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Improvisation of QoS using DBPQ for Queuing Technique in Vo-MAN Network

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Abstract: Voice over Internet Protocol (VoIP) could present excellent services through Mobile Ad hoc Networks (MANETs) platform. It could cover a lot of purpose scenario assortment from safety to comfort in quality of services, the approach is based Improvisation of QoS using DBPQ for queuing technique in VO-MAN(VOIP+MANET) related services. Queuing has become a popular technology for voice statement from side to side internet. Mobile Adhoc Networks (MANETs) presents a high-quality stage for the fast consumption of VoIP service in many request scenario. MANETs provide a significant difficulty that makes the broadcast of real - time application like VoIP a great challenge due to Quality of Service (QoS) requirements. This paper examines the performance of queuing methods in MANets carrying VoIP traffic. Through a simulation study we investigate and assess QoS indicators such as delay, Jitter, Packet Delivery Ratio (PDR) and throughput. In the existing methods the queuing has been performed based on the analysis report which has its drawbacks with mobile nodes. Hence a QoS based intelligent queue technique called DBPQ (Distance Based Priority Queue) has been proposed to analyze the performance of mobile models illustrate a travel pattern of the nodes. This paper proposes a method to evaluate and implement the DBPQ for the improvement of QoS in VoIP. The main objective of this paper is to allocate the bandwidth as per distance calculation by using RSSI algorithm concept in to undertake a fundamental investigation to enumerate the impact of various queuing mechanism for VoIP (QoS) using DBPQ for queuing technique in Vo-MAN (VoIP+MANet) network

Keywords: VoIP, QoS, RED, RSSI, DBPQ, Vo-MAN

I. INTRODUCTION

The present developed advancement in packet switching networks, VoIP (Voice over Internet Protocol) has become an industry desired technology over Public Switching Telephone Networks (PSTN) for voice communication because of its cheap cost [1] [2] [3]. VoIP allows making international calls during VoIP by only for the data usage acts a huge advantage. Although, there are enormous return to using VoIP there are a number of recognized issue such as, packet loss, jitter and latency moving the Quality of Service (QoS) as the connection is well-known through the internet? This paper analyses how a VoIP network is built (in NS2.35) and testing has been carried out for different queuing techniques to evaluate the QoS metrics. Network Simulator (ns2) is use to run some simulations, we have established for excising queuing method there is some drawback due to defeat those drawback we have proposed a new queuing algorithm is called Distance Based Priority Queue (DBPQ).For this research we are using NS2 to implement our VoIP network on either side of two routers..

QoS VoIP is careful Advantages because of

- utilize ease of use of Internet to make phone calls using normal, readily available computer parts, hence low cost
- VoIP allows to put together different web services like calling a retail organization while from your computer to make product inquiries Due to the vast amounts of codec's available for VoIP networks, voice data may be transmitted at rates dissimilar from the standard 64kbps, including changeable bit rates, hence possibility for greater bandwidth efficiency.

II. RELATED WORKS

Wherever MANET (Mobile Ad-hoc network) are independent networks consisting of two or more mobile nodes ready with wireless communication and hence are self-configuring infrastructure less network. The infrastructure, flexibility and low cost are the main description of MANETs. While in the traditional wireless concept mostly single-hop communication is used (from the base station to the user and vice versa).MANETs allows communication not only between a base station and its users, but also directly between individual users. Hence, within a local region multiple transmissions might take place concurrently.

Distance Based Priority Queuing Protocol

MANET applications are concerned by transmit voice connecting system entity. Applications like tele-emergency system need voice message in remote areas and disaster struck zones which lacks telecommunication infrastructure [4] [5] [6]. Hence DBPQ for voice announcement is needed but however, Multi-hop voice delivery through MANET is challenging, as it must offer QoS provisioning by efficiently treatment limits of node speeds, unreliable connectivity, rapid topology change and a disjointed network. Main confront in designing MANET is to give good delay performance, and handling packet losses [8] [9] [20].

VOMAN Distance Vector Algorithm

VOMAN (VOIP+MANET vector routing) is similar to AODV provides loop free routes even while repairing broken links. The AODV protocol does not need global broken up routing advertisement [11] [17], hence the command on the whole bandwidth available to the mobile nodes is noticeably less than in those protocols that do compel such commercial.

