ &\quad \left. - 2\sigma_{v_n^l}^2 (\xi_n^l + \sigma_{v_n^l}^2) + \sigma_{v_n^l}^4 \right] \end{aligned} \quad (64)$$

(64) can be written compactly as

$$(\lambda_n^{l+1})^2 = (1+\beta)^2 \overline{(\lambda_n^l)^2} + \beta(1-\beta) \bar{\lambda}_n^l \xi_n^l + \frac{\beta}{2} \xi_n^l \sigma_{v_n^l}^2 \quad (65)$$

If we define mean Lagrangian multiplier as $\bar{\lambda}_n^l = E[\lambda_n^l]$, it can be shown that at steady-state

$$\bar{\lambda}_n^l = \frac{\xi_n^{\infty}}{2} \quad (66)$$

Similarly, if $\bar{\mu}_n^l = E[\mu_n^l]$, it can be shown that

$$\bar{\mu}_n^{\infty} = \mu \left(1 + \gamma \frac{\xi_n^{\infty}}{2}\right) \quad (67)$$

(63) can be expressed as

$$\overline{(\mu_n^{\infty})^2} = \mu^2 \left[1 + 2\gamma\lambda_n^{\infty} + \gamma^2 (\lambda_n^{\infty})^2\right] \quad (68)$$

(65) is written as

$$\overline{(\lambda_n^{\infty})^2} = \frac{1}{(2-\beta)} \left[\frac{(1-\beta)}{2} (\xi_n^{\infty})^2 + \beta \xi_n^{\infty} \sigma_{v_n^l}^2 \right] \quad (69)$$

Using (62) with $\Omega = I$,

$$\begin{aligned} 2\xi_n^{\infty} + \gamma(\xi_n^{\infty})^2 &= \mu Tr(\tilde{R}) \xi_n^{\infty} + \mu Tr(\tilde{R}) \sigma_{v_n^l}^2 + \\ &\quad \mu \gamma Tr(\tilde{R}) (\xi_n^{\infty})^2 + \\ &\quad \mu \gamma Tr(\tilde{R}) \sigma_{v_n^l}^2 \xi_n^{\infty} \\ &+ \mu \gamma^2 \left(\frac{1}{2-\beta}\right) \left(\frac{1-\beta}{2}\right) Tr(\tilde{R}) (\xi_n^{\infty})^3 \\ &+ \mu \gamma^2 \left(\frac{1}{2-\beta}\right) \left(\frac{1-\beta}{2}\right) Tr(\tilde{R}) \\ &+ \mu \beta Tr(\tilde{R}) \sigma_{v_n^l}^2 (\xi_n^{\infty})^2 + \\ &\quad \mu \beta Tr(\tilde{R}) \sigma_{v_n^l}^2 \xi_n^{\infty} \end{aligned} \quad (70)$$

(70) is cubic in terms of ξ_n^{∞} and can be expressed as

$$A(\xi_n^{\infty})^3 + B(\xi_n^{\infty})^2 + C\xi_n^{\infty} + D = 0 \quad (71)$$

In 71 above,

$$A = -\mu \gamma^2 Tr(\tilde{R}) \left(\frac{1}{2-\beta}\right) \left(\frac{1-\beta}{2}\right) \quad (72)$$

$$B = \gamma - \mu Tr(\tilde{R}) \left[\gamma + \sigma_{v_n^l}^2 \left\{ \left(\frac{1}{1-\beta}\right) \left(\frac{1-\beta}{2}\right) + \beta \right\} \right] \quad (73)$$

$$C = 2 - \mu Tr(\tilde{R}) \left[1 + (\gamma + \beta \sigma_{v_n^l}^2) \sigma_{v_n^l}^2 \right] \quad (74)$$

