VoIP QoS Analysis over Asterisk and Axon Servers in LAN Environment

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***Abstract*—Voice over IP (VoIP) is a developing technology and a key factor in both the emerging cyberspace engineering and also an accomplishment to set up its position in the telecom industry. VoIP technology is based on internet technology; where data packets switching system is used rather than circuit switching. Whereas, an analog signal is changed over into the digital signals in the full-duplex transmission. VoIP technology is replaced with the conventional public switched telephone network (PSTN) system due to the high flexibility and low-cost. The purpose of this research work is to deliberate experimental and computational performance in the view of quality of service (QoS) parameters of VoIP over local area network (LAN) network. The VoIP systems implementation is based on two different operating system framework (Linux and Windows), whereas, Linux-based and Windows-based private branch exchanges (PBXs), such as Asterisk (Linux-open source) and Axon (window-close source) are configured, installed and verified. QoS factors (such as packet loss, delay, jitter, etc.) are observed over the Asterisk and Axon PBXs in a LAN domain with the assistance of Paessler switch activity grapher (PRTG) monitoring tool. The validations of results are looked at for QoS parameters crosswise over both PBXs with data load (i.e., file transfer and HTTP traffic) during VoIP calls. The productivity and execution of Axon and Asterisk have been equated and analyzed over experimental based outcomes.**

***Keywords*—VoIP; asterisk; axon; computational based QoS; LAN; packet loss**

I. Introduction

Voice-over-IP (VoIP) is a technology of telecom industry that makes the opportunity to totally modify the conventional telephone communication. VoIP demands are growing rapidly since last decade. The primary thought of this transformation was to accomplish the rise efficacy, versatility, and cost-effective telephony. IP is not a protocol of voice communication, as it was intended to transport the IP data traffic. On the other hand, its worldwide servers, computers, and workstations make its traditional platform and logical for telephony communications.

The primary goal of VoIP is to accomplish better implementation and speedy response time. The tailback is moved from physical connection restriction such as higher data capacity and better throughput to the processing elements of the performance. To assess the performance of the framework distinctive methods should be considered. Performance valuations activities are practice to describe the activities of the framework. The study based performance assessments are useful for both existing and future frameworks. These frameworks are commonly used to improve system performance.

The expression VoIP is utilized as a part of IP communication for dealing with the exchange of voice data (counting fax transmission) in digitalized shape as opposed to in the customary circuit-switching method of the public switched telephone network (PSTN). Conventional telephone facilities exist for over 100 years. Customary telephone framework conveys voice data on two-wire frameworks as per global standard. VoIP telephone is still comparatively new idea [1]. VoIP is the innovation that permits IP based systems to be utilized for voice applications, for example, communication, voice texting, and video chatting. VoIP specialist organizations utilize the Web to convey voice signals from their systems to the client [2]. IP is not a convention for telephonic communication since it was intended to carry data information. Nonetheless, it is generally used in PCs, workstations and server make it sensible and advantageous framework for the help of telephonic communication [3]. VoIP is intended to supplant the inheritance of time division multiplexing (TDM) technology and systems with an IP-oriented data system. Digitized voice will be conveyed in IP packets over LAN/WAN systems [4], [5]. VoIP can not convey images effectively. Therefore, VoIP needs the following protocols such as media gateway control protocol (MGCP), real-time protocol (RTP), resource reservation protocol (RSVP), H.323, and numerous others to give other services for VoIP client [2]. Explicitly, VoIP can be characterized as: The capacity to make phone calls (i.e., to do all that we can do today with the PSTN) and to send likeness over IP-based network with an appropriate quality of services (QoS) and a much prevalent cost-effective solution [6], [7].

In VoIP, the voice stream is separated into data packages and after that, it is compacted, and forward to end-user by different paths, contingent upon the most effective ways suggested by the congested network, and so on. At the opposite end, the data packages are reassembled, decompressed, and changed over once again into a voice stream through different devices and programming components, contingent upon the type of the call and end-user [8].