In arrange to make the simulation more sensible as in the real world, the environment traffic is added. The simulation is opening with UDP (User Datagram Protocol), and Transmission control Protocol (TCP). To contain improved sympathetic of the variation among the VoIP on hyper and wireless system, package loss, throughput, delay and jitter are measured and analyzed in sense of VOMAN distance vector algorithm [12] [13].

A clean on order VOMAN route attainment scheme nodes do not lie on active paths and does not retain any routing information and does not participate in any periodic routing table exchanges. The previous node provides its services as an intermediate forwarding station to uphold connectivity connecting two other nodes. DBPQ relies on with dynamism establishing route table entries at intermediate nodes [7] [10].

QUALITY OF SERVICES (QoS)

1. Performance Calculations

THROUGHPUT: Throughput is the standard speed of successful information get ahead of in excess of a communication channel. It is calculated in bytes/sec. For example, from client Node 0 to client Node 1, both clients connect to server Node 1 and Node 2, the throughput of VOMAN refers to the total amount of voice information move between Node 0 and Node 1. Immediate Throughput=bytes (received in description node) over one second.

The immediate throughput will make a graph presentation the amount of in order received by the destination node over each second [16] [19]. This is helpful for assess the immediate things of the environment traffic on the pre-existing VoIP traffic.

The standard throughput will create a single value presentation the average throughput for the whole duration of the simulation. The formula is as following:

Overall Throughput=sum numeral of bytes established in description node.

DELAY: When the number of packets in the buffers grows larger than the buffer sizes results in congestion that creates delay between two communicating nodes. Transmission delay, also known as store and forward delay, is experienced once all packets are received at a router and forwarded to the next router.

Transmission delay = L/R ms

Where, L is the packet length in bits is the link bandwidth between two routers in bits/second.

JITER: Jitter is the time difference between packets arriving at a receiving node. This delay affects voice streams, causing discontinuities between expected arrival times. To compensate for this phenomenon of delay, a payout buffer is used to smooth out the voice stream

III. QUEUING MECHANISMS

Single Server Queue Mechanism:

Throughput for drop tail queue in MANET environment with 20 nodes Queue: The mechanism in which the data correspondence is kept or drop forms the queues. In queuing, a table of the correspondence is required, present should be a method on which an alternative that which packet is to be kept and which is to be drop. Single-sever service node consists of a server plus its queue. The issue related to the efficiency is of great significance. When hundreds and thousands computer are connected to each other, their all-around message is with unexpected outcomes and most of the time, these complexity lead into the weak competence of the system and unknown know what the cause some of the competence problems are created due to excess use of the three resources [14] [26]. If rapidly there is excess traffic through the router, congestion is twisted. Queuing algorithms which are implementing in NS2 software and compared are FIFO, RED, and SFQ, PQ [15] [25].Maintaining the Integrity of the Specifications

IV. QUEUE MANAGEMENT ALGORITHMS

Topology: the proposed method uses peer-peer topology and the topology used in this proposed method shown in figure 1. The results obtained in this topology are listed out in table 1. For calculating the queue length the equation 1 is used.

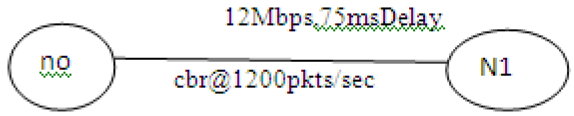

Fig 1: Toplogy Model used in the proposed method

Table 1: Obtained results

Packet Length (Bytes)	**Simulation Time (seconds)**	**Arrival Rate (Mbps) [λ]**	**Service Rate (Mbps)**	**Average Queue Length (Theoretical)**	**Average Queue Length (Result)**
100	4.4	0.96	12	0.0	0.0
400	4.4	3.84	12	0.0	0.0

Queue Length= $((\mu-\lambda)/(\text{packet size}))*\text{simulation time}$ (1)

Since this is a steady bit speed source

- If $\lambda 1 < \mu 2$, following that the queue detachment end to end will be nothing (i.e. queue is unutilized). This is since the packets are approaching at a steady rate (there is no burrstones) and the speed at which the packet are being service is faster than the tempo at which they turn up [18].
- When $\lambda > \mu$, packet carry on to arrive at a steady rate but cannot be serve by the link at that speed. So, more and more packet keeps getting pushed into the queue. Hence, the size of the queue is directly relative to the difference in packet generate and service per unit time. And it also increase with the simulation time [26].