And

$$D = -\mu Tr(\tilde{R}) \sigma_{v_n^l}^2 \quad (75)$$

Now assuming that at steady-state, $\mu Tr(\tilde{R}) \ll 1$ [15], it can be shown that at steady-state, the value of ξ_n^{∞} is close to zero meaning that higher powers of ξ_n^{∞} can be ignored. Consequently, (71) will become,

$$C\xi_n^{\infty} + D = 0 \quad (76)$$

or

$$\xi_n^{\infty} = \frac{-D}{C} \quad (77)$$

Hence, steady-state EMSE of the proposed MNCLMS algorithm in the presence of MAI and noise can be shown to be

$$\xi_n^{\infty(MNCLMS)} \approx \frac{\mu Tr(\tilde{R}) \sigma_{v_n^l}^2}{2 - \mu Tr(\tilde{R}) \left[1 + (\gamma + \beta \sigma_{v_n^l}^2) \sigma_{v_n^l}^2 \right]} \quad (78)$$

By using the same procedure developed above, steady-state EMSE of the LMS, NCLMS, ZNCLMS, and MCLMS algorithms in the presence of MAI and noise can be shown, respectively to be

$$\xi_n^{\infty(LMS)} \approx \frac{\mu Tr(\tilde{R}) \sigma_{v_n^l}^2}{2} \quad (79)$$

$$\xi_n^{\infty(NCLMS)} \approx \frac{-(2 + \gamma \sigma_u^2) + 2\mu Tr(\tilde{R}) \sigma_u^2 \left[1 + \gamma \sigma_u^4 + \gamma^2 \left(\frac{1-\beta}{2-\beta}\right) \{\sigma_u^2 + \sigma_u^4\} \right]}{(2 + \gamma \sigma_u^2) - 2\mu Tr(\tilde{R})} \quad (80)$$

$$\xi_{ZNCLMS}^{\infty} \approx \frac{-\left(2 + \gamma\sigma_{v_n}^2\right) + 2\mu Tr(\bar{R})\sigma_{v_n}^2 \left[1 + \gamma\sigma_{v_n}^4 + \gamma^2 \left(\frac{1-\beta}{2-\beta}\right) \left\{\sigma_{v_n}^2 + \sigma_{v_n}^4\right\}\right]}{\left(2 + \gamma\sigma_{v_n}^2\right) - 2\mu Tr(\bar{R})} \quad (81)$$

$$\xi_{MCLMS}^{\infty} \approx \frac{-\left(2 + \gamma\sigma_{v_n}^2\right) + 2\mu Tr(\bar{R})\sigma_{v_n}^2 \left[1 + \gamma\sigma_{v_n}^4 + \gamma^2 \left(\frac{1-\beta}{2-\beta}\right) \left\{\sigma_{v_n}^2 + \sigma_{v_n}^4\right\}\right]}{\left(2 + \gamma\sigma_{v_n}^2\right) - 2\mu Tr(\bar{R})} \quad (82)$$

VII. TRACKING ANALYSIS OF THE MNCLMS ALGORITHMS FOR THE RANDOM WALK CHANNEL IN THE PRESENCE OF MAI AND NOISE

Tracking analysis of an adaptive filter is performed to study its ability to track down the time variations in the channel. This is due to the fact that statistical properties of the weight vector and error signals are able to track the changes in the input signal variation by relying on instantaneous data [9]. In this section, tracking analysis of the proposed algorithm is performed for a random walk model.

A. Random Walk Model

This section deals with the tracking analysis of the MNCLMS algorithm performed for a random walk channel. A general framework for the tracking analysis of adaptive algorithms is used in this section which can handle random system nonstationarities [16]. This framework is based on an energy conservation relation and is valid for adaptive algorithms whose recursion is of the form

$$\mathbf{w}_n^{l+1} = \mathbf{w}_n^l + \mu_n^l D_n^l f(e_n^l) \quad (83)$$

Where $f(e_n^l)$ represents a general scalar function of the output estimation error e_n^l . For an LMS algorithm $f(e_n^l) = e_n^l$.

