Improved QoS for Multimedia Transmission using Buffer Management in Wireless Sensor Network

Majid Alotaibi
Department of Computer Engineering
College of Computer and Information Systems,
Umm Al Qura University Makkah,
Saudi Arabia

Abstract—**Wireless Sensor Network (WSN) diverts the attention of the research community as it is easy to deploy, self-maintained and does not require predefine infrastructure. These networks are commonly used to broadcast multimedia data from source to destination. However, this kind of data transmission has some challenges, i.e. power and bandwidth limitation with small delay. Art of work mainly focus on optimization either by the shortest route or to minimize the delay by increasing the bandwidth. However, Buffer management is the main constraint to cause delay and loss of packets. In this paper, an approach is presented to manage the buffer and increase the packet delivery ratio (PDR) and reduce delay by assigning the priorities to Intra-coded (I) frame, predictive – coded (P) frame and bidirectional-coded (B) frames dynamically. This approach is very much effective to control the loss of packets in WSN. The presented approach is validated by using Network Simulator 2.**

Keywords—Packet delivery ratio (PDR); multimedia; buffer; priority; delay

I. INTRODUCTION

In the past decade, Internet is most commonly used from education to entertainment. The broadband wireless sensor network (WSN) as well as cellular networks collectively with improved computing power have exposed the entry for an innovative type of utilization to be formed, i.e. audio, video multimedia real time applications. So multimedia transmission needs conventional examines that result in faithful appliances such as surveillance and video conferencing. The Wireless networks give an area to expand these requirements mostly in WSNs to enhance the multimedia communication on these networks. Hence is the necessity of the time to exploit the complete possibilities of such networks, also to understand the reimbursement of multimedia. WSN initiate further challenges for enhanced Quality of Service (QoS), as resource availabilities are less that is channel capacity, buffer capacity and power at every intermediate node. Therefore QoS is viable during a multimedia data transmission session. To present an adequate quality of service in multimedia data transmission appliances, delay and order of packets is the key issue that necessitates to be tackled. i.e., uninterrupted multimedia data packet transmission must run from sender to receiver, and delay, if exist, must be optimized with all the broadcast links. This must be attainable to calculate flow of data packets from every sender in a reasonable approach. As Multimedia data packets are liberal to loss upon certain time limit. So if a few packets get corrupted, multimedia data packets can still be acknowledged. Packet loss mainly occurs in WSNs due to (i)

Route failure, (ii) communication errors and (iii) jamming at nodes which behaves as routers. The loss of data is mainly because of obstruction in the network, transmission errors and failure of the route. The failure of Packets can also happen if the order of data packets depraved due to communication failure. This type of failure can be controlled up to maximum value by the use of Error coding techniques to protect the order of the packet. Failure of the route can be restricted by different techniques like arrangement of the route decision in advance, if failure occurs, or by proper hand off. In real time application, multimedia transmissions usually want QoS guarantees that is huge bandwidth, rigorous delay bound and comparatively error-free multimedia excellence. Multimedia transmission over WSN becomes a demanding because of its dynamic nature as a result the channel is having changeable bandwidths and arbitrary loss of packets. Almost all the existing approaches try to optimize the channel capacity and delay using shortest path algorithm and constant priority approach, very less approaches have been presented that can optimize the buffer, as buffer management is a challenging task in WSN. The time-changing behavior of wireless communication channels and resource allocation made wireless strategy difficult. In this paper an approach called Improved QoS for Multimedia Transmission using Buffer Management is proposed to control the loss of data packets by assigning the dynamic priority to the different data frame packets (i.e. Intra coded (I), Predictive coded (P) and Bi directional coded (B) frames) in the router buffer. This not only minimizes delay but also improves the packet delivery ratio (PDR). The presented approach increases the quality of service (QoS) by assigning the priorities dynamically as per the laxative time of a packet.

The organization of this paper is as follow. Section II presents related work. In Section III we propose our approach, in Section IV we are presenting the simulation results and discussion and conclusion is given in Section V.