Rebounded FIFO Mechanism: First in First out concept is a simple queue organization algorithm: it sets a predefined value for the utmost length of the queue and when this worth is reached, new packets are surplus, until the next empty cushion space to believe original packets .what time using the First in First Out Tail mechanism, all the packet in the move are treat identically, in spite of the type of transfer which it belongs Packet loss will cause the source to reduce the number of TCP packets sent before receiving the acknowledgment. The throughput of the given TCP assembly will then reduce, until the transmitter start again to be given acknowledgments and begin increasing the size of its jamming window.

Adaptive RED queuing Methodology: RED (Random Early Detection) works by at random (based on certain probability) elimination packets at the nodes of the network, before the incidence of obstruction, when the average queue length exceed the predefined minimum threshold. When the standard queue length exceeds the maximum threshold, the chance of rejection become equal to 1. RED monitors the average length of the queues by removal or ECN-marking packets based on statistical probability. If the barrier is nearly vacant, all incoming packets are received. As there is increase in use, the likelihood of removal lately arrived packet also increases. When the cushion is demanding, all received packet are deleted. RED has no QoS separation in the basic version. The versions WRED (Weighted RED) and RIO (RED with In and Out),

Modified STOCHASTIC FAIR QUEUING (SFQ): It is customized type of FQ by the aim of remove its limits as this method reduce the number of necessary queues. One of the most significant drawbacks of this technique is unfair performance with the flows collide with other flows. Thus, as the name reveal, fair is guaranteed as stochastically [15] [25]. It is suitable for employ in elevated pace processor system that covers a wide variety of CPU, memory and justice trade-offs. It offers elegant degradation under overload and sudden failure [27] [28].

The comparative study of the three queuing algorithms is listed out in table 2.

Table 2: Comparison of queuing algorithms

Queuing algorithms used	Traffic patterns	Parameters	Conclusion
RED &Drop tail	TCP	Packet drop	RED is better than Drop tail and DFT in dropping more number of packets.
CBQ, SFQ, DRR, FQ, RED, Droptail	TCP UDP	Packet drop	FQ and SFQ are best as they deliver maximum number of packets, there is no change in case of DRR, Droptail and RED drop max. Packets
Droptail, FQ, SFQ, DRR, RED	TCP UDP	Buffer size Attack intensities	Droptail and RED gives best performance for buffer size 60. FQ, SFQ and DRR are not affected. SFQ is best against attack intensities for buffer size 60.
Droptail, RED, SFQ, FQ, DRR	CBR FTP	Throughput End-to-end	FQ gives maximum throughput and RED is worst. FQ gives maximum delay, RED gives

QUALITY OF SERVICES PARAMETERS

1 Throughput: Throughput is a measure of the date rate (bits per second) generated by the application and is equal to the total data transferred divided by the total time it took for the transfer [6] [8]. Theoretical value of throughput for fixed 75 Mbps per channel (in a 20 MHz channel using 64QAM ¾ code rate) and its practical value is around 45 Mbps/channel. The results of the through put is shown in table 3.

2 Minimum Average Delay or latency: Average Delay measures the average one way latency observed between transmitting an event and receiving it at each sink node. Stoppage is the occasion taken by the packet to crossways from the basis to the purpose. The main source of delay can be further categorized as source-processing delay, propagation delay, network delay and purpose processing delay [17].

3 Various Jitter or Delay variation: Jitter is distinct as an arithmetical difference of the RTP data packet inter-arrival time. It is the difference in the time among packet incoming. In the Real Time Protocol, jitter is calculated in timestamp units [23] [24]. For instance, if you broadcast audio sampled at the standard 8000 Hertz, the unit is 1/8000 of a second. Delay difference is the difference in the holdup introduce by the components along the message path. Jitter is commonly used as an pointer of constancy and stability of a network [29].

