Implementation of Queuing Algorithm in Multipath Dynamic routing architecture for effective and secured data transfer in VoIP

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***Abstract*-VoIP is a technology that allows people to communicate each other with low cost effort in internet service. Quality of Service (QOS) is essential to secure and deliver a data in IP network. Such as (Delay, Jitter and Packet Loss) according to (ITU) International Telecommunication Union standards. To improve the QoS, different types of traffic management systems are used. Queuing is one of the vital mechanisms in traffic Management.**

The idea of this research is study the effect of different queuing algorithm with multipath dynamic routing architecture on VoIP and addresses the most appropriate queuing technique to improve VoIP QoS.

Simulation toll "NS2" is used to implement the task on VoIP network. In this research, Analysis report has been carried out and compared between different queue algorithm such as First In First Out (FIFO), Priority Queue (PQ)and Weight Fair Queuing (WFQ). It is found that PQ and WFQ are the most appropriate to improve VoIP QoS.

Keywords : VoIP, Multipath, FIFO Queue ,PQ Queue ,WFQ Queue, QoS, Internet Protocol, NS2

I. Introduction

Voice over Internet Protocol (VoIP) is a relatively new technology to transmit voice as a packets over an IP network. It has already achieved wide acceptance in global. Performance have been proved, as it is good replacement of Plain Old Telephone System(POTS)[25][26] . The potential of this technology is low cost and free calls. As the people are massively turning to VoIP technology in addition the popularity gives an increasing the need to provide real time voice quality and video service to the network.

II . Quality of Services (QoS) in Multipath VoIP

QoS is considered as potential of the network to produce consistently high-quality voice transmissions. In real-time application, VoIP is extremely bandwidth-and delay-sensitive. Application like E-Mail, FTP,HTTP are not sensitive to delay of transferring information. Therefore QoS of VoIP is having most effective consideration to make sure that the voice packets are not delayed or lost while transferred over the network. According to ITU the different parameters (Delay, Jitter and Packet Loss) can be used to measure the QoS of VoIP[26].

A. QoS treatment with ITU

According to ITU consideration , the quality of service is measured on different parameters like (dlay , jitter, and packet loss)[24][26]. These parameters can be controlled within the range to improve VoIP QoS. These factors briefly described in this section.

B. Latency :

Latency is the time between the moment a voice packet is transmitted and the moment it reaches its destination. It of course leads to delay and echo. It is caused by slow network links. This is what leads to echo.

There are two ways latency is measured: one direction and round trip. One direction latency is the time taken for the packet to travel one way from the source to the destination. Round-trip latency is the time taken for the packet to travel to and from the destination, back to the source. In fact, it is not the same packet that travels back, but an acknowledgement.

Latency is measured in milliseconds (ms) - thousandths of seconds. A latency of 150ms is barely noticeable so is acceptable. Higher than that, quality starts to suffer. When it gets higher than 300 ms, it becomes unacceptable.

Formula (1) shows the calculation of Delay where the Average Delay (D) is expressed as the sum of delays (*di*) , divided by the total number of all measurement (N)[26].

$$D = \sum_{i=1}^{n} di/N$$

C. Jitter in Packet Voice Networks

Jitter is defined as a variation in the delay of received packets. At the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Due to network congestion, improper queuing, or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant.

This diagram illustrates how a steady stream of packets is handled.

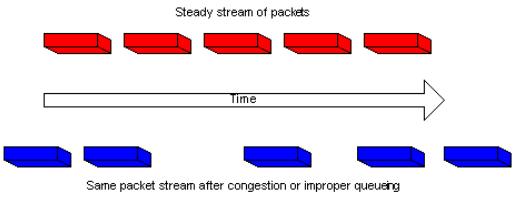

Figure 1: Packet Stream in Steady

When a router receives a Real-Time Protocol (RTP) audio stream for Voice over IP (VoIP), it must compensate for the jitter that is encountered. The mechanism that handles this function is the playout delay buffer. The playout delay buffer must buffer these packets and then play them out in a steady stream to the digital signal processors (DSPs) to be converted back to an analog audio stream. The playout delay buffer is also sometimes referred to as the de-jitter buffer[25].

This diagram illustrates how jitter is handled.