II. RELATED WORK

In wireless broadcasting networks, numerous studies have been elucidated for approved Quality of Service on interruption and fairness such as [1] and [2]. But in WSNs number of issues and challenges happened, such as mobility of nodes, available resources and coordination among the nodes. Jacobson [3] presented an approach called TCP algorithm and the presented approach was improved, recognized as Reno TCP approach [4]. Brakmo et al. [5] presented another approach identified as Vegas TCP enhanced packet delivery

ratio than Reno TCP. Both the approaches are the basic descriptions of the Transmitted Control Protocol. However, these approaches cannot achieve QoS in the WSNs with more delay bandwidth product. Both the two approaches are based on the size of the window, such that if congestion happens, it may reduce and if no jam occurs it may boost the window size. The type of system used in these approaches is recognized as Additive Increase Multiplicative Decrease (AIMD). This approach is traditional not planned for huge size of the window to recuperate subsequent to a retreat; the transmission capacity is also not effectively exploited [6]. Numerous options are being recommended to a present TCP within eminent rate networks i.e. [7]-[13]. Even if these approaches have definite advantages more than standard TCP, but, they do not considerably achieve enhanced with respect to standard TCP.

Avrachenkov and Antipolis [14] proposed an efficient method for buffer dimension (Router buffer). But, this method communicates toward a linear permutation of the regular transmitting speed and delay in the line. This approach fails to compute packet loss and usage of the link. Wei et al. [15] presented an approach to control packet loss in networks in which speed is high also the authors showed the problems occurred in the present TCP approach. Sarker and Johansson [16] got optimum results to estimate the packet loss and delay using a system called LTE (Long Term Evolution). However this approach may not be optimum in multimedia transmission, as this approach takes more time to check behavior of the path within the network. Bauer et al. [17] verified that the expansion of these networks results in the growth of ECN. But on the other side, the proposed approach has no consequence (or slightest consequence) on storing the data packets within the buffer the, particularly while the window size is increased [e.g. 18]. Gettys and Gettys [19] proposed the difficulties in buffering management, i.e. a definite approach is used in the data gram switching networks mainly in PSCN. The difficulties associated with this approach are that the overload of data packets reveals in discontinuation and jitter. Jarvinen et al. [20] addressed RED (Random Early Deduction) technique in which the authors proved that it has not more benefits as it is too moderate to tackle rapid change because of slow start TCP, particularly while the data traffic is inadequate. Each and every one of above methods changes the buffer size either by incrementing or decrementing the window size. From the best of my awareness, no approach has given precise calculation of packets lost that is very essential for data transmission (i.e. Multimedia transmission). As the multimedia transmission is allowed to tolerate only up to definite time limit, after this limit care has to take.

Jingyuan Wang et al. [21] presented an approach to manage the packet loss known as TCP-FIT depends on AIMD technique. However the presented approach did not get the accurate window size as desired value.

Ossama Habachi et al. [22] addressed a technique called MOS (Mean Opinion Score). In presented technique, the authors understood that the source node knows the complete information about the data traffic and the current environment of the wireless network. However this approach fails in WSN because of mobility of nodes. Chen et al. [23] presented an approach called CARM (congestion aware routing protocol) to enhance the QoS by controlling the data packets. In this particular protocol, data rate has been used in metric system to avoid disparity. This method fails to give optimum response, when the routes are less from sender to receiver is or if the anchoring node is not available which acts as a router. Parminder Kaur et al. [24] addressed a efficient packet loss control technique. In this approach the authors tried to manage the packet loss by altering the dimensions of the given network, mobility of nodes and power. But this approach may not be optimum when the mobility of node is more and it consumed more energy because of continuous broadcasting of data packets from sender to receiver.

Ksentini et al. [25] addressed an approach using cross layer optimization technique to maintain QoS for video data coding within 802.11e network. However, the proposed model is static so cannot be used to optimize multimedia data transmission. Shin et al. [26] proposed an approach called MPEG-4 algorithm, through a lonely multimedia data packet flow. This method defines three different categories of frames for video transmission represented by *intra coded I*, *Predictive coded P* and *Bidirectional coded B frames* to streamline the compression. In this approach every data frame have given unique type of priority according to the dependency of frame. However this approach fails for multimedia data transmission because it assigns static priorities all the time. Raji and Mohan Kumar [27] proposed a technique that enhances the QoS for multimedia data transmission. However, the presented technique does not guaranteed, when data packets are not acknowledged positively in the destination side.

Lee et al. [28] presented a dynamic algorithm to improve the QoS for multimedia data transmission. However this approach didn't examine the packet time limit of *Predictive coded* frame P and *bidirectional coded* frames B, if *Intra* coded data frame packets are still present within the buffer. Which results loss of *Predictive coded* data packets and reduces the QoS .Because of immense priority of *intra* coded data packets, so *predictive* coded data frame packets are not permitted to move early while their laxative time is over (expiry time of a packet). The presented approach also takes care of each and every acknowledgement which increases delay.

