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Performance Analysis of G.711 and G.729 Codec Schemes under Various Queuing Techniques in Voice over Internet Protocol Transmissions

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Abstract

Advancements in internet technology have enabled the integration of different traffic types i.e. data, video, and voice into a single network. This technology offers many benefits but also presents some challenges. Real-time traffic services such as VoIP require a certain Quality of Service (QoS) which cannot be guaranteed on the Internet therefore, key performance metrics become all the more important. The choice of codecs and queuing techniques becomes crucial for ensuring optimal performance, especially in networks with diverse traffic types. This research therefore compared the effects of the combinations of these tools (i.e. queuing techniques and codec schemes) on the quality of VoIP. A simulation approach using the OPNET Modeler 14.5 tool has been used to simulate a network supporting three different types of traffic namely: FTP traffic, Video conferencing traffic, and VoIP traffic. While maintaining the same topology and traffic of the network, different types of codec schemes and queuing techniques have been tested through the measurement of parameters such as delay and throughput. The custom Queuing technique showed the best performance overall while FIFO suffered the highest delay. The graphs were observed to follow the same pattern regardless of the codec scheme used however, G729 performed the better of the two as it received higher amounts of voice traffic and slightly lower delays compared to G711.

Keywords: Codec Schemes, Queuing Techniques, Voip, Opnet, Quality Of Service, Video Conferencing, File Transfer Protocol, Performance Metrics.

1. Introduction

The increasing trend in the use of voice communication over the Internet comes with the demand for high-quality voice data. Voice over Internet Protocol (VoIP) is a technology that allows users to make telephone calls over a data network instead of the traditional Public Switched Telephone Network [1]. The efficiency and quality of a VoIP system are influenced by several factors, including queuing techniques and codec schemes, which play important roles in shaping its performance. While each of these technologies has distinct roles to play in a network, it is important to analyze and identify optimal combinations that enhance the quality and efficiency of VoIP, addressing the need for a comprehensive understanding of how these factors interact with one another.

The study undertakes a performance analysis of two widely used VoIP codec schemes, G.711 and G.729, focusing on their performance characteristics under different queuing techniques within a network. G.711, a high-bitrate (64 Kbps) codec and G.729, a codec which offers a good level of call quality at a low bit rate of 8Kbps, represent two distinct approaches to voice data transmission [2,3].

The goal is to investigate how these codec schemes perform under various queuing techniques namely; Priority Queuing (PQ), Custom Queuing (CQ), Weighted Fair Queuing (WFQ), and First in First out (FIFO). Queuing governs how packets are buffered while waiting to be transmitted thus affecting the performance of the applications and utilization of network resources.

The quality of a VoIP application is determined by its quality-of-service parameters such as delay, jitter, and packet loss however end-to-end delay, and traffic sent and received are considered for this research [4].

2. Literature Review

A codec is a device or computer program for encoding or decoding a digital data stream or signal [5]. There are several codecs used for VoIP communication each having its bandwidth and characteristics. Both G711 and G729 are International Telecommunication Union (ITU-T) standards for encoding. G.711, also known as Pulse Code Modulation (PCM) consumes a higher bandwidth compared to more modern codecs [2]. G.729 is a low-bitrate codec designed to conserve bandwidth while maintaining acceptable voice quality. It utilizes the CS-ACELP algorithm, a form of algebraic code-excited linear prediction that efficiently compresses voice signals [3].

Queuing techniques play a crucial role in influencing packet delay, jitter, and overall voice quality in VoIP communications [6]. They include FIFO, PQ, WFQ, CQ, etc. In FIFO queuing, the packet that comes first in the buffer is treated first [3] while PQ prioritizes packets based on assigned priority levels and transmits them in decreasing order of priority [7]. CQ addresses the biggest shortcoming of PQ ensuring a guaranteed minimum bandwidth to each queue, thereby protecting low-priority queues [8]. WFQ assigns weights to different queues, allowing for proportional sharing of bandwidth among different types of traffic [9].

K. Balasundaram in [10] gives a comparative analysis of three queuing techniques namely; FIFO, PQ, and WFQ. Three simulation models of network topologies were tested to evaluate the network performance with each topology consisting of a different number of routers. The study was carried out on some QoS parameters namely; Traffic dropped, Traffic Received, and packet end-to-end delay with the results showing that the WFQ technique has a superior quality over the other techniques.