Table 3: Throughput results in the proposed method

No. of Nodes	G.711	G.723	G.729
2	125.54	25.18	45.65
4	220.31	46.51	80.11
6	357.79	69.63	129.56

This kind of behavior is observed because when the number of nodes increases, the number of packets being transmitted also increases. G.711 codec has the highest throughput as a result of highest packet size. The next higher throughput is for G.729. The G723 codec has the lowest throughput [21] [22]. The simulation for average jitter is shown in figure 2 and graphical representation of the throughput is shown in figure 3. The graphical representation of jitter results when it applies on N number of nodes is shown in figure 4.

No. of Nodes	G.711	G.723	G.729
2	0.00000280511761838571	0.00000000041106273204	0.00000280511761838571
4	0.00002307936990258500	0.00000000067609654823	0.00000639130887373232
6	0.00005893413528256280	0.00000000082569284516	0.00001086508500230500

Fig. 2: simulation for average jitter results

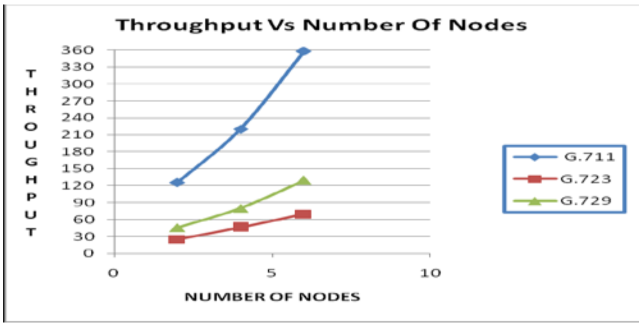

Fig. 3: throughput results when applies on number of nodes.

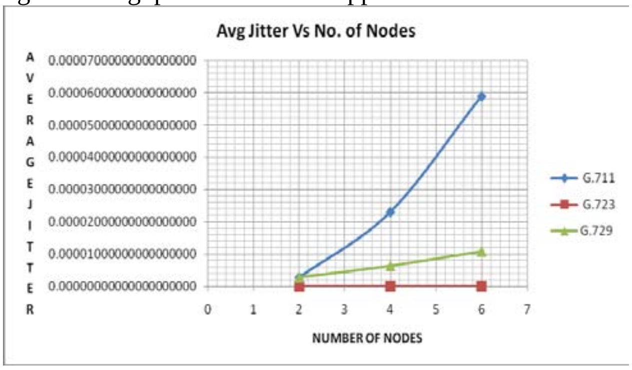

Fig. 4: jitter results when applies on number of nodes

Quality of Service (QoS) parameters is observed for VoIPv4 and VoIPv6.

DISTANCE BASED PRIORITY QUEUE DESCRIPTION

It can be used in an environment of the Tele-emergency cases. This Queue consists of two stages. In first stage, priority level decision depends on priority levels and in second stage, the Distance Metrics Calculations are taken place in which the bandwidth will be allocated adaptively according to the distance parameter, ie .. Bandwidth will be allocated with the long and short distance respectively.

Distance Based Priority Queue: The distance based priority queue is shown in figure 5

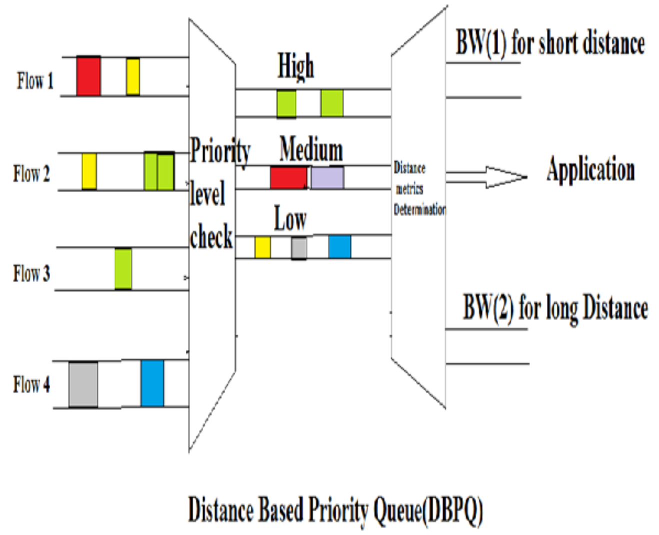

Fig. 5: Distance Based priority queue

The advantages of DBPQ are Efficient queuing Techniques for handling the dates in Tele-Emergency Cases and High Throughput. But DBPQ has some limitations. Those are Time Delay for the Bandwidth Allocation, Application and Used in real time application as VoIP. The flowchart of DBPQ is shown in figure 6.