Ding et al. [29] presented an approach to increase QoS based on energy of the node in wireless sensor networks. Li et al. [30] proposed a model to optimize the loss in wireless sensor networks by using cross layer technique. This approach optimizing the loss by allocating the channel based on resource accessibility. However this approach may not be useful, when the routes are less and packet transmission is more from transmitter to receiver.

The presented approach controls the lost packets by assigning priorities to *intra coded*, *predictive coded* and *bidirectional* coded data packets dynamically and also takes care of expiry time of all types of data packets which increases PDR and decreases the delay of packet transmission from sender to receiver.

III. PROPOSED APPROACH

In the proposed approach a control technique is used to streamline the data packets and the source rate control mechanism is employed to reduce the problems associated with the multimedia data over the WSC (wireless sensor channels).

Here the optimization of buffer management in WSN is done by giving the priorities to multimedia data packets. Fig. 1 represents the network buffer.
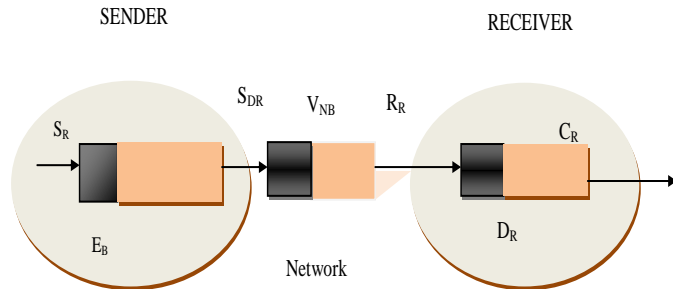

Fig. 1. Network model of buffer.

Where,

E_B - Encoder Buffer; S_R – Source Rate; S_{DR}- Sender Rate,

V_{NB}- Virtual network buffer, R_R – Receiver Rate; D_R – Decoder buffer; C_R - is the consuming rate.
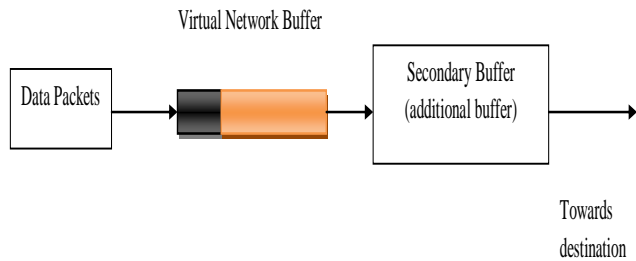

Fig. 2. Buffer management block diagram.

Fig. 2 represents the block diagram of the buffer management in which the data packets are transmitting towards destination via the secondary buffer and the virtual buffer (i.e. primary buffer). Here the secondary buffer is used only to find the status of the loss, and accordingly assign the priorities to the multimedia data packets to increase the QoS. Fig. 3 represents a proposed mapping method called *Priority based* Scheduling *Mapping method* PBSMM. The presented approach dynamically allotted the priority to the different multimedia packets on the bases of the threshold time limit of a packet, particularly called the *laxity time* of a packet. In the presented method, the secondary (additional) buffer is split into four separate buffers called *I-Coded Packets*, *P-Coded packets*, *B- Coded packets* and *Direct Access Based (DAB)* packets. The initial three i.e. I, P, B buffers are used to compress only video information. However the additional buffer i.e. DABS is used to transfer the multimedia data packets towards the subsequent anchoring node/destination

