Performance Comparison of Different Codecs over VoIP with Different Queuing Algorithms for IP Based Network

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Abstract **- This paper compares the performance of the popular VoIP codecs G.711, G.723.1 and G.729 for different queuing algorithms over an IP network based on SIP architecture, using RTP as a transport protocol. This is based on different simulations in order to evaluate the performance of each codecs on different queuing algorithms and find the best codec. For this objectives this paper analyse QOS parameters, principally jitter, End-to-End delay, packet delay variation, MOS. The OPNET Modeler 14.5 has been used for implementing this work and monitoring all results in order to choose the best codec for transporting voice over IP based network with queuing algorithms**

Keywords - VoIP, G.711, G.723.1, G.729, SIP, QOS, FIFO Queuing, Priority queuing, WFQ, jitter, End-to-End delay, Packet Delay Variation, OPNET Modeler 14.5.

I. INTRODUCTION

VoIP has become one of popular technology in the world. It is widely used in the network to transmit the voice, file, video from one system to another system. VoIP network is very cheaper compared to the PSTN network. It is very cheap for installation and single connection can be providing by ISP for data transfer and voice transfer. PSTN network provided only one connection for voice transfer. Here we are using SIP (Session Initiation Protocol) for establishing and terminating the connection between the user, it is proposed by the IETF (Internet Engineering Task Force) as a standard for IP telephony [2]. In this paper we are using different VoIP codecs G.711, G.723.1, G.729 which are operating at different bit rates and with different configuration also. These codecs are varying with QOS parameters like End-to-End delay, jitter, and packet delay variation [1].

In this paper we will discuss VoIP technology in section II, different queuing methods in section III and Simulation scenarios in section IV. Simulation results and conclusion will be discussed respectively in section V and section VI.

II. VOIP BASICS

A. *VoIP system*

In VoIP network we are using SIP or H.323 as signalling protocol. These protocols are used to create, control and terminate the session between the users. Once a session is established then Real Transport Protocol (RTP) or User Datagram Protocol (UDP) are used to transport the data over IP.

B. *Codecs*

Codec used to convert analog signal into digital signal and digital signal into analog signal. The analog signal is first sampled based on sampling rate of 8 kHz and then quantization take place in which every sample can be represented by 8 or 16 bits. Encoded output can be encapsulated into RTP packet or UDP packet for transmission over the IP network. Here we are using different audio codecs like G.711, G.723.1 and G.729. They are varying with different parameters like bit rate, delay and encoding algorithms [2]. G.711 is high bit rate codec with 64 kb/s whereas G.723.1 and G.729 operate at low bit rates of 5.3 kb/s and 8 kb/s respectively [1].

C. *Parameters of QOS for VoIP*

End to End delay: End-to-End delay occur due to the encoding and decoding delay and jitter buffering delay. End-to-End delay is acceptable up to 150ms and above that it will degrade the quality of voice at the receiver side ([1], [2]).

Jitter: An End-to-End delay variation between the two consecutive packets is called jitter [1]. Packet loss can be tolerable up to 10% for high bit rate codec G.711 and 5% for low bit rate codec G.729 [1].

Throughput: How much data can be transferred from source to destination in given amount of time is called throughput. Throughput can measure packet per second which is generated by the application [1].

D. SIP(Session Initiation Protocol)

Session initiation protocol is a signalling protocol used for establishing, modifying and terminating session over IP network. In SIP RTP is used to carry the data and send to the receiver.

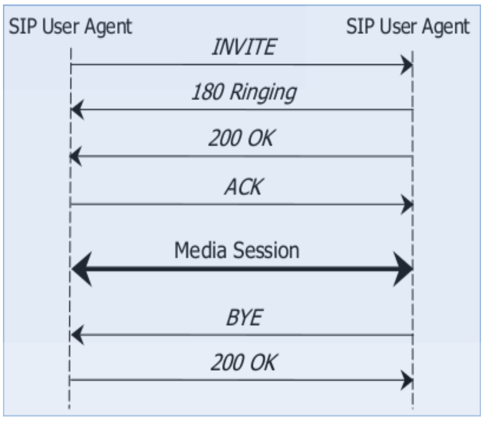

Figure 1: Basic SIP Call Flow [3]

As shown in Figure 1, SIP call flow establishes a session. An endpoint in SIP is known as a User Agent or UA. A UA wishing to establish a session creates an INVITE and sends it to another UA. The other user gives response to invite by 180 ringing to indicate that it is alerted by the signal. If the session is created successfully then it gives the response with 200 OK. To acknowledge the receipt of 200 OK the UA responses with ACK signal [3].