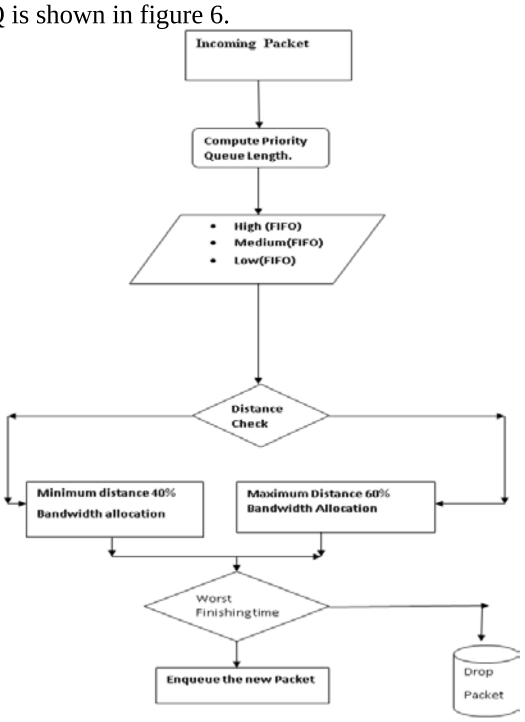

Fig. 6: Flow chart of the DBPQ

DBPQ Implementation with RSSI

Mathematical derivation for Distance Based Priority Queue

To calculate the distance using RSSI(Received Signal Strength Indication)To determine the distance between transmitter and a receiver node which is short and long to allocate the bandwidth as per priority and distance.

RSSI is generic radio receiver technology metric.

The distance using RSSI can be calculated using the FRIIS transmission formula

$$P_r = \frac{P_t G_t G_r \lambda^2}{(4\pi d)^2 L} \rightarrow\rightarrow\rightarrow\rightarrow\rightarrow eqn \dots\dots(3)$$

Where

$P_r: Receiving\ Power, P_t: Transmitting\ Power,$

$G_t: Gain\ of\ Transmitting\ antenna\ (to\ find\ distance)$

$G_r: Gain\ of\ receiving\ antenna\ , \lambda: Wavelength,$

$L: Packet\ loss$ and $d: Distance\ between\ node$

i.e., To calculate distance between two nodes by receiving signal using equation 4.

$$d = \sqrt{\frac{P_t G_t G_r \lambda^2}{(4\pi)^2 L P_r}} \rightarrow\rightarrow\rightarrow\rightarrow\rightarrow eqn \dots\dots(6)$$

Where Node A and node B are considered to be in communication of MANET. Let us assume i as the centre of coordination system

$N: Set\ of\ nodes\ in\ network$

$P_i: Set\ of\ one\ hop\ neighbor\ of\ node\ i$

$D_{ij}: Distance\ between\ node\ i\ and\ j$

$D_i: Overall\ distance\ D_{ij}$

To classify the conditions of positioning as

High

Medium

Low

- d< : Bandwidth allocation is 40%
- d> : Bandwidth allocation is 60%

$i \in P_a ; i \neq b, i \neq a$

i defines its system of local coordinates

x-axis line(ip)

y-axis line(iq)

nodei is centre of the system

ix=0, iy=0

P is the node axis of

$P_x = d_{ip}, \quad P_y = 0$

q is locater coordinate axis

$q_x = d_{iq} \cos\alpha\ P_y = d_{iq} \sin\alpha \rightarrow\rightarrow\rightarrow eqn \dots(3)$

where, $\alpha = P_{iq}$

Using the theorem of "Al-Kashi"

The reference system of the proposed method is shown in figure 7 and The system of localization local diagram is shown in figure 8.

