Indian Journal of Science and Technology, *Vol 9(30), DOI: 10.17485/ijst/2016/v9i30/95653, August 2016* *ISSN (Print) : 0974-6846* *ISSN (Online) : 0974-5645*

A Novel Approach of DBPQ with RSSI Queuing Technique for VoIP QoS over MANET

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Abstract

Objectives: 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 of safety to comfort in the quality of services, the approach is based Improvisation of QoS using DBPQ for queuing technique in VO-MAN (VOIP+MANET) network. **Methods/Statistical Analysis:** Queuing has become a popular methodology for voice statement from side to side internet scenario. Mobile Ad hoc Networks (MANETs) presents a high-quality stage for the fast consumption of VoIP service in many applications scenario. Through a simulation study, we investigate and assess QoS parameters 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. **Findings:** This paper proposes a method to assess and realize the DBPQ for the development 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 mechanisms for VoIP (QoS) using DBPQ for queuing technique in Vo-MAN (VOIP+MANET) network. **Applications/Improvements:** The proposed system achieves significantly enhanced performance in network simulation than the existing systems by utilize better bandwidth allocation and high throughput access.

Keywords: VoIP, QoS, RED, RSSI, DBPQ, Vo-MAN, Mobile Ad Hoc Network

1. Introduction

The present developed advancement in packet switching networks, VoIP (Voice over Internet Protocol) a become an industry desired technology over Public Switching Telephone Networks (PSTN) for voice communication because of its cheap cost[1,2]. VoIP allows making international calls during VoIP by only for the data usage acts a huge advantage. Although, on that point is an 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 simulation, we have established for excising queuing technique 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 system on either side of two routers[3].

MANET (Mobile Ad-hoc network) is self-governing networks consisting of two or more movable nodes ready with wireless message and hence is self-configuring communications less network. The infrastructure, flexibility and low cost are the main description of MANETs[4]. While in the traditional wireless concept mostly single-hop communication is used MANETs allows communication not only between a base station

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and its users, but also directly between each user. Hence, within a local region multiple transmissions might take place concurrently[5].

1.1 Proposed Contributions

- MANET applications are concerned with transmitting voice connecting system entity.
- Applications like tele-emergency system need voice message in remote areas, which lacks telecommunication infrastructure[6].
- Figure 1 shows the flow chart analysis in Voice release from side to side MANET is demanding, 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 present good delay performance, and handling packet losses[7].

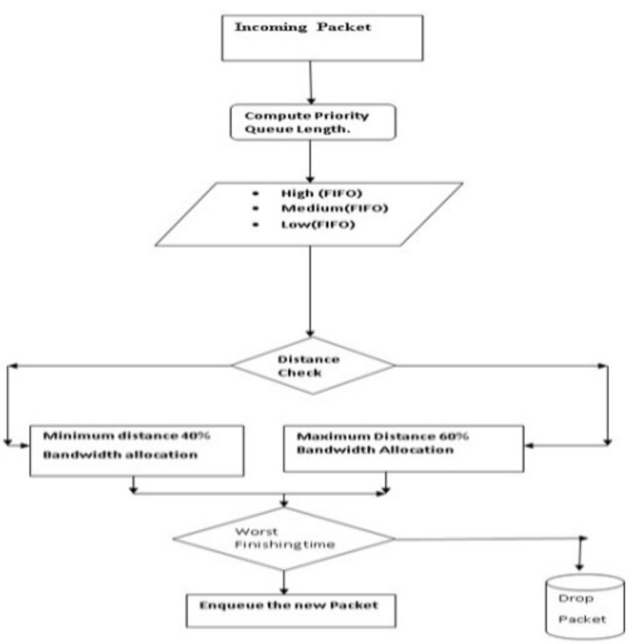

Figure 1. Flow Chart.

1.2 Overview of the proposed 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 hence the command on the whole bandwidth available to the mobile nodes is noticeably less than in those protocols that do compel such commercial[8,9].

In setting 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[10].

The previous node provides its services an intermediate forwarding station to uphold connectivity connecting two other nodes. The distance calculated between node to node using RSSI mechanisms to allocate certain bandwidth level by using DBPQ relies on with dynamism establishing route table entries at intermediate nodes[10].

2. Implementation and Analysis the (QoS) Parameters

2.1 Performance Calculations throughput

Figure 2 shows the throughput is the standard speed of successful information get ahead of in excess of a communication channel. It is counted 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 quality of voice information move between Node 0 and Node 1. Immediate Throughput=bytes (received in description node) over one second[11,12].