III. Queuing Algorithms

There are various queuing algorithms used to control the transmitted packet in the network. The queuing algorithm also affect the packet delay by decreasing the time that packet wait to be transmitted [4].

A. FIFO Queuing

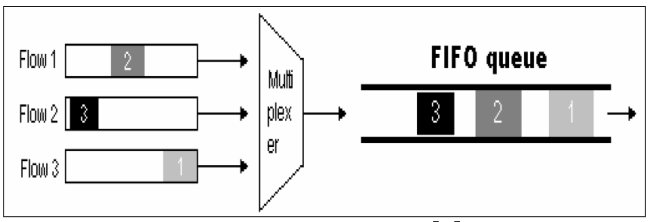

Figure 2: FIFO Queue [4]

The basic principle of FIFO is that the first packet transmitted from the transmitter side, it will receive first at the receiver side. If the FIFO queue is full with the packet then router can dropped that packet. We can easily see in figure 2 [4].

B. Priority Queuing

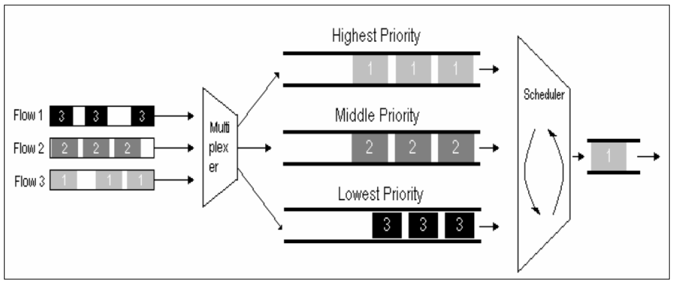

Figure 3: Priority Queuing [4]

The basic principle of the priority queuing depends on priority of the packets, packet with highest priority are transmitted first at the output side and then send those packet which have low priority.

C. WFQ Queuing

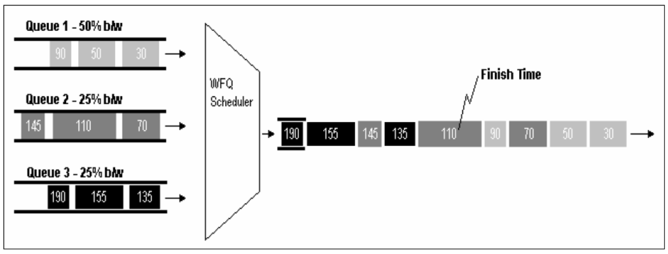

Figure 4: WFQ Queuing [4]

The waited-fair queuing provides fair bandwidth to each application to control the delay, packet loss, jitter. The packets are placed into queue according to the TOS field in the IP header is use to identify the weight. We can easily see in the figure 4 that low bandwidth traffic gets high level of priority.

IV. Simulation Scenarios

OPNET (Optimized Network Engineering Tools) simulator is used to compare three VoIP codecs over three different queuing algorithms. OPNET is discrete event simulation tools and we can get simulation of different types of network in OPNET [5].

This paper present three scenarios for each codecs G.711, G.723.1 and G.729 over queuing algorithms. In OPNET Modeler we can create IP network which shown in figure 5[5].

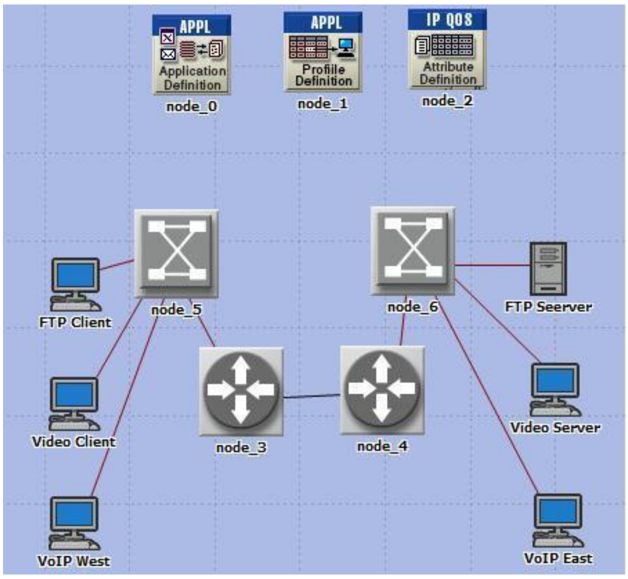

Figure 5: IP Network

Here we are using an Application Configuration and Profile Configuration object in the network. We used switch, IP router and Ethernet work station as a FTP client, Video client, VoIP A, VoIP B and Video server.