In [11], the effects of different queuing disciplines on packet delivery for three applications: FTP, Video, and VoIP were studied using OPNET. This paper presented how the choice of a queuing discipline affects the applications and utilization of network resources in routers. PQ and WFQ were found to be most appropriate for VoIP, and WFQ for video while the results for FTP were not presented.

In [12], a comparison was made between the use of FIFO and PQ mechanisms in a mixed traffic scenario (file transfer and VoIP applications) in a MANET. PQ was implemented based on packet Type of Service (ToS) with VoIP data packets being given priority. PQ was seen to give a better Quality of Service (QoS) as opposed to FIFO.

S. Gurrapu [13], evaluated the performance of VoIP using G.711, G.723, and G.729 codecs over Wi-Fi networks while increasing the number of nodes. Mean Opinion Score (MOS), average end-to-end delay and disconcert were evaluated and discussed. The study concluded that the G.711 codec was the best codec for VoIP over Wi-Fi. It is important to note that the results may not be the same for a wired network.

The author in [14] evaluated the performance of different VoIP codecs over the WiMAX network. G711, G723, G726, G728, and G729 were evaluated using network performance metrics such as MOS, packet end-to-end delay, jitter, and packet delay variation. The simulation results showed that G723 performed better than the other codecs as it showed the highest average MOS value and traffic received while having the lowest jitter and end-to-end delay.

Ahmed [15] studied the performance of FIFO, PQ, and WFQ for different codec formats namely; G711, G729, and G723 in a wide area network. All three queuing disciplines were tested for the three codec schemes. This researcher reported that all the collected jitter values were below the maximum acceptable value of 50 milliseconds. The units for end-to-end delay graphical results were seconds however the researcher recorded them to be in milliseconds without converting them accordingly. A study done in [16] where multiple VoIP codec schemes were evaluated using NS-2 simulation tool showed that G711 had the highest end-to-end delay. This network was made up of four LANs connected via the internet.

A wired Local Area Network with multiple traffic types was designed in [17] focusing on enhancing network performance and Quality of Service (QoS), performance metrics like average end-to-end delay, jitter and traffic received were compared under three queuing techniques namely; FIFO, PQ and WFQ. WFQ and PQ are seen to have the highest amount of VoIP traffic received with FIFO having the lowest amount.

3. Research Methodology

A. Research Methods

The figure below shows a summary of the steps taken during the research process.

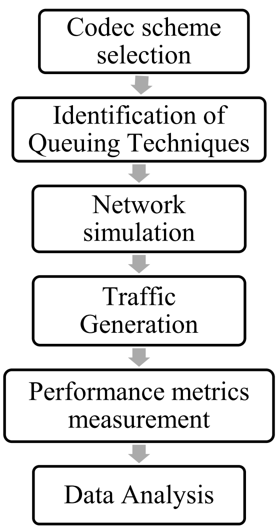

Figure 1. Research Process

B. Simulation

Simulations are employed by network designers and engineers to test a designed model on a platform that duplicates the real environment. The behavior of the simulated network or system can be studied to predict its strengths and weaknesses before implementing the model in a real environment [18]. In this research, the OPNET Modeler 14.5 was used to simulate a model of the network under investigation.

C. Reliability and Validity

The use of the OPNET simulation tool provides a controlled environment for experimentation, enhancing internal validity. The experiment can be replicated through the description of the methodology ensuring the reliability of the study. The literature reviewed also shows that various studies have used simulation as a methodology for evaluating different research problems. The results of this research were compared with those from other papers that undertook similar studies using different simulation tools.

D. Network Model

The type of network chosen for this research is a wired local area network supporting three different types of network traffic namely: File Transfer Protocol (FTP) traffic, Video conferencing traffic, and VoIP traffic. The network was set up as shown in Fig. 2. While maintaining the same topology and traffic of the network, different types of codec schemes and queuing techniques were tested resulting in different scenarios.