The random walk model for a channel is given by

$$\mathbf{w}_n^{l+1} = \mathbf{w}_{o,n}^l + q_n^l \quad (84)$$

q_n^l in (84) is assumed to be zero mean, i.i.d, with a positive definite co-variance matrix $E[q_n q_n^T] = Q$ and is also statistically independent of the input regressor and the MAI plus noise, whereas \mathbf{w}_n^o is the unknown system to be tracked. For tracking analysis of an adaptive algorithm, a very important measure is its steady-state tracking EMSE (ξ_n^{∞}) defined as

$$\begin{aligned} \xi_n^{\infty} &= \lim_{l \rightarrow \infty} E \left[|e_{an}^l|^2 \right] \\ &= E \left[\left\| \mathbf{v}_n^l \right\|_{\bar{R}}^2 \right] \end{aligned} \quad (85)$$

Where $\mathbf{v}_n^l = \mathbf{w}_n^o - \mathbf{w}_n^l$ is the weight error vector for the random walk channel.

B. Fundamental Energy Relation for the Random Walk Channel

In this section, the fundamental energy conservation relation [9] is used to develop the framework for the tracking analysis of the proposed MNCLMS algorithm. By using (19) and (84), it can be shown that

$$\mathbf{v}_n^{l+1} = \mathbf{v}_n^l - \mu_n^l e_n^l D_n^l + q \quad (86)$$

Consider the (83), which is given by

$$\mathbf{w}_n^{l+1} = \mathbf{w}_n^l + \mu_n^l D_n^l f(e_n^l) \quad (87)$$

Subtracting both sides of the (87) from $\mathbf{w}_{o,n}^{l+1}$

$$\left(\mathbf{w}_{o,n}^{l+1} - \mathbf{w}_n^{l+1}\right) = \left(\mathbf{w}_{o,n}^{l+1} - \mathbf{w}_n^l\right) - \mu_n^l D_n^l f(e_n^l) \quad (88)$$

In case of an LMS algorithm, $f(e_n^l) = e_n^l$, so (88) becomes

$$\mathbf{w}_{o,n}^{l+1} - \mathbf{w}_n^{l+1} = \left(\mathbf{w}_{o,n}^{l+1} - \mathbf{w}_n^l\right) - \mu_n^l D_n^l e_n^l \quad (89)$$

Now we transpose both sides of (89)

$$\left(\mathbf{w}_{o,n}^{l+1} - \mathbf{w}_n^{l+1}\right)^T = \left(\mathbf{w}_{o,n}^{l+1} - \mathbf{w}_n^l\right)^T - \mu_n^l e_n^l \left(D_n^l\right)^T \quad (90)$$

By multiplying both sides of (90) with D_n^l from left yields

$$\left(\mathbf{w}_{o,n}^{l+1} - \mathbf{w}_n^{l+1}\right)^T D_n^l = \left(\mathbf{w}_{o,n}^{l+1} - \mathbf{w}_n^l\right)^T D_n^l - \mu_n^l e_n^l \left(D_n^l\right)^T D_n^l \quad (91)$$

Equation (91) in terms of a priori error and a posteriori errors can be expressed as

$$e_{p,n}^l = e_{a,n}^l - \mu_n^l e_n^l \left\| D_n^l \right\|^2 \quad (92)$$

For the case when $\Omega = I$, 86 becomes

$$\mathbf{v}_n^{l+1} = \mathbf{v}_n^l - D_n^l \left[\frac{e_{an}^l - e_{pn}^l}{\left\| D_n^l \right\|^2} \right] + q_n^l \quad (93)$$

By evaluating the energies on both sides of (93) We get

$$\left\| \mathbf{v}_n^{l+1} - q_n^l \right\|^2 + \frac{1}{\left\| D_n^l \right\|^2} \left| e_{an}^l \right|^2 = \left\| \mathbf{v}_n^l \right\|^2 + \frac{1}{\left\| D_n^l \right\|^2} \left| e_{pn}^l \right|^2 \quad (94)$$

Since q_n^l is a zero mean stationary random vector and is independent of the input regressor vector and the MAI plus noise, so (94), can be expressed as

$$\left\| \mathbf{v}_n^{l+1} \right\|^2 - \left\| q_n^l \right\|^2 + \frac{1}{\left\| D_n^l \right\|^2} \left| e_{an}^l \right|^2 = \left\| \mathbf{v}_n^l \right\|^2 + \frac{1}{\left\| D_n^l \right\|^2} \left| e_{pn}^l \right|^2 \quad (95)$$

(95) is the random walk tracking model for MNCLMS algorithm. The energy relation is used to evaluate the excess mean-square error at steady state.