Application Configuration is an object used to define and configure different type of application in the network. Here we configured three applications like FTP, VoIP and Video. In Profile configuration object we can configure starting time as well as ending time of the simulation. QOS Attribute node is mean of attribute configuration details that assess protocols at the IP layer. It deals with the three queuing profiles: FIFO, Priority queuing and Weighted-fair queuing [4].

V. Simulation Results

In this section, the measurement is analyzed and discusses to evaluate the QOS performance for each in different queuing algorithms. Here we take our simulation for the period of 50 seconds. After the implementation of the simulation, 195 seconds is chosen to be the period of simulation in order to display the results.

From figure 6 to 9 are the simulation results for different codecs over FIFO Queuing algorithm. From the figure we can say that G.711 and G.729 are good codec compared to G.723 because we got the less packet delay variation, jitter, End-to- End delay and less number of packets dropped in the network. We got the more packets at the receiver side in G.711 and G.729.

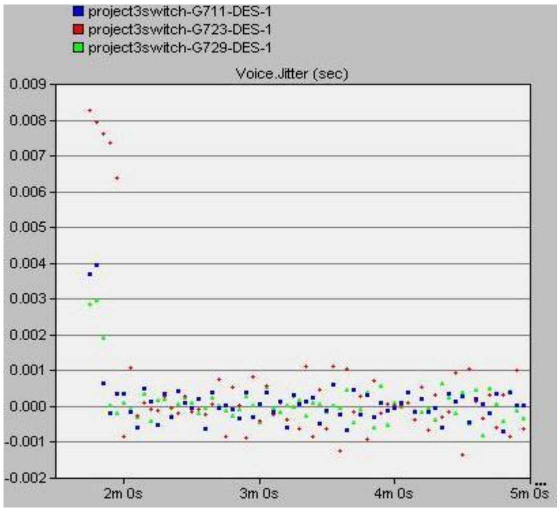

Figure 6: Jitter for FIFO Queuing

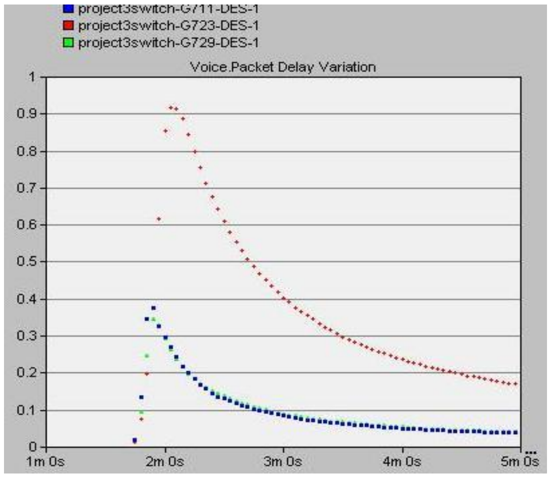

Figure 7: Packet Delay Variation for FIFO Queuing

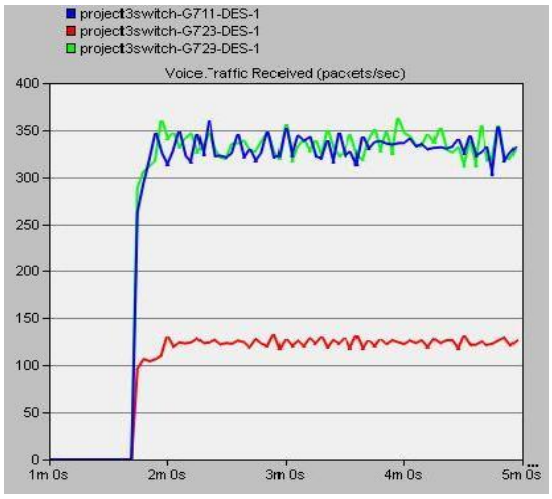

Figure 8: Traffic Received for FIFO Queuing

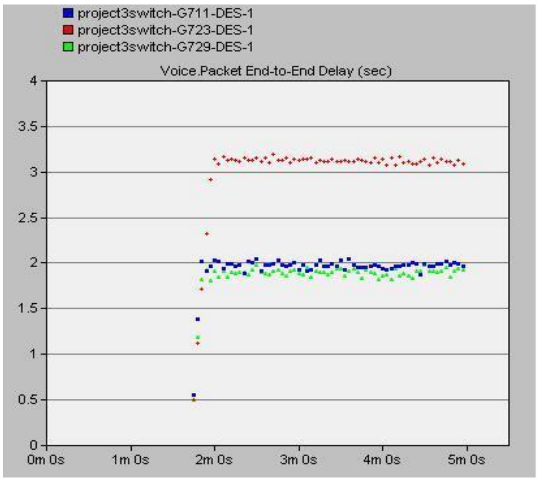

Figure 9: End-to-End delay for FIFO Queuing

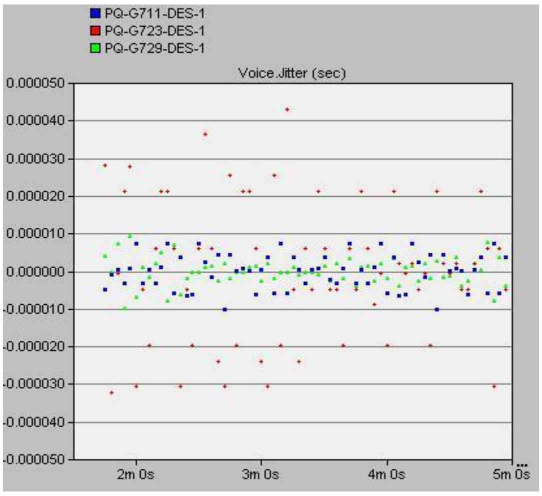

Figure 10: Jitter for PQ

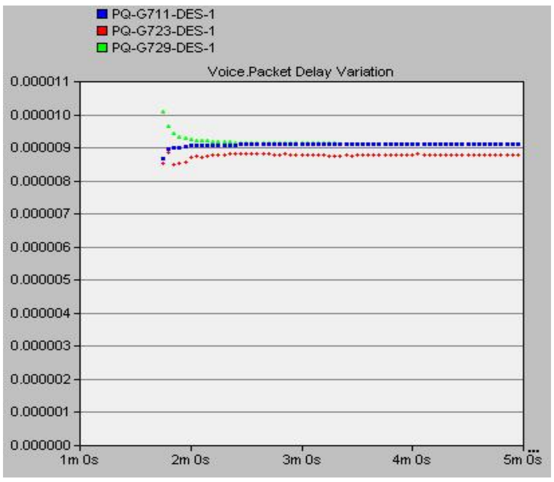

Figure 11: Packet Delay Variation for PQ

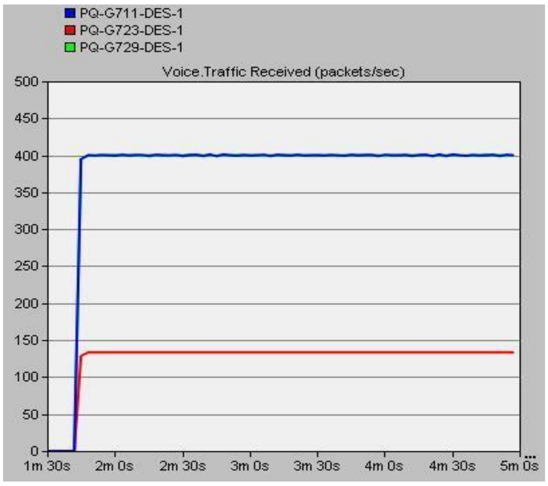

Figure 12: Traffic Received for PQ

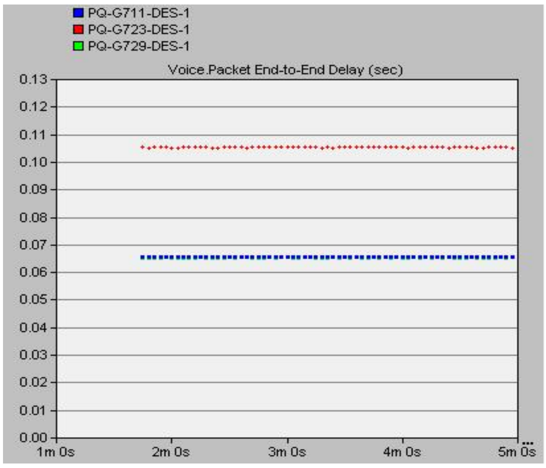

Figure 13: Packet End-to-End delay for PQ

From Figure 11 to 13 we can see easily that G.711and G.729 got the less jitter, less End-to-End delay and high number of packet received at the receiver side. In PQ algorithms packet delay variations are nearly same for every codecs.