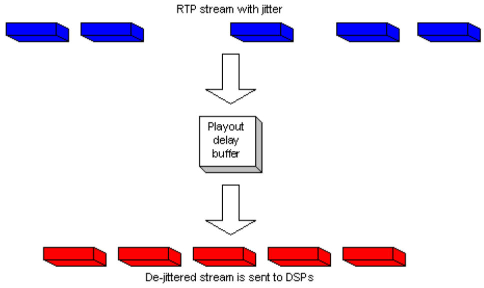

Figure :2 RTP with Jitter

If the jitter is so large that it causes packets to be received out of the range of this buffer, the out-of-range packets are discarded and dropouts are heard in the audio. For losses as small as one packet, the DSP interpolates what it thinks the audio should be and no problem is audible. When jitter exceeds what the DSP can do to make up for the missing packets, audio problems are heard.

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This diagram illustrates how excessive jitter is handled.

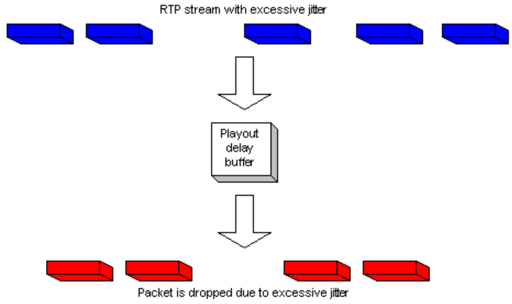

Figure: 3 Dropping packets

In this, Voice packets can tolerate only about 75Milliseconds (0.075 sec) but is preferred be 40 Milliseconds (0.040 sec) of jitter delay .

Equation (2) shows the calculation of jitter (j). Both average delay and jitter are measured in seconds. Obviously, if all (di) delay values are equal, then D = di and J = 0 (i.e., there is no jitter) [6].

$$J = f = \frac{1}{N-1} + \sum_{i=1}^{N}(di - D) \qquad {}^{2}$$

D. Packet loss:

Packet loss is the term used to describe the packets that do not arrive at the intended destination

that happened when a device (router, switch, and link) is overloaded and cannot accept any incoming data at a given moment [7]. Packets will be dropped during periods of network congestion. Voice traffic can tolerate less than a 3% loss of packets (1% is optimum) before callers feel at gaps in conversation [5].

Equation (3) shows the calculation of packet loss ratio defined as a ratio of the number of lost packets to the total number of transmitted packets Where N equals the total number of packets transmitted during a specific time period, and NL equals the number of packets lost during the same time period [6].

Loss packets ratio = (NL / N) × 100%

III. Queuing concept in Multipath VoIP:

Basically the network is planned to serve with different possibilities of traffic to send packets to multiple destination simultaneously. It can be called as Multicasting technique. The router acts as a major role in the resource allocation mechanism , so it has to implement any one of queuing algorithm , that manages how the packets are buffered during the transfer. Different queuing discipline can be used since it effects the packet latency by decreasing the time that the packets wait to be transferred[25]. In this paper, there are 3 commonly used queuing techniques are analyzed namely

First-In-First-Out (FIFO) , Priority Queue (PQ), Weight Fair Queue(WFQ) on multipath dynamic network topologies.

A. FIFO Queue:

The basic principle of FIFO queuing is that the first packet that arrives at a router is the first packet to be transmitted. An exception here happened if a packet arrives and the queue is full, then the router ignores that packet at any conditions

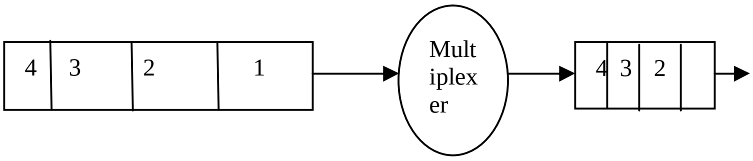

Figure: 4 FIFO Queue

E. PQ (Priority Queue):

The principle idea of PQ queuing depends on the priority of the packets, a highest priority are transmitted on the output port first and then the packets with lower priority and so on. When congestion occurs, packets with lower-priority queues will be dropped [25].