node directly, whose acknowledgements are not received correctly within the given amount of time. The data packets may not be received correctly at the subsequent intermediate node or end node either by jamming or due to noise in communication. In the scheduler mechanism first priority is specified to *I* coded packets, then to *P* coded and finally given to *B coded* data packets. Because *I* coded packet loss have more consequence on multimedia transmission. Conversely, if *B* coded packet is lost, it affects itself only. At this juncture, the buffer *DAB* will resolve the issue of packets whose acknowledgements are not received properly or not reached the end node (destination) prior to the packet expiry time.
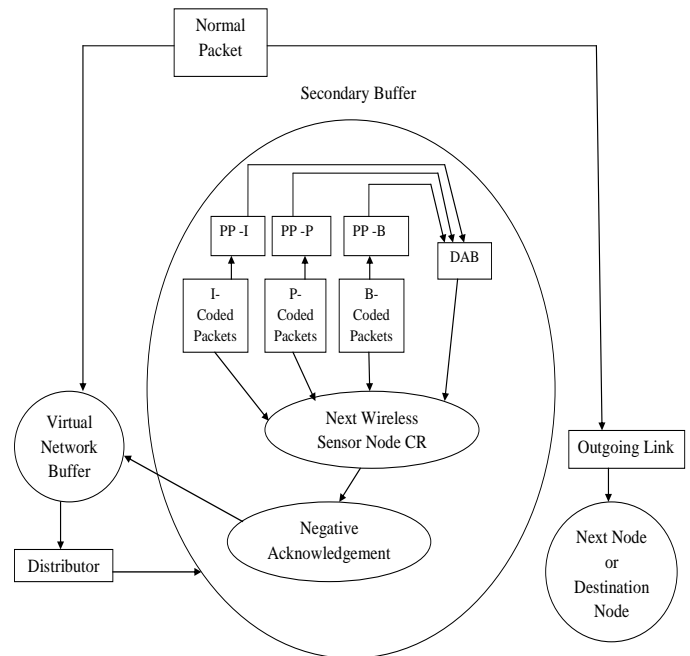

Fig. 3. Priority based scheduling mapping method.

In this approach, a service is being introduced to move the packets ahead called *Packet Forwarding Service* (PFS), so an extra field is inserted in the packet header. An extra field of six bytes space is provided as a part of header to accumulate the hop count and the laxity time. The laxity time can be represented as:

Laxity time = time limit - present time

Where time limit is the time, up to which packet is alive and present time is the time taken by the packet to reach anchor/destination node. Thus, packets whose expiry time is more than the laxity time is directly transfer to the *DAB* by the use of a switching technique from which they can move directly towards the next node or destination node. So, the packet loss and delay are minimized. The packets which are directly going to DAB are marked as Priority Packets (PP-I, PP-P and PP-B). The PPs are those data packets whose entry time is more than the threshold time limit and waste the bandwidth. So to save the bandwidth and delay, a separate threshold time limit is defined for every type of data packet. The *DAB* is used to access the packets which are not received properly by the next sensor node or end node within the allotted time (refer Fig. 4). It transmits intra coded data

packets I and checks the *time Limit* of Intra coded I frame packets i.e. $\mathcal{T}_{LI}$. If the time limit does not exceed $\mathcal{T}_{LI}$, it will forward the packets constantly towards next sensor node or end node via intra coded buffer I. Let the *Time* of a packet (t_p) is equivalent to the time limit of a data packet (time limit for voice data packet is 150 ms and for video it is 400 ms Reddy et al. [31]). Subsequently the data packet is precisely given to the *DAB*, in which it will access accurately to the next node. Likewise, if the time limit of *predictive coded* P and bidirectional coded *B* packets are $\mathcal{T}_{LP}$ and $\mathcal{T}_{LB}$ correspondingly and let this time limit exceeds, then the data packets of these frames will also accessed directly to DAB and can arrive at the end node without any further delay. The DAB buffer is again split into three additional buffers represented as DAB_I, DAB_P and DAB_B to take the packets priority wise which are not being received properly, initially it will consider the intra coded packets through DAB_I, next it will access data packets of predictive coded packets through DAB_P and lastly bidirectional coded via DAB_B.