Professional VoIP alike Skype, WhatsAppp, and IMO includes numerous private branch exchanges (PBXs), Internet, WANs and different elements with colossal networks. The scope of the proposed strategy is confined to LAN environment and that excessive examination and investigation VoIP PBXs such as Asterisk and Axon on various Operating Systems. The testing of QoS utilizing overseeing tolls alike Paessler switch activity grapher (PRTG). The exploration work includes:

- Strategy planning and setup of VoIP system by utilizing Linux and Windows Operating Systems with required necessary equipment's.
- Installation and configuration of VoIP PBX on Asterisk (Linux) and Axon (Windows).
- Call establishing along with HTTP and file transfer load over both (Asterisk and Axon) PBXs to evaluate QoS performance.
- Monitoring and analyzing QoS parameters such as Jitter, packet loss, and packet interruption/delay.

In Section 2, we review the strategic role of VoIP in enterprises. Section 3 addresses the Asterisk & Axon designing and implementation. We describe experimentation and analysis in Section 4. In Sections 5, we summaries the findings and conclusion.

II. The Strategic Role of VoIP in Enterprises

A. Value to Enterprises

VoIP is implementing a strategically starring role cleverly; whereas, VoIP applications will revolutionary change business style such as e-commerce, call center, and email as of now have. VoIP is not only a cost-effective method to run telephone calls. It is a radical new scope of voice traffic that would increase the worth for an enterprise. A significant number of VoIP companies are concentrating on the application development for the VoIP market. It will drive a new regime for new business-changing applications. There are unlimited conceivable outcomes, which well expand on its applications that emphases the boosting VoIP market [9]. Deployment of VoIP applications blast internet business.

A case could be voice-based website empowering, whereas online voice contact and dialog for the customer to perusing for purchasing the products [10]. Instead of detaching the PC link; at that point dialing your telephone number to get more product information. The buyer taps on your client service web-symbol, a VoIP link will establish with call agent, who can talk specifically with the client about products. Furthermore, a call agent can guide more website pages or request forms to the client to finalize the deal on the spot.

B. VoIP Flexibility over PSTN

VoIP can give benefits that are harder to apply over the PSTN network. For example:

- The capacity of VoIP to handle more than one caller over a single line connection without any additional phone lines.
- VoIP provides secure calls protocol, for example, secure real-time transport convention (SRTP). The utmost difficult task over the traditional telephone system is establishing a secure call, for example, digital transmission and digitalizing network. A challenging issue in the traditional telephone system is encoded and verified data stream.
- VoIP support location independence system; just a quick and established internet link is expected to get an association from anyplace to a VoIP supplier.

VoIP can integrate and support various services over the internet, including massage services, video streaming and voice calls over file transferring, managing caller list, audio conferencing, etc. [11].

C. Challenges in VoIP

The IP based networks are inalienably less trustworthy. While, the circuit-switch telephony network does not give a secure mechanism that information is conveyed in consecutive order; in other words, it does not support QoS insurances. However, VoIP executions may confront issues extenuating such as jitter and latency [12].

Voice, text, and video are delivered in packets over IP-based network with constant maximal capacity. Such type of framework is tended to DoS assaults and congested than conventional circuit switching networks. Traditional circuit switch network is lacking the capacity to support more connections, but convey the rest of the information without any delay. The quality of real-time data, for example, voice calls over the IP network reduces the performance significantly.

It is difficult to manage static delay because a data packet comes from different paths. Though, Delay can be minimized by identifying delay sensitivity in voice packets such as “DiffServ”. Constant delays are particularly tricky when coming through satellite communication. It occurs due to the long round-trip propagation delay. That is around 400–600 milliseconds over geostationary satellites [13]. A reason for the congestion is delay and packet losses that can be maintained strategic planning of traffic engineering.

IP packets should be properly reassembled at the receiver side. It might be possible, that receiving information have missed, out of order, delayed packets. Though, it is necessary to make sure that the voice streams keep up an appropriate time consistency. The impacts of jitter can be moderated by voice packets storage at jitter buffer on packets receiving and before transforming into analog signals. In spite of the fact that this may cause further delay in the network. This maintains a strategic distance from a condition well known as underrun buffer. Whereas, the received voice packets won't process until the next voice packet received appropriately. Even though, voice packets are delayed or lost in the VoIP network that causes jitter or voice absence during a voice call.