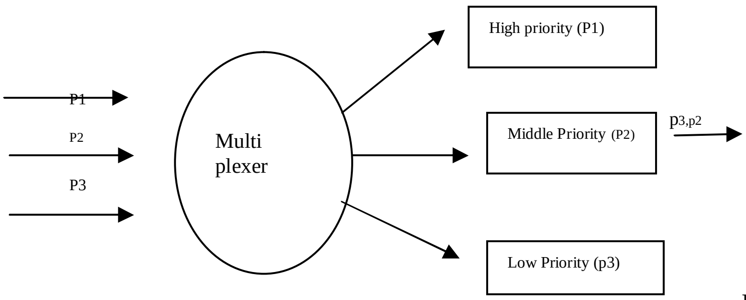

Figure: 5 PQ (Priority Queue) Diagram

P1-Queue1, P2- Queue2 P3- Queue3

C. WFQ (Weighted-Fair Queuing)

The Weighted-fair queuing discipline provides QoS by provides fair (dedicated) bandwidth to all network traffic for control on jitter, latency and packet loss. The packets are classified and placed into queues

According to information ToS field in IP header is use to identify weight (bandwidth). The Weighted-fair queuing discipline weights traffic therefore a low-bandwidth traffic gets a high level of priority. A unique feature of this queuing discipline is the real-time interactive traffic will be moved to the front of queues and fairly the other bandwidth shares among other flows[15].

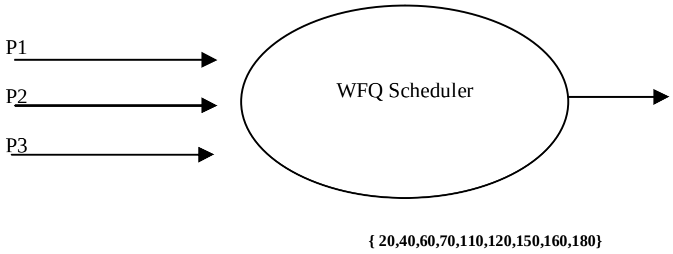

Figure: 6 WFQ Queuing Diagram

P1 = { 20,40,70} 50% Bandwidth
P2= { 60, 120.150} 25% Bandwidth
P3={110,160,180} 25% Bandwidth

IV. Simulation work with Multipath topology

The objective is to minimize the packet loss during the data transfer. So here queuing techniques achieves both security and integrity of data. Assume that T(N) - total number of nodes used to simulate the P(M) – Total number of Path in the topologies[26][15].

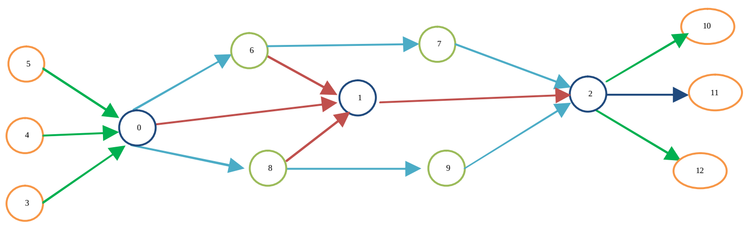

Figure : 7 Structural design of queuing implementation over Multipath topology[20]

There are M path , such as { P1,P1,P3…………PM)
There are N node, such as (T1,T2,T3……………Tn)

Let P= [P1, P2... Pm] denotes the possibilities characteristics of the paths, where Pi (i = 1,2,…,M) is the probability that the path i is compromised. A dynamic routing scheme is used to find the less packet loss with N shares onto the M available paths[12].

$$ni \ge 0, \sum_{i=1}^{M} \mathrm{ni} = \mathrm{N}$$

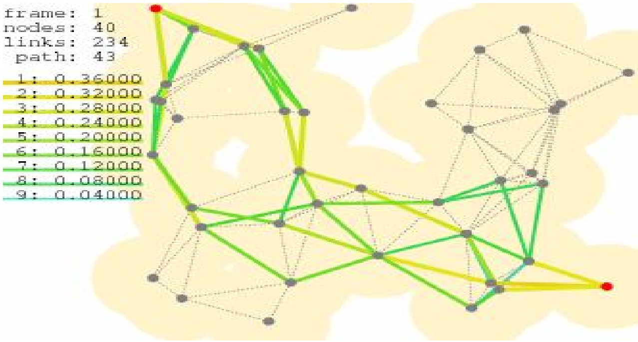

Figure : 8 Simulated Network Topology.(NS2)

V. *Simulation Result and Observation:*

In this research, a Comparison between the effects of different queuing duplicates such as FIFO, PQ and WFQ on VoIP QoS. To measure the QoS of the VoIP application during collected statistics (parameters) such as: Voice delay (sec), Voice jitter (sec), Voice traffic sends (packet/sec) and Voice traffic received (packet/sec). The duration of simulation is 5 minutes and the results are obtained as shown in figure.