Fig. 4 represents the flow diagram of PBSMM for successful multimedia type of data transmission. The packets coming from input sensor nodes are directly stored in the buffer as indicated by the scheduler mechanism.
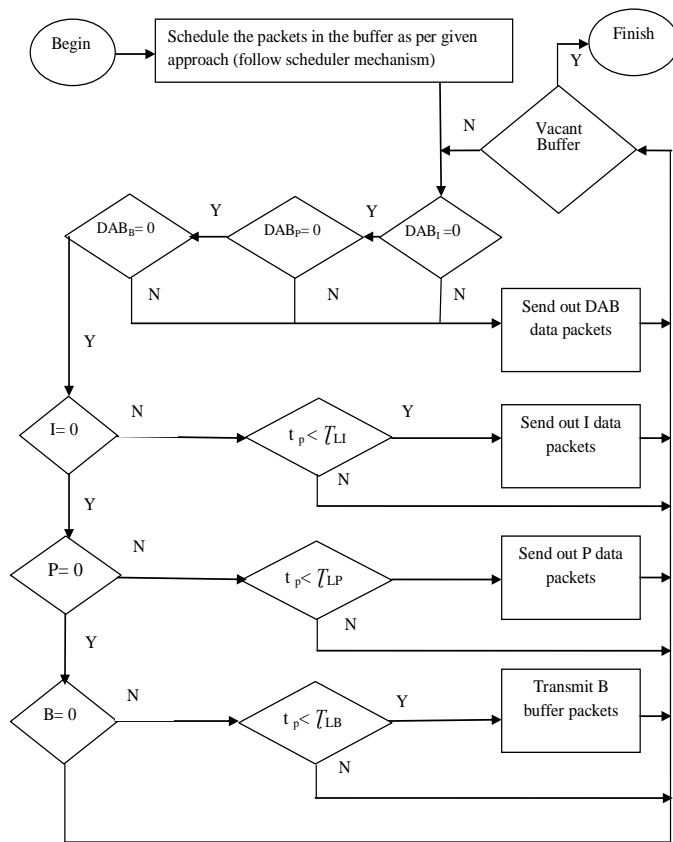

Fig. 4. Flow diagram of PBSMM for multimedia communication.

Whenever DAB is unfilled, it checks intra coded data packets and also it checks the time limit of intra coded data packet. If the time of the packet is not more than the time limit of a packet, subsequently intra coded data packets are accessed to the intra-coded buffer for communication. Similarly, predictive & bidirectional coded data packets are accessed via their particular buffers for communication. If the time of the data packet is equal to time limit of the packet, subsequently the data packets of these frames are accessed to DAB directly for communication.

The presented method is different from Lee approach [28], with respect to the below mentioned facts.

- Secondary buffer is connected with all the Wireless Sensor nodes to assign the priority to data packets dynamically.

- Considering only acknowledgements of data packets which are either acknowledged wrongly or damaged because of noise or due to other network effect.

- One extra buffer (i.e. DAB) is placed beside Intra coded I frame packets, predictive coded P frame packets and bidirectional coded B packet buffers which gives more interest to the time limit of a packet. This also takes care of the data packets that are not received properly before the time limit of a packet.

A brief mathematical modeling is presented to calculate the packet time limit, for this purpose an assumption is made that the transmission of packets between two nodes follow duplex system of communication.

For realistic transmission between end users a router must exist prior to establish a connection among any sensor node via an intermediate node. So jamming may take place at any intermediate node if more packets are entering to the node in association with the leaving ability of the given node. With no loss of generalization, we are using the below mentioned facts.

1) The buffer capability of the router is 'µR' data packets and data packets arriving from input nodes follows average value equal to 'γ' packets/sec. The output capacity of the router buffer is 'δ' packets/sec. such that $\gamma \le \delta$.

2) The secondary (additional buffer) buffer is recommended immediately after the virtual buffer. The additional secondary buffer splits in four associate buffers namely Intra coded, Predictive coded, Bidirectional coded and Direct Access Buffer (DAB) with capacities µI, µB, µP and µDAB correspondingly.

3) The number of regular packets/second transfer to Intra coded, Predictive coded and Bidirectional coded to the subsequent sensor node is δI, δP and δ B correspondingly, and also the subsequent proceeding time is TI, TP and TB correspondingly.

4) In case of non-reception of Ack. (packets not received), the time requisite to transfer once again data packets via 'DAB' buffer is τ DAB , such that $\tau \, DAB \le (t\, p + \frac{RTT}{2})$, RTT→ round trip time and t p is the time to live of a data packet, 'τ DAB'→ sum of entity time of lost data packets.

5) Let on an average the number of packets lost from Intra, Predictive and Bidirectional coded buffer be ф I, ф P and ф B correspondingly.