Multi-path routing has been proposed in VoIP communication in which voice packet is received from different routes [14]. It is effective to avoid packet loss and delay. To improve the VoIP call quality, capillary routing has been suggested in which the raptor codes or particularly fountain codes are used for voice data packets diversity to achieve better reliability in the VoIP networks.

D. Susceptibility to Power Failure

IP Telephones and VoIP phone connectors interface with switches or modems router that commonly relies upon the accessibility of mains [15]. Several VoIP vendors offer customer premises equipment (CPE) e.g., routers/modems with batteries based power supplies to guarantee continuous connectivity up to a few hours. These battery-supported gadgets are normally intended for analog phone sets.

The vulnerability of telephone network to electricity failure is a typical issue, even with a customary analog solution in regions where numerous clients buy the latest handset that have various modern features such as wireless connectivity, voice messaging, telephone directory, etc. Modems are currently accessible with lithium batteries that have better battery backup [16].

E. Emergency Calls

It is hard to find and arrange clients geologically in an IP based network. Thus, an emergency call can not be routed to a close-by call center. Therefore, sometimes VoIP network may divert the emergency calls to another user. A PSTN call has a straight connection between a phone number and a physical zone [17]. A PSTN phone is connected with telephone exchange through a pair of wires that shows the physical connectivity between user and telephone exchange. Once a line is associated with user number, the phone exchange identifies it with the wires, and this relationship will once in a while change. Even if an emergency call originates from that number, at that point the physical location is known to dialer.

IP based communication isn't so straightforward. A wideband supplier may know the area where the wires ended, but still, it does not really permit the mapping of an IP address to that area. IP numbers are frequently dynamically allotted, so, an internet service provider (ISP) may assign an address for online access, or at the time a wideband switch/router is busy [18]. The ISP perceives individual IP number, yet does not really realize what physical area to which it communicates. The wideband suppliers know the physical area, however, is not really following the IP addresses being used.

III. ASTERISK & AXON DESIGNING AND IMPLEMENTATION

The Digium is the main investor for the Asterisks evolving, that is an open code software package, that works as soft-PBX for telecommunication. It is based on the Linux framework rather than a windows-based framework. There are numerous Linux distributors, for example, Ubuntu, Gentoo, Fedora Core, and so forth. The issue with fedora core-based Linux operating system is that the kernel system has been altered, therefore, the drivers for the Digium cards can not be accumulated. In addition, as various modules should be integrated with the system, that slow down the whole system performance even with the fast speedy servers. Visibly, utilizing Fedora core is not an ideal choice for the proposed methodology. Whereas, Ubuntu is developed particularly for Intel-based PCs that makes it speedy operating system than the other Linux flavors. Ubuntu-based Linux framework is available at the Ubuntu Linux site [19], plus the details of installation stages for Ubuntu Linux are accessible at the official site [20]. The accompanying is illustrated the imperative steps of Ubuntu Linux installations:

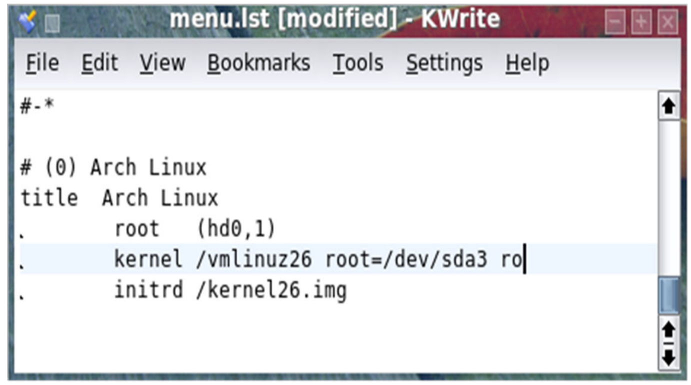

Fig. 1. Commands for manuals installation of Linux.