The blue line represent FIFO algorithm; whereas the red line represent PQ algorithm; whereas the green line represent WFQ algorithm.

Figure (a) shows the (Jitter = 0) using PQ and WFQ algorithms; whereas jitter using FIFO algorithm = 0.0038 sec but did not exceed the time constraint (0.075 sec).

Figure (b) shows the end to end delay using PQ and WFQ algorithms acceptable value but using FIFO algorithm did exceed the time constraint (0.15 sec).

Figure (c) shows the voice traffic received using FIFO algorithm = 380 (Packets/sec), the voice traffic received using PQ and WFQ = 400 (Packets/sec).

Figure (d) shows the voice traffic sent in all three cases = 400 Packets/sec.

Figure (5) shows that in FIFO queue, the delay does exceed the time constraint 150 ms and more packet loss. According to NS2 , packet loss ratio is the ratio of packets dropped to the total packets transferred to this cloud multiplied by 100%.

Packet loss ratio = 20/400×100% =5%.

In PQ and WFQ algorithms have no packet loss and end to end delay with acceptable ratio.

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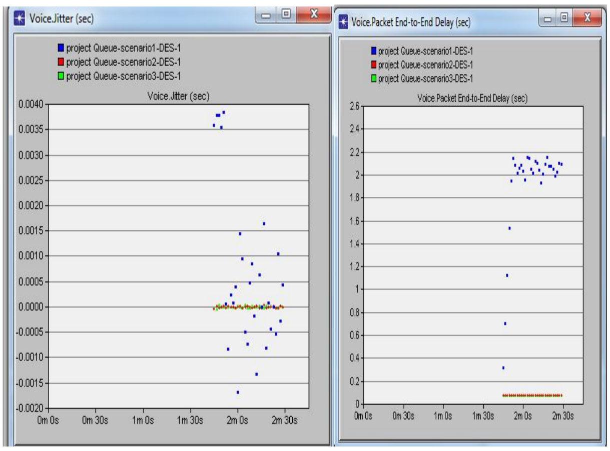
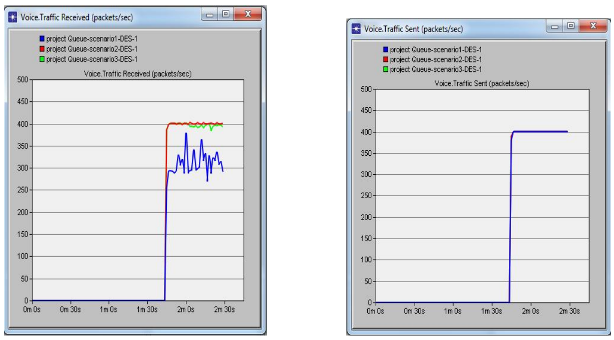


a) Voice Jitter

b) Voice Packet End to End Delay



C) VoIP Traffic Received (Packet/Sec)

d) VoIP Traffic Send (Packet / Sec)

Figure : 9 Comparative study of Queuing discipline

Parameters	**FIFO**	**PQ**	**FWQ**
Packet traffic sent(Pack/Sec)	**400**	**400**	**400**
Packet traffic Received (Pack/Sec)	**380**	**400**	**400**
Voice packet End-To-End-Delay(sec)	**2.15**	**0.076**	**0.076**
Voice Jitter (sec)	**0.00383**	**0.00003**	**0.00003**
Packet Loss	**20**	**0**	**0**

Table.1 : Report of Queuing implementation on VoIP quality[4]

VI. Conclusion:

The presented research regards with the affects of different queuing disciplines on the performance of VoIP using Multipath Dynamic Algorithm with NS2. Simulations results allow us to conclude that; Improving the QoS of voice traffic based on the Priority and Weighted-fair queues are the most appropriate scheduling schemes because the values of the parameters are within the acceptable range such as delay, jitter, packet loss.

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