The data packets, arriving from buffer is identical to the summation of every data packet in the entity buffers in the secondary buffer i.e.

$$\mu_R = \mu_I + \mu_P + \mu_B \qquad (1)$$

μ_R' also can be given as

$$\mu_R = \delta \times \tau \qquad (2)$$

Where 'δ' $\rightarrow$ data rate, and 'τ' is the time to process the packets from intermediate buffer to secondary buffer. Also

$$\mu_R = \mu_{RI} + \mu_{RP} + \mu_{RB} \qquad (3)$$

where μ_{RI}, μ_{RP} and μ_{RB} are the output data rates of Intra coded, Predictive coded and Bidirectional coded buffers correspondingly (when $\mu_{R\,DAB} = 0$ i.e. empty 'DAB' buffer, so 'δ_{DAB}' is zero). The buffer capability is equivalent to rate into time, so 'μ_R' can also be given as:

$$\mu_R = \delta_I T_I + \delta_P T_P + \delta_B T_B \qquad (4)$$

Let the data packets are lost because of delay or due to some other cause with a mean rate of δ_Q and time T_Q. Thus, the total packets lost in all buffers can be represented as 'μ_{RDAB}' and can be shown as:

$$\mu_{RDAB} = \delta_{DABI} T_{DABI} + \delta_{DABP} T_{DABP} + \delta_{DABB} T_{DABB} \qquad (5)$$

Where β_{DABI}, β_{DABP} and β_{DABB} are the data rates of packets lost from Intra, Predictive and Bidirectional buffers correspondingly, T_{DABI}, T_{DABP} and T_{DABB} are the handing out time of lost packets of Intra, Predictive and Bidirectional correspondingly from secondary buffer to next sensor node.

Also 'μ_{RDAB}' can also be given as:

$$\mu_{RDAB} = \phi_I + \phi_P + \phi_B \qquad (6)$$

Where 'μ_{RDAB}' is the sum of all packets lost that flows via 'DAB' buffer to the subsequent sensor node or end node. The data packets coming from Intra, Predictive and Bidirectional packets will loss if,

$$\frac{\mu_{RI}}{\delta_I} > T_I$$

$$\frac{\mu_{RP}}{\delta_P} > T_P$$

$$\frac{\mu_{RB}}{\delta_B} > T_B$$

Or else, the data packets coming from these buffers received effectively at the subsequent sensor node or end node. The replica of the lost data packet be able to accept effectively via 'DAB' buffer, if,

$$T_{DABI} < (T_I + \frac{RTT}{2})$$

$$T_{DABP} < (T_P + \frac{RTT}{2})$$

$$T_{DABB} < (T_B + \frac{RTT}{2})$$

Since the buffer takes data packets through any sub buffer (i.e. intra, predictive & bidirectional), when the packet time fulfills the condition $t_p < T_L$.

IV. SIMULATION RESULTS AND DISCUSSION

In this section, the outcome of the presented approach is compared with the results of Lee and Raji methods. The results are validated by NS-2. Fig. 5 shows the deviation of packet delivery ratio (PDR) against the simulation time of end user pairs.
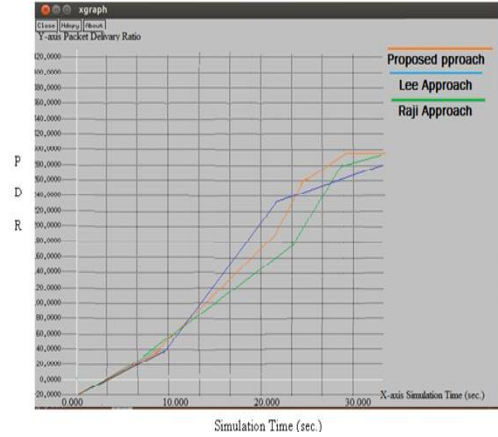

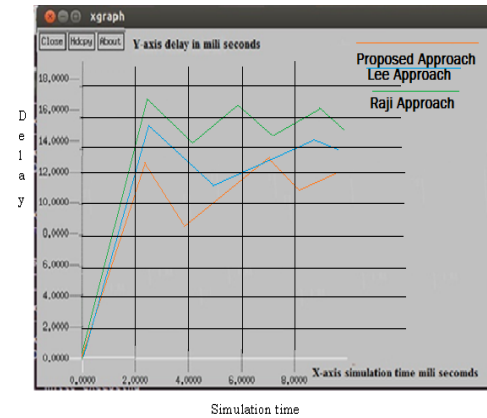Fig. 5. Variation of PDR Vs Simulation Time.

Fig. 6. Variation of Delay Vs Simulation Time.