TABLE I
COMPUTATIONAL COMPLEXITY PER ITERATION FOR DIFFERENT ALGORITHMS FOR REAL VALUED DATA IN TERMS OF THE REAL MULTIPLICATIONS, REAL ADDITIONS AND REAL DIVISIONS

Algorithm	×	+
LMS	$2L+1$	L
RLS	L^2+5L+1	L^2+3L
MNCLMS[4]	$2L+1$	$2L+6$
MIMO-MNCLMS(LE)	$2ML+8$	$2ML+4$

C. Tracking Steady-State EMSE of the MNCLMS Algorithms

A7): q_n is a zero-mean stationary random vector process with a positive definite covariance matrix Q and is statistically independent of the input regressor vector \mathbf{x}_n^l and MAI plus noise Z_m^l sequence .

Taking expectation on both sides of equation (95) and using equation (44) assumption A7) and the fact that at steady-state, $\mathbf{v}_n^{l+1} = \mathbf{v}_n^l$, the following equation can be obtained

$$2E[\mu_n^\infty \xi_n^\infty] = Tr(Q_n^\infty) + E[(\mu_n^\infty)^2 \|D_n^\infty\|^2 (e_n^\infty)^2] \quad (96)$$

Where $Tr(Q_n^\infty) = [q_n^\infty (q_n^\infty)^T]$. By using equation (96) together with assumption A4 yields

$$2\mu_n^\infty \xi_n^\infty = Tr(Q_n^\infty) + (\mu_n^\infty)^2 Tr(\tilde{R}) (\xi_n^\infty - \sigma_{v_n^l}^2) \quad (97)$$

By substituting the values of $\bar{\mu}_n^\infty$, $\bar{\lambda}_n^\infty$, $\overline{(\mu_n^\infty)^2}$ and $\overline{(\lambda_n^\infty)^2}$ in equation (97), we get the steady-state EMSE of the proposed algorithm.

$$\xi_n^\infty \approx \frac{\mu Tr(\tilde{R}) \sigma_{v_n^l}^2}{2 - \mu Tr(\tilde{R}) [1 + (\gamma + \beta \sigma_{v_n^l}^2) \sigma_{v_n^l}^2]} + Tr\left(\frac{Q_n^\infty}{2\mu}\right) \quad (98)$$

Since q is assumed to be an i.i.d, therefore $Tr(Q_n^\infty) = L\sigma_q^2$, so equation (98) will be reduced to

$$\xi_{n(MNCLMS)}^\infty \approx \frac{\mu Tr(\tilde{R}) \sigma_{v_n^l}^2}{2 - \mu Tr(\tilde{R}) [1 + (\gamma + \beta \sigma_{v_n^l}^2) \sigma_{v_n^l}^2]} + \frac{L\sigma_q^2}{2\mu} \quad (99)$$

Where L is the length of the filter.

VIII. COMPUTATIONAL COMPLEXITY OF LINEAR EQUALIZER

Computational cost is an important aspect of any algorithm. Higher computational cost can render an algorithm useless. A trade off between computational cost and the performance is possible, i.e. if the increased cost results in considerable performance gain, then higher cost can be ignored. In this section, we present the computational costs of few algorithms. As can be seen in table I, computational cost of the proposed algorithm is higher when compared to [4] but MNCLMS is for SISO- CDMA case, whereas, the proposed algorithm is for MIMO CDMA case. This algorithm can be used for run-time applications such as channel estimation, tracking and channel equalization. MIMO systems are being used in modern wireless standards.