The immediate throughput will make a graph presentation the amount of in order received by the destination node over each second. 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[13]. Overall Throughput = sum numeral of bytes established in description node.

2.2 Performance Calculations 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. Figure 3 shows the 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.

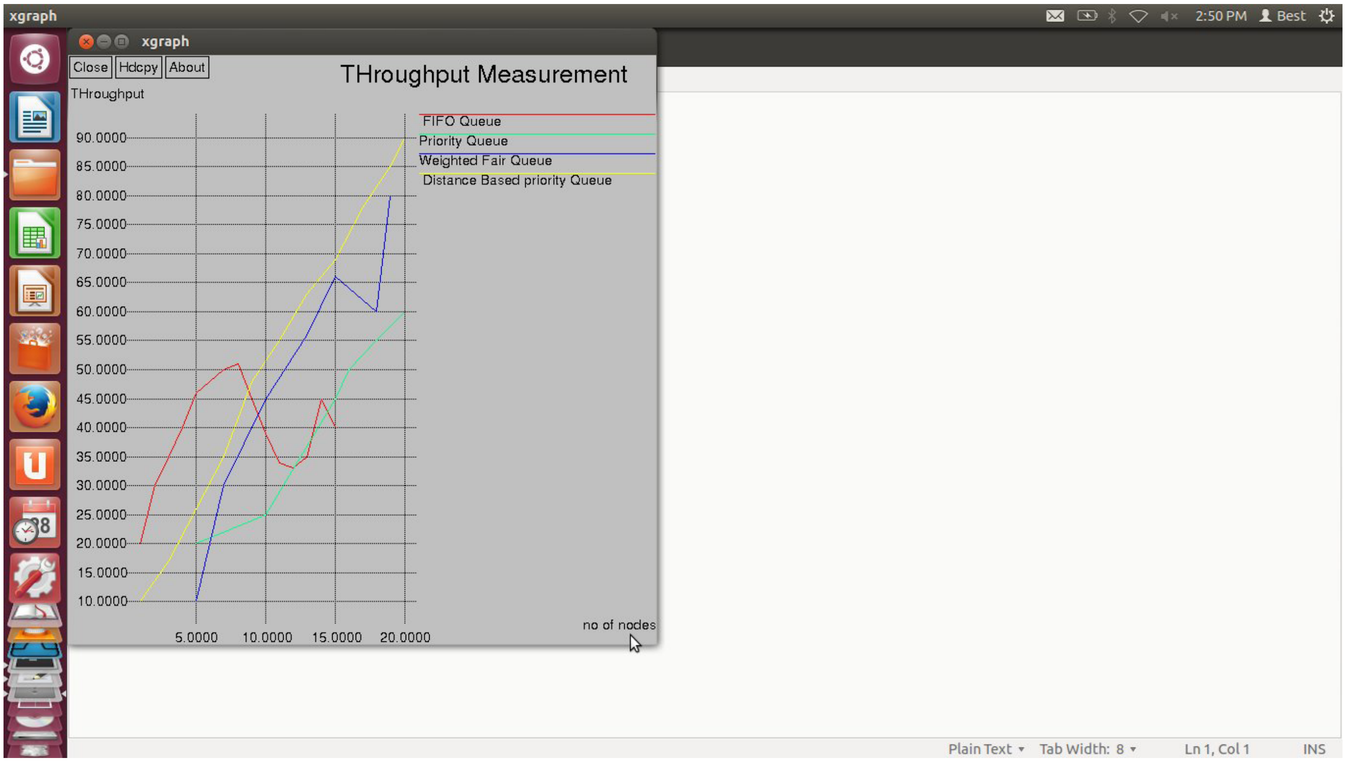

Figure 2. Performance Analysis of Throughput.