From the figure it has been observed that if the source and destination nodes are far from each other (i.e. multiple hops), as the time increases, the PDR also increases. The, presented approach generates higher values than Lee approach and Raji approach. Fig. 5 shows that initially the PDR of Lee and Raji model is enhanced when contrast to my model because of sustaining a routing table that shows an original overhead. However, after elapse of time, the planned method has very much higher PDR, once routing table is maintained, the planned method saves maximum 'Intra', 'Predictive' and 'Bidirectional' data packets that are lost in Lee and Raji approach. Fig. 6 represents the deviation of delay versus simulation time in proposed approach and existing approaches.

From Fig. 6 it is observed that the delay in presented approach is less when compared with the existing approaches.

As the number of hops increase, the sensing time of existing approaches will increase and take extra time to sense and transfer the data packets into the channel with more delay.

Table 1 represents the simulation atmosphere and the different framework values calculated.

TABLE I.   SIMULATION PARAMETERS

Parameters	Values
Simulation Time	60 seconds
Number of nodes	1 to 60
silence Time	40 sec.
Medium Accesses Control type	802.11
Network Area	1000m x 1000m
Transmission capacity	4Mbps
Data Traffic	Audio, Video
Area of Network	1000m x 1000m
Simulation Speed	3,7,9,12,14 m/sec
Pause Time	60 sec.
Size of the Packet	512 bytes

V. CONCLUSION

Multimedia data transmission with enhanced QoS can be acquired by manipulating a suitable control strategy for smooth flow of multimedia data packets at the router for a multi-hop link. In this approach the QoS of multimedia data applications in WSNs is enhanced to control the packet loss by exploiting the uniqueness of the multimedia data frame and the packet priorities. Disparate the Raji and Lee approach, the presented approach authenticates the data transmission of all type of multimedia data packets (i.e. Intra, Predictive, and Bidirectional packets) with no increment in the delay. The offered method also maintains the sequence of data packets as per the given time plan of data packets, hence the presented approach is very efficient for multimedia data applications in multi-hop WSNs.

REFERENCES

[1] N. Vaidya, P. Bahl and S. Gupta 'Distributed Fair Scheduling in a Wireless LAN' In Proceedings of ACM MOBICOM 2000, pp. 167–178,2000.

[2] T. Nandagopal, T. Kim, X. Gao and V. Bhargavan 'Achieving MAC Layer Fairness in Wireless Packet Networks' In proceedings of ACM MOBICOM 2000, pp. 87–98,2000.

[3] Jacobson. V, "Congestion avoidance and control", Proc. of ACM SIGCOMM 88, pp. 273- 288,1988.

[4] Jacobson. V, and Berkeley "TCP Evolution from 4.3 Reno", Proc. of the British Columbia Internet Engineering Task Force, 1990.

[5] Brakmmo. L S, S. W. O'Malley and . L. Peterson "TCP Vegas: New techniques for   congestion detection and avoidance", Proc. of ACM SIGCOMM '94, pp. 24-35,1994.

[6] S. Floyd "Highspeed TCP for large congestion windows". Internet draft, draft   –flow-tcp   highspeed-   01.txt.   December. (http://www.icir.org/floyd/hstcp.html), 2003.

[7] D. Katabi, M. Handley, and C. Rohrs "Congestion control for high bandwith-delay product networks", Proc. of ACM SIGCOMM 2002.

[8] T. Kelly "On engineering a stable and scalable TCP variant", Cambridge University Engineering Department Technical Report, CUED/FINFENG/TR 435,2002.

[9] A. Zanella, G. Procissi. M. Gerla, and M. Y.Sanadidi "TCP westwood. Analytic model performance evalution", Proc. of IEEE Globecom,2001.

[10] D. Leith. R. Shorten. and Y.Li "H-TCP: A framework for congestion control in high- speed and long-di.stance networks". HI Technical Report. (http://www.hamilton.ie/net/htcp/),2005.)

[11] Xu, L K. Harfoush, and L. Rhee "Binary increase congestion control for fast long-distance networks", Proc. of IEEE INFOCOM. 2004.

[12] Jin, C D. Wei, S. H. Low, G. Buhrmaster, J. Bunn, D. H. Choe, R. L. A. Cottrell, J. C. Doyle, W. Feng, O. Martin, H. Newman, F. Paganini, S. Ravot, and S Singh "FAST TCP: from theory to experiments", IEEE Network, vol. 19, no. 1, pp. 4-11,2005.

[13] Tan, K J. Song, Q. Zhang, and M. Sridharan "A compound TCP approach for high-speed and long distance networks". In Proceedings of IEEE Intocom,2006.