1) **Hard drive segmentation:** Two segmentations are required to effectively installed Linux operating systems. They are root (/) and swap segments. Where root is utilized to store the files systems in the hierarchal order. Whereas, the swap segmentation is utilized for the extra drive space of the Linux framework. Generally speaking, the swap segment quota should be twice bigger than the aggregate of a physical drive. As we can see from the Fig. 1, the root segmentation (˜/dev/sda3) is made with the portion of 145Gb while the swapped segment (˜/dev/sda1) is 2048Mb. Moreover, the boot segment (˜/dev/sda2) is additionally made, despite the fact that not required, keeping the boot loader documents inside the initial 1024 cylinders of the physical hard disk.
2) **Determination of the software installation bundles:** In this section, we pick the software's that we need to install for Asterisk system. It is not important to install each and every bundle since they can be installed later after the completion of operating system installations.
3) **Kernel installation:** The best decision is to choose the kernel v 2.6 which is generally utilized in Linux operating systems.
4) **In the last, the setup of boot loader for Ubuntu Linux:** The GRUB boot loader is chosen by GRUB config-file that will train the GRUB on how to boot the working operating system.

The vmlinuz26 kernel is used. The root segment is placed at ˜/dev/sda3, or in other words, it is placed at the third partition of the physical memory. In addition, the image file of kernel26 carries the commands for GRUB to load the kernel. It is initially used only at the operating system booting time.

A. Asterisk based PBX Implementation

As it earlier said that Asterisk is a VoIP framework, therefore, we will execute VoIP utilizing intense prospect of Asterisk. Asterisk gives the podium to actualizing the VoIP.

To assemble the Asterisk system, we should induce the GCC compiler (3.x. or updated version). GCC is initially composed for the GNU working framework compiler. Asterisk additionally required the bison, a parser-based encoder package which substitutes yacc and ncurses for command-line interface

(CLI) usefulness. The cryptographical archive in Asterisk needs Open shell, plus it's upgrading bundles.

The Asterisk system is taken from the official site [21]. The installation stages that required to run for Asterisk system are as mentioned below:
Stage 1: Decompressed then start from the source code folder:

```
[root~]#tar -zxvf asterisk-1.4.0-beta3.tar.gz
[root~]#cd -zxvf asterisk-1.4.0
```

Stage 2: Compose, then install the program's package:

```
[root~]#./configue
[root~]#make
[root~]#make install
```

B. Session Initiation Protocol (SIP) Configuration

Soft-telephones are used for the clients at the LAN system. SIP user accounts are established at the Asterisk system. These records are uploaded at /etc/asterisk/sip.conf folder. There are two different records in this document. One account is represented as AXUSR-1 and the second one is signified as AXUSR-2. The mystery factor is the private key (paswd) that the customer is using to verify with the Asterisk system. The network address translation (NAT) will train the Asterisk system that the customers are located inside the LAN network by using NAT. To deal with the call for AXUSR-1; the AXUSR-1_VOIP dial design setting will be utilized. Whereas, the client AXUSR-2 the dial design setting will be AXUSR-2_VOIP. Whereas, the customer PCs are equipped with X-Lite to manage VoIP calls since it utilizes the SIP-based protocol that is maintained through the Asterisk system. The X-Lite software configuration should match the factors of AXUSR-1 at the sip.conf document of the Asterisk system. Especially, The secret key ought to be announced as the mystery in sip.conf record.

C. Phone Extensions Configuration File

The phone dial design settings account for SIP client and analog phones are placed into the /etc/asterisk/extensions.conf document. The phone dial-design for dial-out is utilized to deal with the calls. For instance, Asterisk will manage and onward the call to SIP client account AXUSR-2. When another distinctive number is squeezed, the call will be sent to client AXUSR-1.

D. Windows-based Axon VoIP Platform

Windows-based IP-PBX Axon has the ability to actualize the versatile PBX answer for the VoIP calls. It can deal with the telephone call on IP based systems. It strengthens boundless expansions for the calls and backings up to 64 phone lines simultaneously. Axon is available on the official Axon site. In the wake of downloading, the distinctive extensions are made for the VoIP system. We have established the two extensions in the current situation. The WAXUSR-1 extensions for client 1 and the WAXUSR-2 extensions will be for the client 2. Presently we have created a call over the extensions with the X-lite software on windows working framework.

TABLE I. TYPES OF LOAD & SIZES OVER ASTERISK SYSTEM.