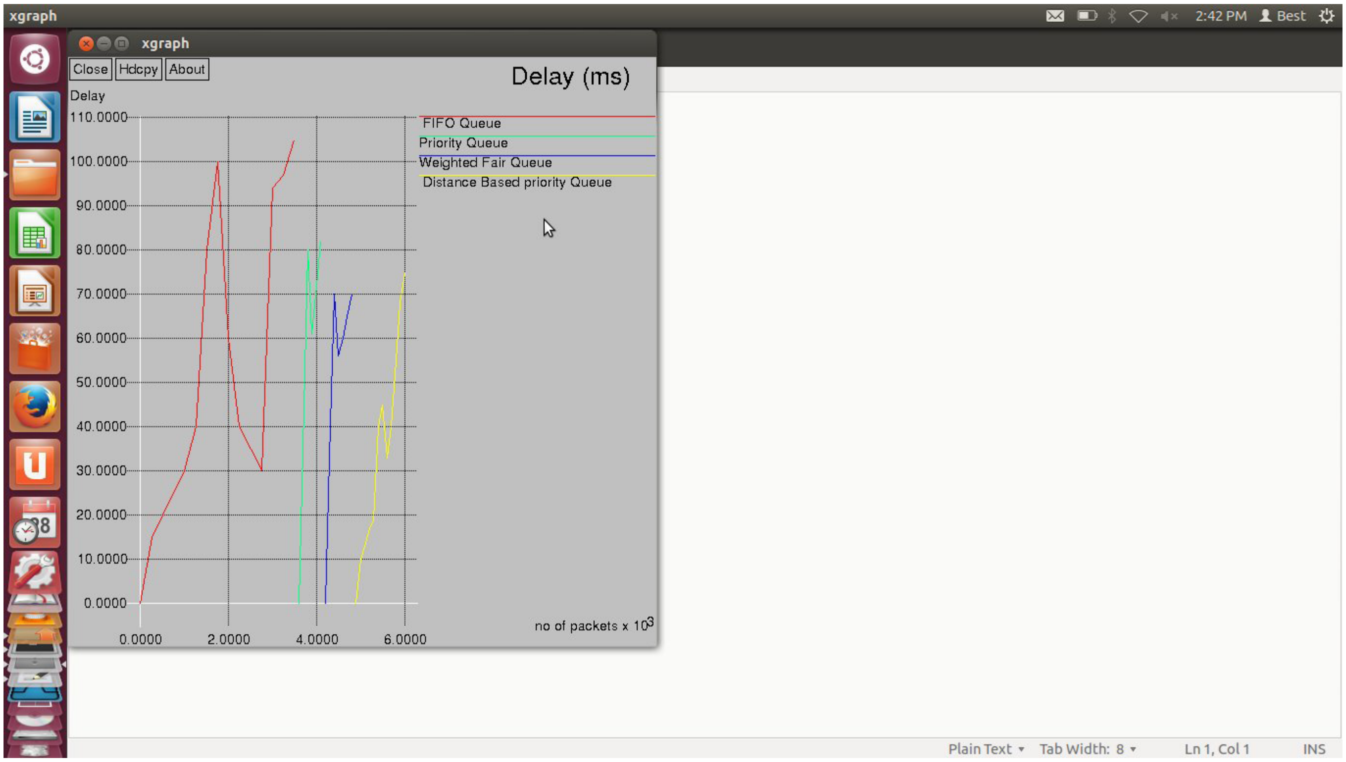

Figure 3. Performance Analyses for Delay.

Transmission delay = L/R ms

Where,

L is the packet length in bits is the link bandwidth between two routers in bits/second[14].

2.3 Performance Calculations Jitter

Figure 4 shows the 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.

3. Reliable Existing Queuing Mechanisms

3.1.1 Single Server Queue Mechanism

Throughput for drop tail queue in MANET environment with 20 nodes low performance and more delay obtained. The mechanism in which the data correspondence is kept or drop forms the queues. In queuing, a table of the parallelism is required, present should be a method on which an alternative that which packet is to be kept and which is to be dropped[15,16].

Single-sever service node consists of a server plus its queue. The issue related to the efficiency is of great significance. Signal interference plays an important part in the performance analysis of mobile ad hoc networks. The arrival of node in data packet is huge but the service provides a single in manner so delay is very high and more number of packet drops is obtained[17].

3.2 Adaptive Proposed Queue Management

Formulae used

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

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. Most of the packet is approaching at the speed of data.
- When $\lambda > \mu$, packet carry on to arrive at a steady rate but cannot be serve by the link at that speed[18].