[14] Avrachenkov,K and S. Antipolis "Optimal choice of the buffer size in the Internet routers", Proc. of the 44th IEEE Conference on Decision and Control, and the European Control   Conference 2005, Seville, Spain,2005.

[15] D.Wei,X C. Jin, S. H. Low, and S. Hegde, FAST TCP: Motivation, Architecture, Algorithms, Performance, IEEE/ACM Transactions on Networking, vol. 14, no. 6,2006.

[16] Z. Sarker and I. Johansson, Improving the Interactive Real Time Video Communication with Network Provided Congestion Notification, IAB / IRTF Workshop on Congestion Control for Interactive Real-Time Communication,Newyork, 2012.

[17] S. Bauer, R. Beverly and A. Berger "Measuring the state of ECN readiness in servers, clients,and routers", Proc. of the 2011 ACM SIGCOMM conference on Internet Measurement Conference (IMC '11). ACM, New York, NY, USA, pp. 171-180.

[18] M. Allman, "Comments on Bufferbloat", In SIGCOMM Computer Communication Rev., vol. 43, no. 1, pp. 30 – 37, 2013.

[19] J. Gettys, and J. Gettys,"Bufferbloat: Dark Buffers in the Internet", IEEE Internet Computing, vol. 15, pp. 95 – 96,2011.

[20] I. Jarvinen, , A. Ding, A. Nyrhinen, and M. Kojo,"Harsh RED: Improving RED for Limited Aggregate Traffic", Proc. of the 26th IEEE International Conference on Advanced Information Networking and Applications (AINA 2012).

[21] Jingyuan Wang, Jiangtao Wen, Jun Zhang and Yuxing Han*"TCP-FIT: An   Improved TCP Congestion Control Algorithm and its Performance"* 978-1-4244-9921-2/11/©2011 IEEE.

[22] OussamaHabachi, Yusuo Hu, Mihaela Van Der Schaar, YezekaelHayel, And Feng Wu *"Mos-Based Congestion Control For Conversational Services In Wireless Environments"* IEEE Journal On Selected Areas In Communications, Vol. 30, No. 7, 2012.

[23] Chen, Hakey, M. Jones, A .D .S Jayalath"Congestion-Aware Routing Protocol for Mobile Ad Hoc Networks*"* 1-4244-0264- 6/07©2012 IEEE.

[24] Parminder Kaur, Ranjit Singh *"A Systematic Approach for Congestion Control in Wireless Ad hoc Network using Opnet*", International Journal of Advanced Research in Computer and Communication Engineering Vol. 2, Issue 3, 2013.

[25] A. Ksentini, M. Naimi and A. Gueroui'Toward an Improvement of H.264 Video Transmission over IEEE 802.11e Through a Cross-layer Architecture'IEEE Communications Magazine, Vol. 44, No. 1, 2006.

[26] P. Shin, S. Lee and K. Chung 'A Cross-Layer based Video Transmission Scheme using Efficient Bandwidth Estimation in IEEE 802.11e EDCA' Journal of KIISE : Computer Systems and Theory, Vol. 35, No. 3,2008.

[27] V. Raji and N. Mohan Kumar (2014) 'An effective stateless QoS routing for multimedia application s in MANETs' International Journal of Wireless and Mobile Computing Volume 7 Issue 5, Pages 456-464 Inderscience publications.

[28] S. Lee and K. Chung 'The Study of Dynamic Video Frame Mapping Scheme for Multimedia Streaming over IEEE 802.11e WLAN'International Journal of Multimedia and Ubiquitous Engineering, Vol. 8, No. 1, pp. 163-174, 2013.

[29] Xu Ding , Lei Shi ,, Jianghong Han and Jingting Lu, 'The Study of Cross-layer Optimization for Wireless Rechargeable Sensor Networks Implemented in Coal Mines' Sensors 171, 16(2) pp 1-24, 2016

[30] Li, C., Wang, J., Li, M., 'An Efficient Cross-Layer Optimization Algorithm for Data Transmission in Wireless Sensor Networks", International Journal of Wireless Information Networks, pp. 1–8, 2017

[31] Reddy T. B., John P john, C. S. R. Murthy 'Providing MAC QoS for Multimedia Traffic in 802.11e based Multi Hop Ad hoc Wireless Networks' Computer Networks, Elsevier, Vol. 51, No. 1, pp. 153-176,2007.