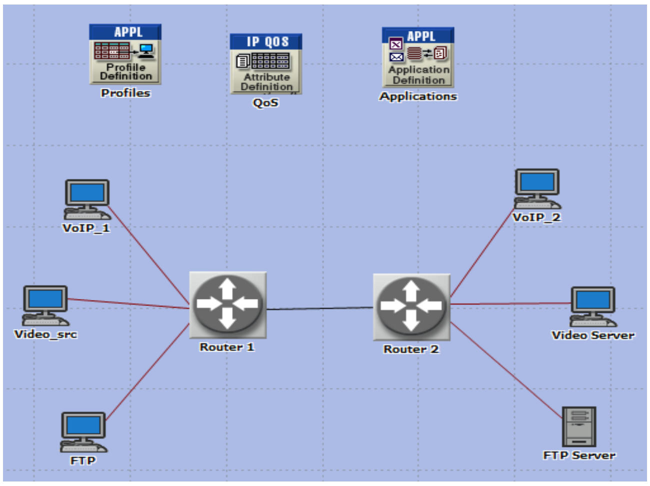

Figure 2. Network Model

4. Results and Analysis

The findings of the study have mainly been presented in the form of graphs and tables as shown below. The parameters considered were end-to-end delay, traffic sent, and traffic received.

A. The G711 Codec Scheme

This scenario was simulated to study the effects of using the G711 codec scheme for the different queuing techniques under investigation.

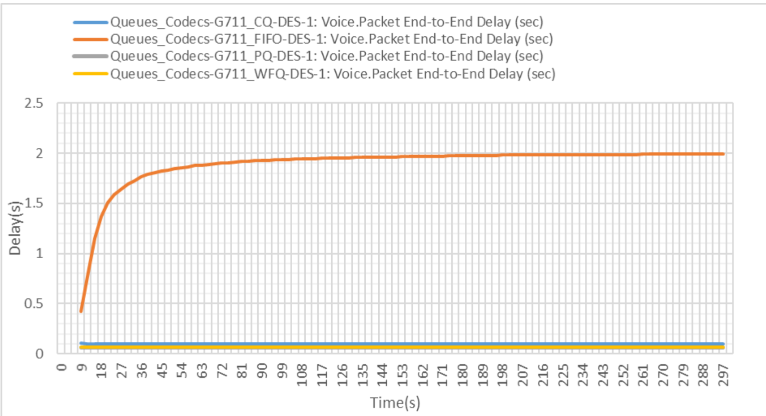

Figure 3. Voice packet End-to-End Delay

The VoIP traffic suffered the highest delay under FIFO queuing as seen in Fig. 3. The remaining three queuing disciplines experienced about the same amount of delay with WFQ and PQ having the lowest.

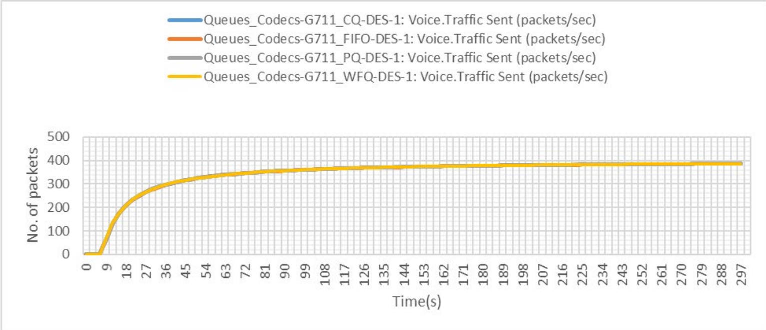

Figure 4. Voice Traffic Sent

The number of voice packets sent per second was the same for all queuing disciplines therefore, only a single line appears on the graph in Fig 4. This is because the graph is in an 'overlaid statistics' presentation.

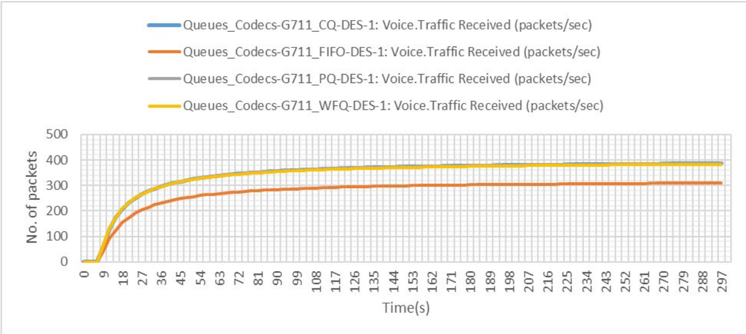

Figure 5. Voice traffic received

In Fig. 5, the number of voice packets received per second was the same for CQ, PQ, and WFQ. They therefore have their curves overlaid with one another. FIFO received the lowest amount of voice traffic in this scenario.