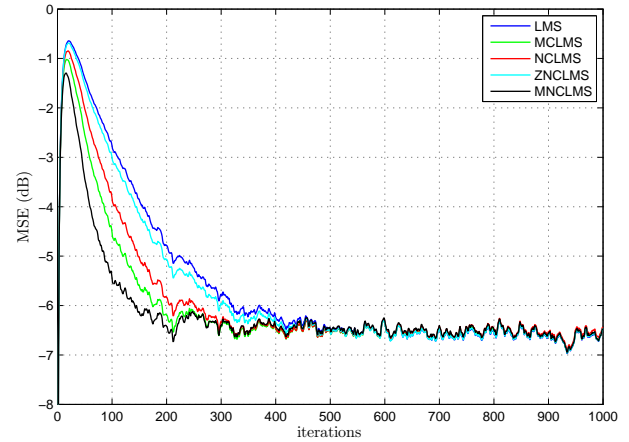


Fig. 1. MSE behavior for different algorithms in an AWGN environment with $K = 10$ at 20dB

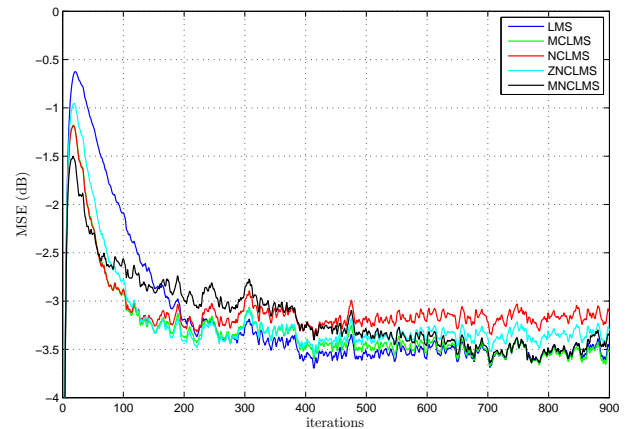


Fig. 2. MSE behavior for different algorithms in an AWGN environment with $K = 20$ at 20dB

IX. SIMULATIONS

Behavior of the step size of the MIMO-CDMA MNCLMS algorithm is shown in figure 3 for 10 subscribers. As can be seen, in the transient state, the MIMO-CDMA MNCLMS algorithm has the largest step-size value when compared to the other algorithms and, thus, yields the fastest convergence. Also, in the steady state, the step-size parameter of the MNCLMS algorithm was reduced to the smallest value amongst all algorithms. Same behavior is achieved for 20 subscribers as shown in figure 4

A. Interference Cancellation in Rayleigh Fading Channel

In this section, simulation results are presented to assess the performance of the MNCLMS algorithm for the MIMO CDMA LE case. The performance of the proposed MIMO-CDMA MNCLMS algorithm is compared with the standard LMS, MCLMS noise constrained LMS and zero noise algorithms. The average MSE is the performance measure through which the algorithms are assessed. A 2×2 MIMO system is

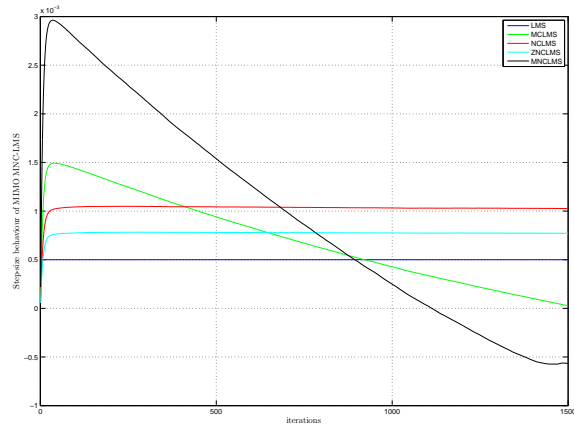


Fig. 3. Behavior of time-varying step size of the MNCLMS algorithm for $K = 10$ at 20 dB SNR