3.2.1 Rebounded FIFO in DBPQ Mechanism

First in First out concept is a simple queue organization algorithm: it sets a predefined value in overall distance performance for the utmost length of the queue and

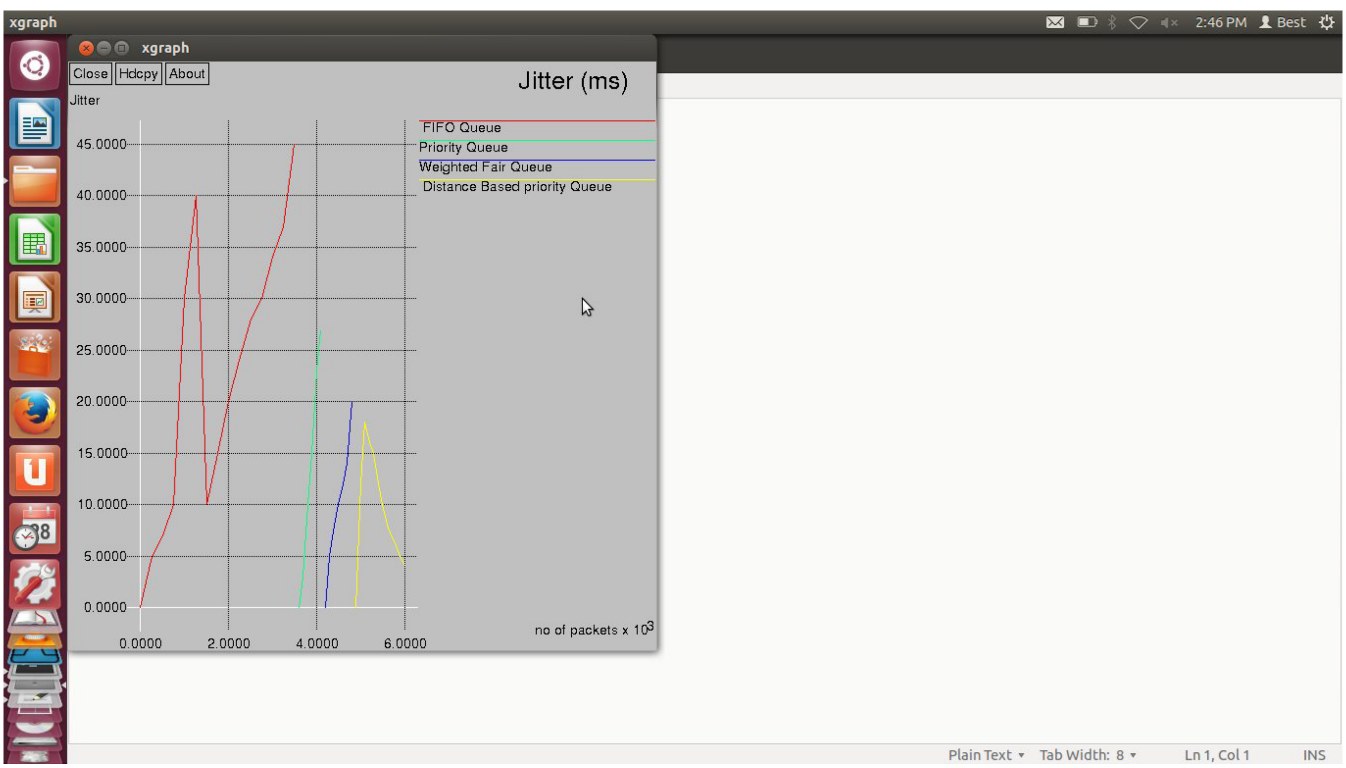

Figure 4. Performance Analyses for Jitter.

when this world 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 or treat identified, 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[19,20].

3.2.2 Adaptive RED Queuing DBP Methodology

RED (Random Early Detection) works by at random (based on certain probability) elimination packets at the nodes of the web, before the incidence of obstruction, when the average queue length exceeds the predefined minimum threshold. When the standard queue length goes past the maximum threshold, the chance of rejection becomes equal to 1. RED monitors the medium length of the queues by removal or ECN-marking packets based on statistical probability. If the barrier is nearly vacant, all incoming packets are taken[21,22].

4. Adaptive Approach in Distance Based Priority Queue Description

It can be used in an environment of the Tele-emergency cases.

This Queue consists of two stages.

1st –stage in priority level decision depends on priority levels.

2nd stage Figure 5 shows the Distance Metrics Calculations are taken place in which the bandwidth will be allocated adaptively according to the distance parameter, i.e... Bandwidth will be allocated with the long and short distance respectively[23].