Type of queuing	CQ	FIFO	PQ	WFQ
Av. traffic sent (Pkts/sec)	386.59	386.56	386.63	386.63
Av. traffic received (Pkts/sec)	386.57	309.90	386.62	383.40
Throughput	99.90%	80.20%	99.90%	99.20%
Delay(sec)	0.101	1.893	0.066	0.066

Table 1. Summary of Voice Traffic Parameters Under G711

B. The G729 Codec Scheme

The following results were collected from the scenario simulated to study the effects of using the G729 codec scheme for the different queuing techniques under investigation.

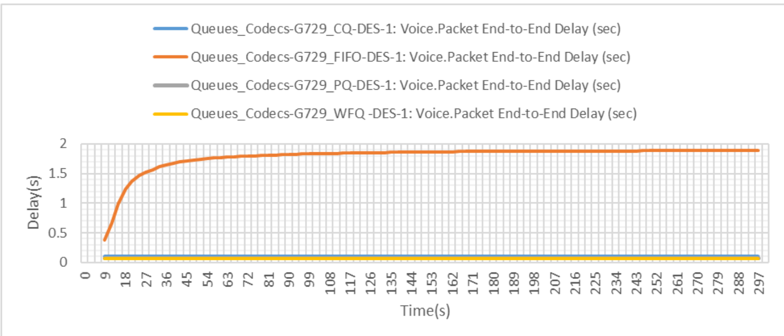

Figure 6. Voice Packet End-To-End Delay

The voice packet end-to-end delay under the G729 codec scheme was the highest in the FIFO queuing discipline while the rest of the queuing disciplines recorded an almost zero delay each.

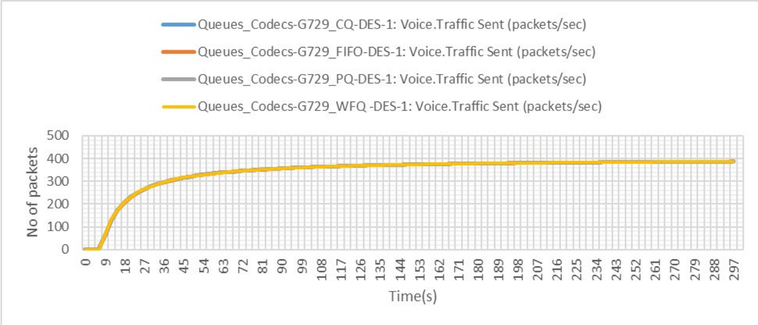

Figure 7. Voice Traffic Sent

The same amount of voice traffic was sent for all queuing disciplines therefore the graph appears to have a single line due to the 'overlaid statistics' presentation option chosen in the simulation application.

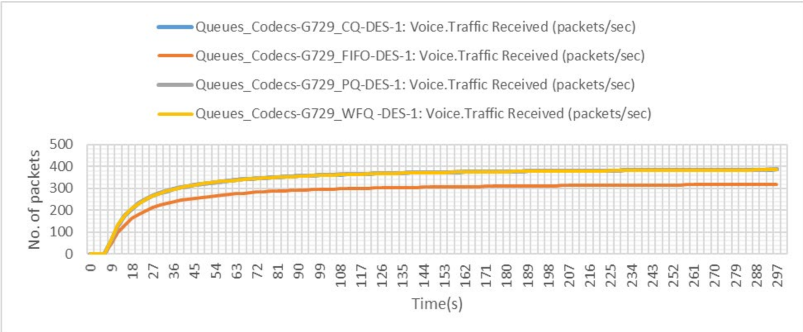

Figure 8. Voice Traffic Received

The same amount of voice traffic was received in CQ, PQ, and WFQ while FIFO received less.

Type of queuing	CQ	FIFO	PQ	WFQ
Av. traffic sent (Pkts/sec)	386.59	386.55	386.62	386.62
Av. traffic received(Pkts/sec)	386.51	319.45	386.61	384.99
Throughput	99.98%	82.60%	99.90%	99.60%
Delay(sec)	0.100	1.787	0.066	0.065

Table 2. Summary of Voice Traffic Parameters Under G729

C. Validity

Figure. 9 is an extract from the results of [17] showing the average end-to-end delay experienced by different codec schemes as the number of VoIP connections in a LAN was increased. This was done using NS-2 simulation tool and did not consider queuing techniques. It shows that more delay is experienced under the G711 codec as compared to G729. These reults agree with the results collected in this research where G711 is observed to suffer from higher delays compared to G729 for most of the queuing techniques used.