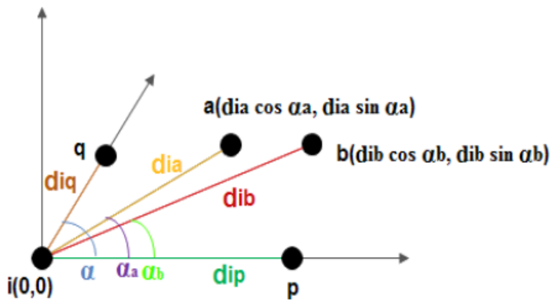

Fig. 7: Reference diagram of the proposed method

$$d_{pq}^{\ 2} = d_{ip}^{\ 2} + d_{iq}^{\ 2} - 2 d_{ip} d_{iq} \cos\alpha$$

α is calculated using the following formula

$$\alpha = ar\ \cos\left(\frac{d_{ip}{}^2 + d_{iq}{}^2 - d_{pq}{}^2}{2\, d_{ip}\, d_{iq}}\right) \to\to\to eqn \ldots (4)$$

The calculation coordinates P_i is

$$a\in\ P_i \begin{cases} ax = d_{ia} \cos\ (\alpha a) \\ ay = d_{ia} \sin\ (\alpha a) \end{cases}$$

$$\alpha a = a_i p = a_r \cos\left(\frac{d_{ip}{}^2 + d_{ia}{}^2 - d_{pa}{}^2}{2\, d_{ip}\, d_{ia}}\right) \to\to \to eqn \ldots (5)$$

For b, coordinates

$$b\in\ P_a \begin{cases} bx = d_{ib} \cos\ (\alpha b) \\ d_{ib} = d_{ia} + d_{ab} \\ by = d_{ia} \sin\ (\alpha a) \end{cases}$$

$$\alpha b = b_i p = a_r \cos\left(\frac{d_{ip}{}^2 + d_{ib}{}^2 - d_{pb}{}^2}{2\, d_{ip}\, d_{ib}}\right) \to\to \to eqn \ldots (6)$$

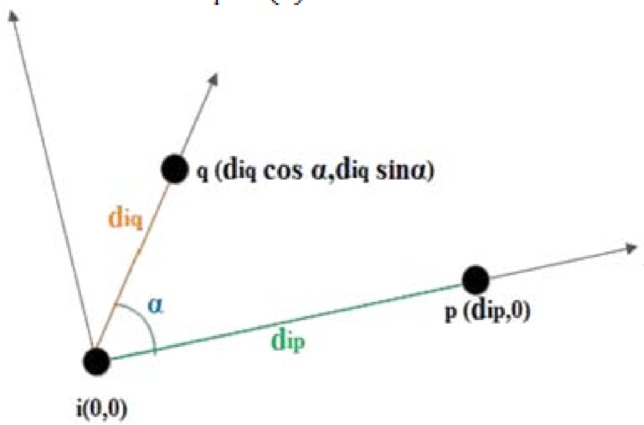

Fig. 8: The system of localization local

The results of the RREQ(Rout Request), RREQ(Rout reply) is listed out in table 4

Table 4:Results of the RREQ

RREQ's source	RREQ's destination	Time	Distance	Distance/ Pr
7	0	0,00128055	165,397	317,654
8	0	0,00128093	277,895	897,138
3	0	0,001281	299,47	1041,43
17	0	0,00128117	351,078	1430,93
15	0	0,00128119	356,321	1474,36
19	0	0,00128119	358,238	1490,13
0	8	0,00301585	277,731	896,079

PERFORMANCE ANALYSIS: The performace Analysis of the proposed method for dalay is shown in figure 9 and the performace analysis of the proposed method for jitter in shown in figure 10, the performce analysis of the proposed method when packet loss is shown in figure 11 and the performce analysis of the proposed method of throughput is shown in figure 12