Asterisk	Load Type	Load Size
Experiment 1	File Transfer	800MB
Experiment 2	File Transfer	500MB
Experiment 3	HTTP Load	11759KB
Experiment 4	HTTp Load	2097KB

IV. EXPERIMENTATION AND ANALYSIS

A. Quality of Service (QoS) Parameters Monitored

The accompanying QoS factors are analyzed for both (Axon and Asterisk) PBXs over LAN network with the help of PRTG monitoring tool:

- Packets delay variation
- Maximum jitter
- Packets replicated
- Packets sequences
- Packets losses
- Down-time

B. Paessler Router Traffic Grapher (PRTG)

PRTG is observing software for monitoring, analyzing and mapping the system traffic over the network. It has the ability to analyze and monitor the performance of VoIP data traffic. The ratio of voice calls drop is higher when the user datagram protocol (UDP) packets are not received timely due to the packets disorder and packets losses. PRTG provides the likelihood to act rapidly and keep up a high caliber of administration and furthermore dissect and delineates the distinctive QoS parameters, such as jitter, delay, packets delay, packets loss and so forth.

PRTG gives two approaches to observing and analyzing VoIP execution. we need to include the accompanying sensors modules in the PRTG for analyzing QoS in VoIP.

- Integral QoS estimation sensor
- Cisco IP-SLA sensor

C. QoS Parameters Monitored across Asterisk VoIP Server

To assess the QoS performance distinctive experimentations with different load size and types have been carried by dialing calls over the Asterisk server as shown in Table I. The calls are dial over the customers of the Asterisk servers and after that analyze the QoS factors through PRTG software.

1. File Transfer Loads across Asterisk (800MB & 500MB)

The experiment-1 is carried by dialing a call from the customer AXUSR-1 to customer AXUSR-2 over Asterisk server with the assistance of X-Lite soft-telephone. The steady measurements are observed by the PRTG system for the QoS factors and a call is made during a file transfer over a VoIP network with a fixed load of 800MB. Maximum average jitter is observed around 31ms over the call. The minimum average jitter is found around 2ms. In the event that we consider the entire parametric quantity of jitter factors for each interim of time over the PRTG then we observe that the maximal peak of the jitter stayed 62ms during the call with file transfer load. This 62ms is the pinnacle value as appeared in Table I. The maximal average packet delay is around 236ms and minimal average packets delay is around -117ms. It has likewise been seen thru the experiment that the packets losses are almost 10% for the call time.

Whereas, the experiment-2 is managed with file transfer load 500MB over the VoIP system. Though, the call is produced over the computer for Asterisk server. It evidently recognizes the distinction of QoS parametric factors with various load values. Herein experiment the maximal average jitter is observed around 16ms during call time. The minimal average jitter is observed around 0ms. The maximal packets delay is almost 128ms and minimal average packets delay is -56ms. The packets lost are 0%. So, it has been recognized that the effect of the diverse loads over the system affects the QoS performance.

2. HTTP Loads across Asterisk (11759KB, 2097KB)

Now, experiment-3 is directed with HTTP Load 11759KB by dialing call over the customers of an Asterisk server. It is examined that the maximal average jitter is around 4ms over the call term of time. The minimal average jitter is observed to be 0ms. The maximal average packets delays are 25ms and minimal average packets delays are -22ms. The packets losses are almost 0%.

However, the experiment-4 is implemented with HTTP load 2097KB during the call between the client and the Asterisk server. Though, the maximal average jitter is seen around 6ms. The minimal average jitter is observed around 0ms. The maximal average packets delays are 52ms and minimal average packets delays are around -19ms. The packets losses are almost 1%. The QoS based experimental results are concise in Table II.

D. QoS Parameters Monitored across Axon PBX

In this case, a combination of experiments has been conducted by making calls for windows-based clients over Axon PBX. Furthermore, multiple calls with different load type and magnitudes are dialed between two users and then investigated the QoS factors through PRTG over the Axon PBX as shown in Table IV.