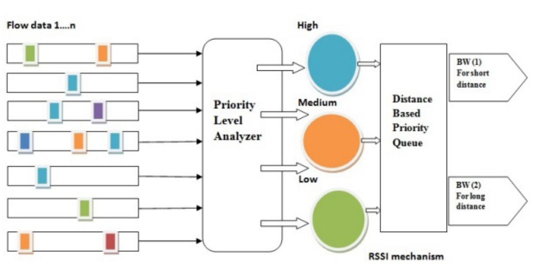

Figure 5. Functional Diagram of Distance Based Priority Queue.

- Advantages
  Efficient queuing Techniques for handling the dates in Tele-Emergency Cases.
- High Throughput.

4.1 DBPQ Algorithm Formation

1 Source Node – S1 to SN
2 Destination Nodes – D1 to DN
3 if S Node is a far node to D then
4 for each [P] in Node P = priority is selected
5 if MAX DIST [P], selected Bandwidth allocated
6 MIN DIST [P] selected Bandwidth is allocated
7 ENQUEUE RRP Packets in MINDIST to MAXDIST
8 end if
9 end do
10 else
11 for each source node is Deque the packet
12 endif
13 end

5.2 DBPQ Implementation with RSSI Algorithm Formations

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[24].

RSSI is generic radio receiver technology metric,

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

$$DNP_r = \frac{SNP_t\, BPG_t\, BPG_r\, AN\lambda^2}{(4\pi d)^2 E2EL} \rightarrow\rightarrow\rightarrow\rightarrow\rightarrow eqn \ldots\ldots(1)$$

DNP_r: Destination Node Receiving Power
SNP_t: Source Node Transmitting Power
BPG_t: Bandwidth Priority Gain of Transmitting antenna.
BPG_r: Bandwidth Priority Gain of receiving antenna
$AN\lambda$: Arrival Node Wavelength per node packet
$E2EL$: Ene to End Packet loss
d: Distance between node in overall prameters

i.e., to calculate distance between two nodes by receiving signal

$$d = \sqrt{\frac{SNP_t BPG_t BPG_r AN\lambda^2}{(4\pi)^2 E2ELDNP_r}} \rightarrow\rightarrow\rightarrow\rightarrow\rightarrow eqn \ldots\ldots(2)$$

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

AP_i: Set of Adaptive one hop neighbor of node i

D_{ij}: Distance between node i and j

CD_i: Closed Overall distance D_{ij}

To classify the conditions of positioning as
High Bandwidth allocation
Medium Bandwidth allocation
Low Bandwidth allocation
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)
Node is centre of the system
Ix=0, Iy=0
Z is the node axis of

$$Z_x = d_{ip}, \quad Z_y = 0$$

Y is locater coordinate axis

$$Y_x = d_{iq} \cos\alpha \, Z_{y=} d_{iq} \sin\alpha \rightarrow\rightarrow\rightarrow eqn \dots (3)$$

Where, $\alpha = Z_{iq}$

$$d_{ZY}^2 = d_{iZ}^2 + d_{iY}^2 - 2 d_{iZ} d_{iY} \cos\alpha$$

α **is calculated using** The following formula

$$\alpha = ar \cos\left(\frac{d_{iZ}^2 + d_{iY}^2 - d_{ZY}^2}{2\, d_{iZ}\, d_{iY}}\right) \rightarrow\rightarrow\rightarrow eqn \dots (4)$$

The calculation coordinates Z_i is

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

$$\alpha a = a_i Z = a_r \cos\left(\frac{d_{iZ}^2 + d_{ia}^2 - d_{Za}^2}{2\, d_{iZ}\, d_{ia}}\right) \rightarrow\rightarrow\rightarrow eqn \dots (5)$$

For b, coordinates

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

$$\alpha b = b_i Z = a_r \cos\left(\frac{d_{iZ}^2 + d_{ib}^2 - d_{Zb}^2}{2\, d_{ip}\, d_{ib}}\right) \rightarrow\rightarrow\rightarrow eqn \dots (6)$$

5.3 Discussion of Result RREQ(Rout Request), RREQ(Rout Reply)

Table 1. (Performance analysis)

Routing Protocol	Time	Total Packet	Throughput	Jitter
AODV	150mS	1040	1000	0.3971
AODV	180mS	1150	1100	0.4578
AODV	210mS	1220	1200	0.4888

6. Simulation and Result Analysis

Table 1 shows the presentation and Performance analysis metrics of Proposed DBPQ algorithms beneath a variety of constraint such as node thickness and mobility have been simulated using ns2[25,26]. The simulation is resolute by a trace file and presentation of the scheme is compare to node concentration and mobility. The network simulator-2 is used to simulate the function of cache and query index nodes to extract the popular data item based on routing information[27,28].