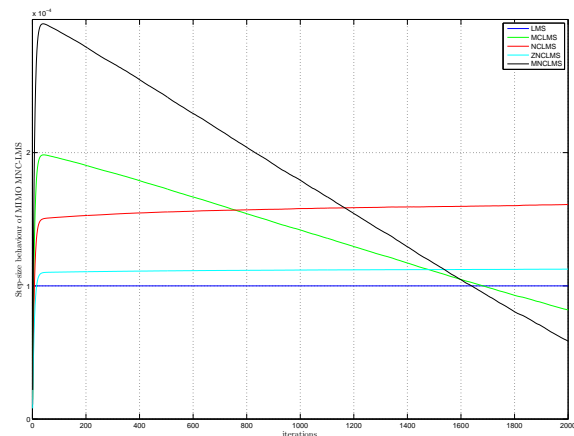


Fig. 4. Behavior of time-varying step size of the MNCLMS algorithm for $K = 25$ at 20 dB SNR

considered here. Random signature sequences of length 31 and rectangular chip waveforms are used. The SNR is kept at 10 dB and 20 dB respectively.

The results of the comparison of the convergence speed of these algorithms for 10 and 25 subscribers, in a Rayleigh fading channel, are depicted, respectively, in Figure.5 and figure 6. In both cases, it can be seen that the MIMO-CDMA MNCLMS algorithm converges faster than the rest of the algorithms. In the case of 10 subscribers, the proposed algorithm was able to achieve an MSE at around -4.8 dB at around 200 iterations. In case of 25 subscribers, the proposed MNCLMS algorithm was able to achieve MSE at -5.2 dB and in around 240 iterations.

X. CONCLUSION

In this paper a new constrained MIMO CDMA LMS type algorithm (MNCLMS) for multiuser wireless environment is presented. The MNCLMS algorithm may be referred to as a generalized constrained adaptive algorithm which includes the MCLMS, the NCLMS and the ZNCLMS algorithms as

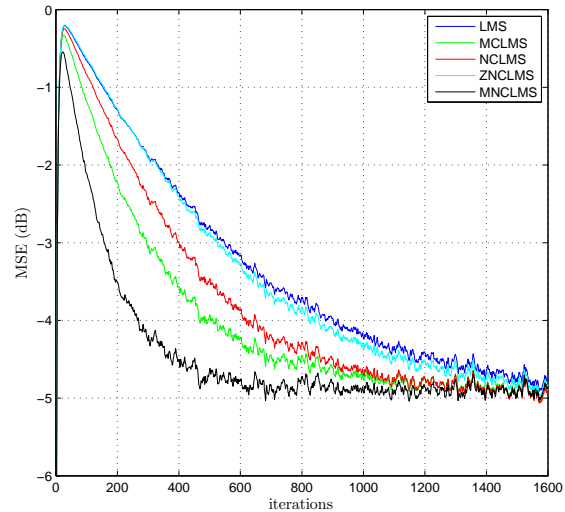


Fig. 5. MSE behavior for different algorithms in Rayleigh Fading environment with for $K = 10$ at 10 dB SNR

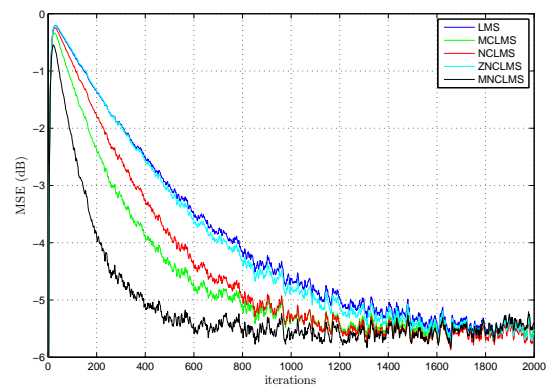


Fig. 6. MSE behavior for different algorithms in Rayleigh Fading environment for $K = 20$ at 20 dB SNR

special cases. In our analysis, we have performed a thorough comparison of the proposed algorithm with a number of other constrained algorithms and it is shown that our algorithm has outperformed its competitors.

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