1. File Transfer Loads across Axon(800MB & 500MB)

The experiment-1 is carried by dialing call between the two users (WAXUSR-1 to client WAXUSR-2) over Axon based VoIP-PBX with the assistance of X-Lite soft-telephone. The results are acquired with the help of PRTG soft monitoring tool for the QoS factors by applying constant file transfer load 800MB during the call. The maximal average jitter is estimated around 6ms over the whole call term. The minimal average jitter is observed around 0ms. If we analyze the entire parametric jitter evaluates for each interim of time over the PRTG then we observe that the maximal peak of the jitter stayed 17ms during entire call duration. The minimal average jitter is observed around 0ms. The maximal average packets delay is around 33ms and minimal average packets delay is -30ms. It has likewise been analyzed that the packets loss is 0% for the call span as specified in Table III.

In addition, the experiment-2 is led in a similar way as experiment-1 with 500MB file exchange load of over the network. Therefore, it has been reasoned that the effect of the diverse Loads over the system that affects the QoS esteems. In that experiment, maximal average jitter is observed around 1ms amid the call duration. The minimal average jitter is seen around 0.07ms. The maximal average packets delay is 11ms and minimal average packets delay is -10ms. The packets losses are around 1%.

2. HTTP Loads across Axon(11759KB & 2097KB)

The experiment-3 is conducted with 11759KB HTTP based load. In this experiment, maximal average jitter is around 2ms. The minimal average jitter is observed around 0ms. While, the maximal average packets delay is 10ms and minimal average packets delay variation is -13ms.The packets loss is almost 0%.

While the experiment-4 is applied to with 2097KB HTTP load over the entire call length. Whereas, the maximal average jitter is observed around 1ms. The minimal average jitter is observed around 0ms. The maximal average packets delay is 11ms and minimal average packets delay is -11ms. The packets losses are approximately 0%. The experimental outcomes of QoS over Axon are concise in Table IV.

E. Comparative Study & Analysis of the Results

In the view of the experiments led above for the Asterisk & Axon PBXs to analyze the QoS factors. An analytical investigation is carried out that evidently demonstrate the variation in the QoS performance over the networks.

1. Packets lost in Asterisk and Axon systems with constant file transfer & HTTP loads are presented below in Fig. 2 and 3

A comparison-based analytical study of packets losses over Asterisk and Axon frameworks with file exchange load is demonstrated in Fig. 2. It has been distinctly shown that the packets loss for Axon based systems traces the low limit of percentage as a contrast with the Asterisk call with file transfer load. Thus, the outcome in the terms of packets lost is superior for Axon VoIP framework over the Asterisk system. Moreover, a comparative assessment of both VoIP based PBXs (Asterisk and Axon) frameworks with HTTP load are carried out. Whereas, in Fig. 3, we found that the peak values of packet loss for asterisk reach 10% twice. While it reaches 10% for Axon only once during the entire call duration. Its shows that Axon has better performance than Asterisk with HTTP load.

2. Jitter performance with File Transfer & HTTP Loads for both Asterisk and Axon systems is shown in Fig. 4 & 5

Performance analysis of jitter in Asterisk and Axon framework with constant file load transfer is presented in Fig. 4. It has been evidently revealed from the above graph that the Axon system has less jitter than Asterisk with File Transfer Load. Hence, the outcomes in the term of jitter have a superior

TABLE II. QoS VALUES MONITORED BY PRTG OVER ASTERISK.

Asterisk	Jitter Min	Jitter Average	Jitter Max	Packet Loss	PDV Min	PDV Average	PDV Max
Experiment 1	2ms	8ms	31ms	10%	117ms	0ms	236ms
Experiment 2	0ms	5ms	16ms	0%	56ms	0ms	128ms
Experiment 3	0ms	5ms	4ms	0%	22ms	0ms	25ms
Experiment 4	0ms	3ms	6ms	1%	19ms	0ms	52ms

TABLE III. TYPES OF LOAD AND SIZES OVER AXON SYSTEM.