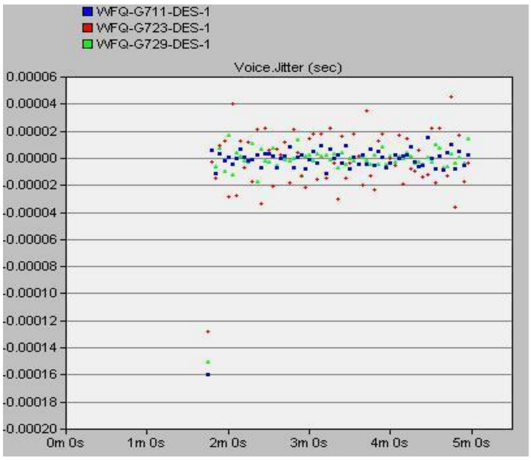

Figure 14: Jitter for WFQ

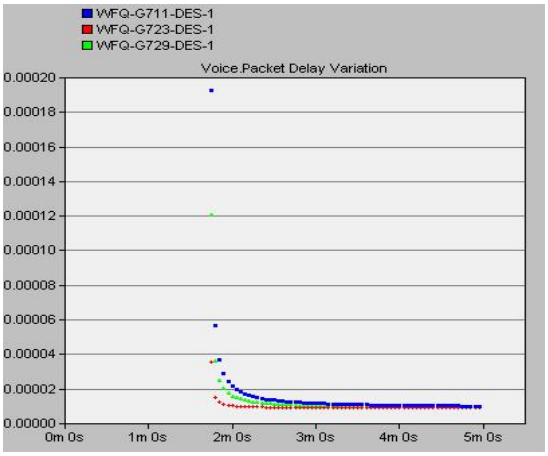

Figure 15: Packet Delay Variation for WFQ

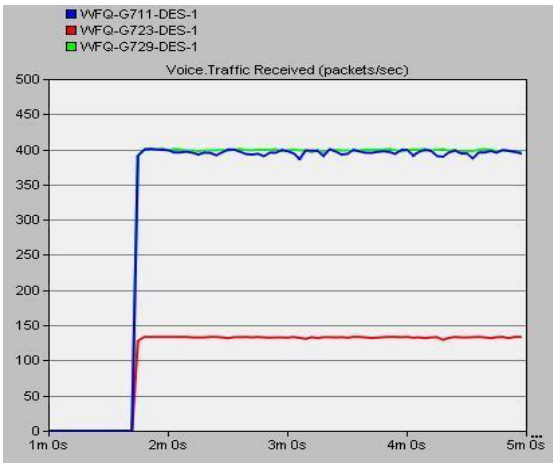

Figure 16: Traffic Received for WFQ

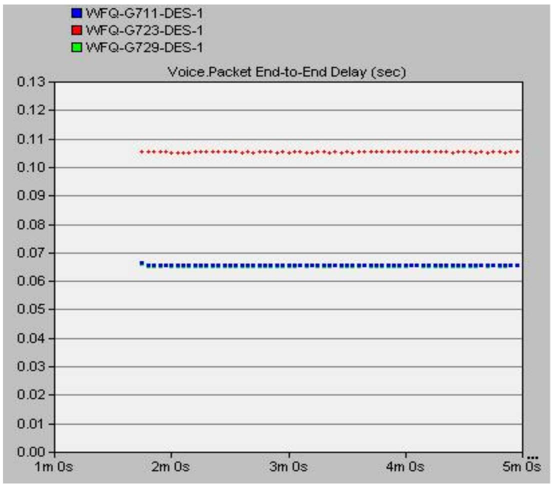

Figure 17: End-to-End Delay for WFQ

From figure 14 to 17 we can say that for WFQ algorithm, G.723 codec get the higher value in jitter, End-to-End delay compared to other codecs. Packet delay variations for all the codecs are nearly same. Traffic received for G.723 codec is low compared to other codecs.

VI. CONCLUSIONS

This paper compares three audio codecs G.711, G.723 and G.729 for VoIP using IP network. The results shows that jitter, End-to-End delay in case of G.723 are high compared to G.723 and G.729 in all the algorithms. Packet delay variation for FIFO is high and for PQ and WFQ algorithms are nearly same. Traffic received at the receiver side is low in G.723 codec for all algorithms and for G.723 and G.729 are nearly same. In short G.711 and G.729 codecs are good for FIFO, PQ, and WFQ algorithms compared to G.723.

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