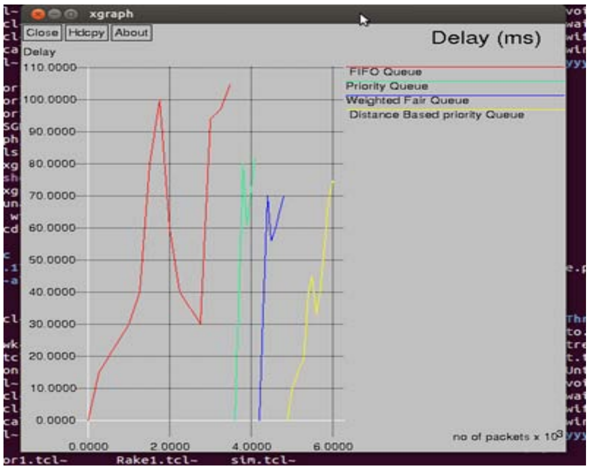

Fig. 9: performace Analysis of the proposed method for dalay

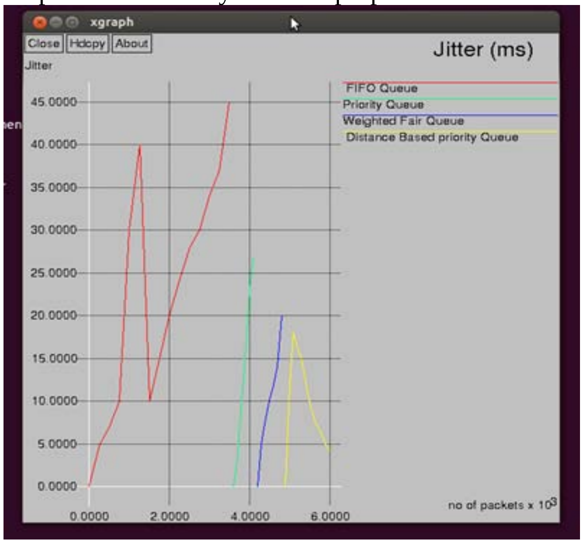

Fig. 10: performace Analysis of the proposed method for jitter

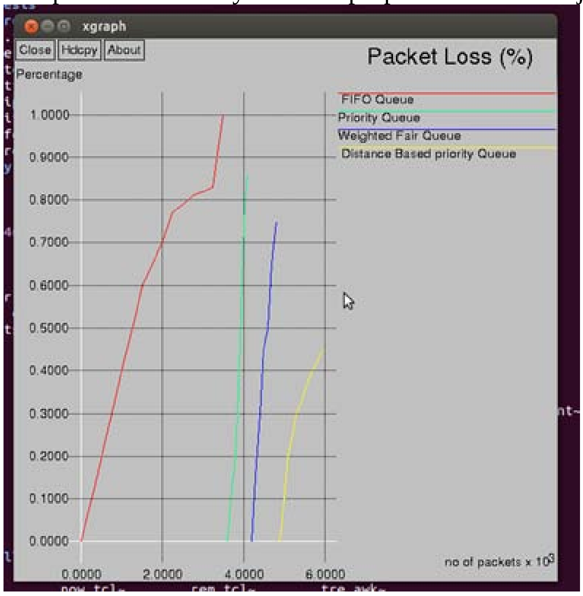

Fig. 11: performace Analysis of the proposed method for packet loss

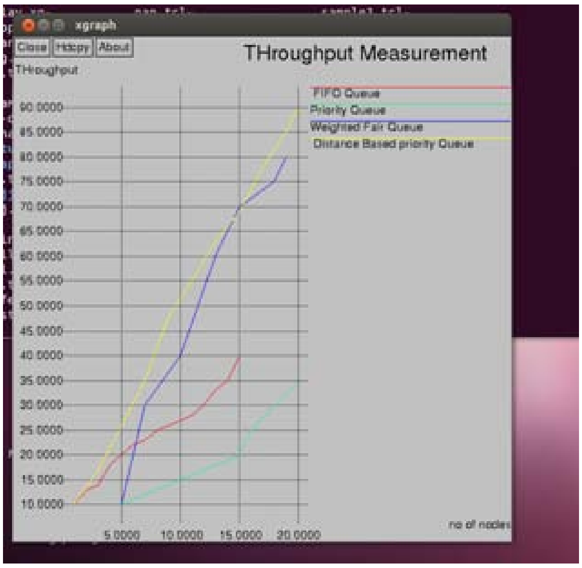

Fig. 12: performace Analysis of the proposed method for throughput

V. CONCLUSION

In this proposed DBPQ queue technique has been implemented using RSSI algorithm. The performance evolution has been made with respect to QoS performance metrics such as delay, throughput packet loss, and jitter. The performance analyses were conducted comparing with FIFO, PQ, WFQ,SFQ. According to simulation result DBPQ gives better result in case of throughput, jitter, and packet loss. As a feature work the node energy level should be tested and increased to avoid delay performance. In the result scenario Improvisation of QoS using DBPQ for queuing technique in Vo-MAN network has been simulated in network simulator

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