Axon	Load Type	Load Size
Experiment 1	File Transfer	800MB
Experiment 2	File Transfer	500MB
Experiment 3	HTTP Load	11759KB
Experiment 4	HTTP Load	2097KB

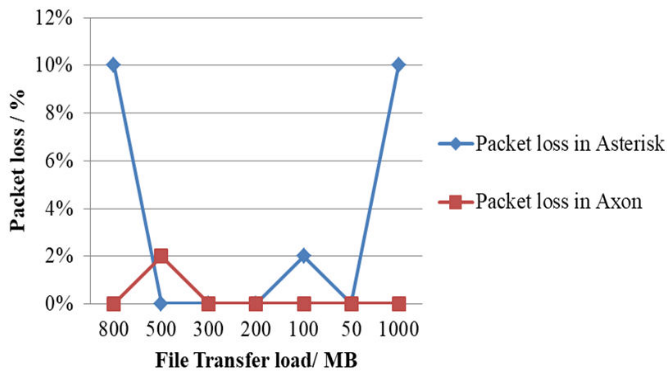

Fig. 2. Packet losses in the Asterisk and Axon with file transfer load.

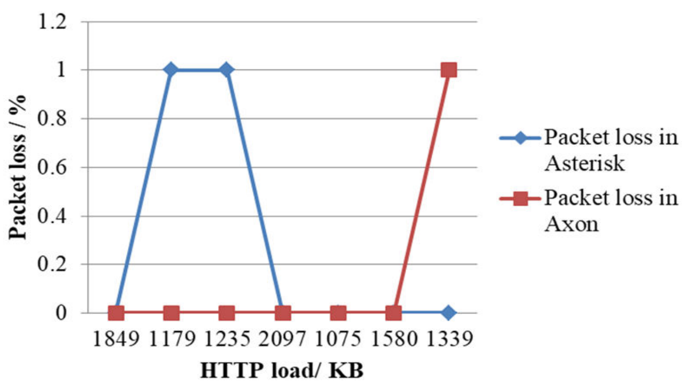

Fig. 3. Packet loss in Asterisk and Axon with HTTP Load

improvement for Axon network than the Asterisks systems. On the other hand, an assessment is made for jitter performance over Asterisk and Axon systems with HTTP load as shown in Fig. 5. It is evidently depicted that the jitter maximal has better values for Asterisk as compare to the Axon system with HTTP loads. The jitter for Asterisk traces to the value of 17ms for an HTTP Load of 2097KB.

3. Packet Delay Variation with File Transfer & HTTP Loads for both Asterisk and Axon systems is shown in Fig. 6 & 7.

Packet Delay Variation performance of Asterisk and Axon systems with file transfer load is shown in Fig. 6. It has been