6.1. Performance Analyses for Delay

6.2 Performance Analysis for Jitter

6.3 Performance Analysis for Packet Loss

6.4 Performance Analysis for Throughput

7. Conclusion

In this research DBPQ queue technique has been implemented using RSSI algorithm. The performance evolution has been prepared with respect to QoS performance metrics such as delay, throughput packet loss, and jitter which shows in Figure 4, Figure 6, Figure 2 in the performance analyses were conducted comparing with FIFO, PQ, and WFQ. 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 tired 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.

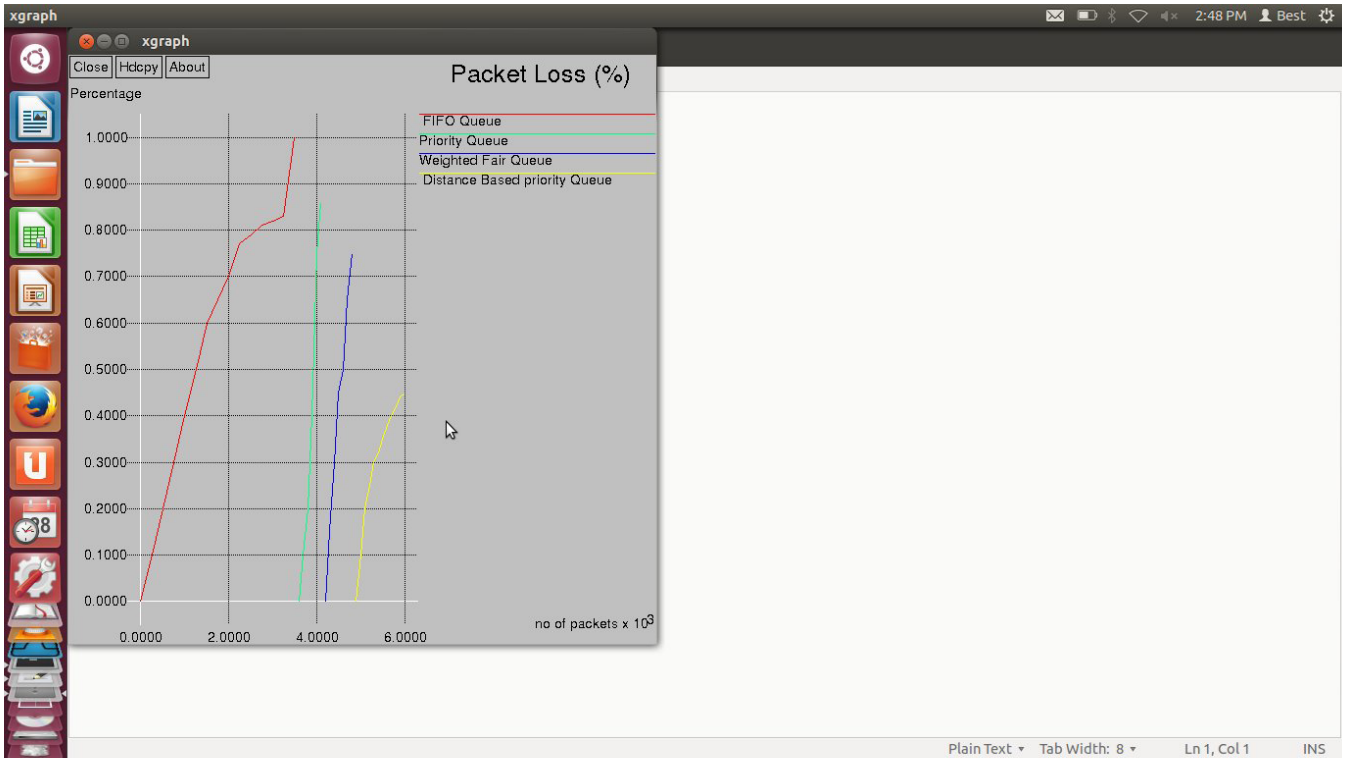

Figure 6. Performance Analyses for Packet Loss.

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