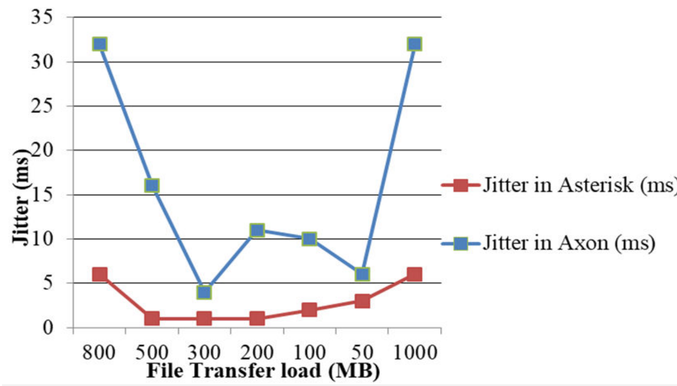

Fig. 4. Average Jitter in Asterisk and Axon with file transfer load

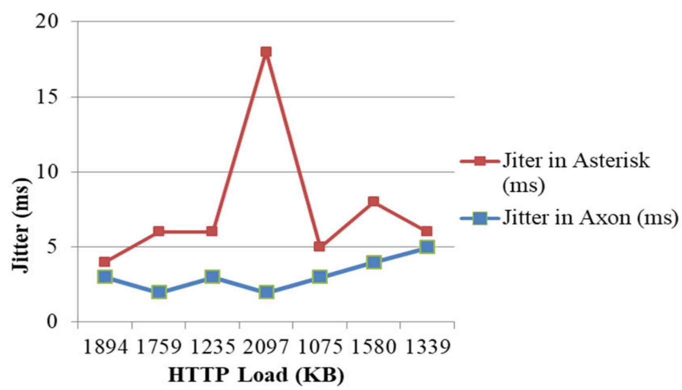

Fig. 5. Jitter in Asterisk and Axon with HTTP Load

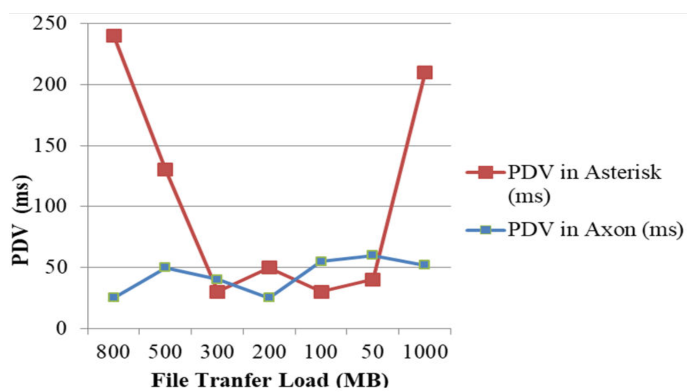

Fig. 6. Packets delay variation in Asterisk and Axon with file transfer load

manifestly shown from the above graph that the Axon system has low-level spikes than Asterisk with File Transfer Load. Therefore, the performance in terms of packet delay deviation is superior for Axon than the Asterisk network. Moreover, valuation is conducted for Packet Delay Variation over Asterisk

TABLE IV. QoS VALUES MONITORED BY PRTG OVER AXON.

Axon	Jitter Min	Jitter Average	Jitter Max	Packet Loss	PDV Min	PDV Average	PDV Max
Experiment 1	0ms	2ms	61ms	0%	30ms	0ms	336ms
Experiment 2	0.07ms	1ms	16ms	1%	10ms	0ms	11ms
Experiment 3	0ms	1ms	2ms	0%	13ms	0ms	10ms
Experiment 4	0ms	1ms	1ms	0%	11ms	0ms	11ms

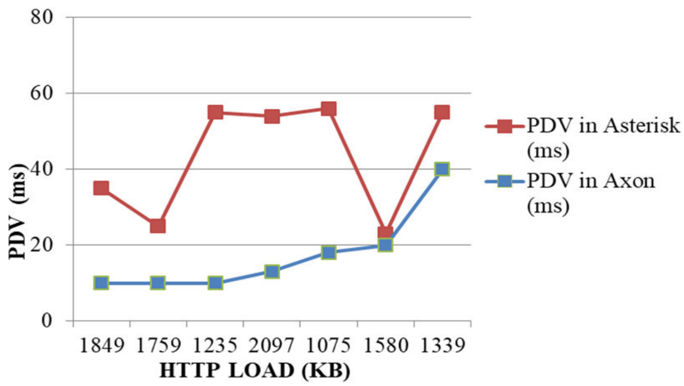

Fig. 7. Packets delay variation in Asterisk and Axon with HTTP load

and Axon systems with HTTP load as shown in Fig. 7. It is basically described that the Packet Delay Variation with HTTP Loads with greater values for Asterisk VoIP system as equating to the Axon system. It's proving that Axon has better performance than the Asterisk system in terms of Packet Delay Variation. Finally, a comparative analytical study described that the Axon based PBXs systems have better outcomes in terms of packet loss, jitter and packet delay variation.

V. Findings and Conclusion

The diverse IP PBXs gives more productive methods for calls steering than the PSTN PBX and with the expanding significance of the IP system. The demand for soft-digitalized PBXs will be expanded. Large numbers of IP based PBXs are available in the market for the call managing. They are giving a decent platform for VoIP networks.

We have analyzed the two versatile PBXs in this research paper. The performance is evaluated in the view of QoS parameters for the VoIP data traffic by producing several experimentations and call processes over a LAN network. As we distinguish that the LAN has an important role in the computer communication system, therefore it has been the emphasis of the overall project to study the VoIP QoS factors over the LAN network.

Several experiments are conducted with a different type (file transfer and HTTP load) and size (800MB, 500MB, 11759KB, and 2097KB) of data load during call over the two different soft PBXs (Asterisk and Axon). It has been summarized that the VoIP network performance of windows-based Axon is vastly improved in the terms of QoS factors as a contrast with the Asterisk VoIP framework in the LAN network.

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