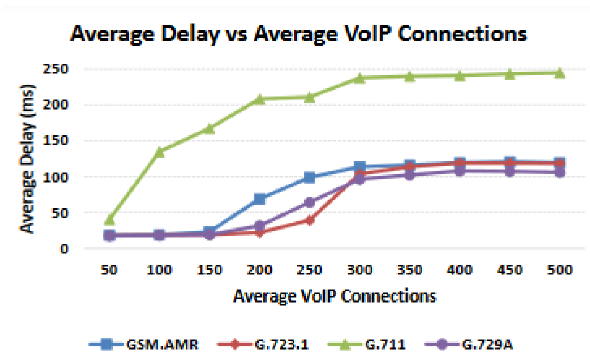

Figure 9. Average end-to-end Delay [17]

Figure. 10 below shows graphs depicting voice traffic received in bytes per seconds in a LAN in [18] where the type of codec scheme used was not taken into consideration. This network however also carried other traffic types apart from VoIP such as HTTP and video conferencing. The least amount of voice traffic was received under the FIFO queuing technique with WFQ and PQ recording almost the same amounts. These results are in line with those collected in figures 5 and 8.

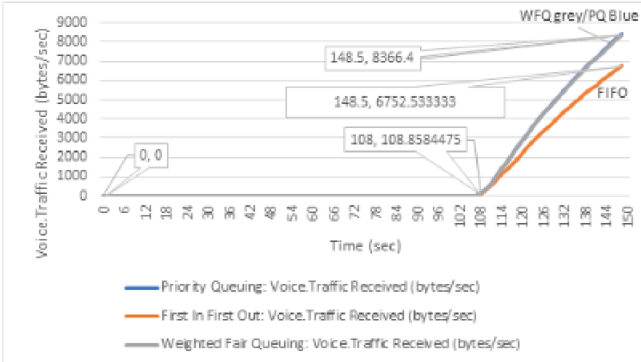

Figure 10. Voice Traffic Received (Bytes/Sec) [18]

5. Discussion

Parameters considered in this research were packet end-to-end delay, traffic sent, and traffic received.

1) G711 Codec Scheme

a) Packet End-to-End Delay

PQ and WFQ were observed to have suffered the least voice packet end-to-end delay while FIFO experienced the highest. According to the literature reviewed PQ gives the highest priority to real-time traffic such as voice and will transmit all voice packets available in the queue before attending to any other types of packets [19,20]. This significantly reduces the delay between the transmitter and the receiver. In FIFO, all packets are put into the same queue, and the queuing delay increases as congestion increases which affects the performance of real-time applications such as voice. Voice packets queued for longer than the maximum acceptable delay are dropped creating a bigger gap (delay) between those at the transmitter's side and those at the receiver's end. This therefore could be the reason for the FIFO technique having the highest delay. These results coincide with those in [10] where the delay for both PQ and WFQ was seen to be almost zero with FIFO delay being the highest.

b) Traffic Sent vs Received

Almost the same amount of traffic was sent in all four queuing techniques with FIFO receiving the lowest amount compared to the other three which received close to the same amount. FIFO suffered the highest delay hence experiencing the most packet drops compared to the others. This therefore resulted in it having the lowest throughput. In [10]and [11], both scholars reported observing similar results except that both papers did not include custom queuing.

2) G729 Codec Scheme

a) Packet End-to-End Delay

The packet end-to-end delay results under the G729 codec scheme appeared to have a similar pattern to those of the G711 codec scheme with FIFO still having the highest delay while the rest of the queuing techniques experienced almost zero delay. The difference observed is that the delay experienced was slightly lower for the G729 codec. It can therefore be concluded that G729 performed better than G711 in terms of delay. G711 was expected to be the better of the two since it experiences no compression [21] as compared to G729 with a compression delay of 10ms [22], which adds to the end-to-end delay. Moreover, the literature also shows that G711 requires a bandwidth of 64kbit/s, higher than that of G729 which is 8kbit/s. Therefore, a lower bandwidth may result in congestion causing some packets to be dropped and high packet losses worsen end-to-end delays.

b) Traffic Sent vs Received

Because FIFO experienced a slightly lower delay under the G729 codec scheme, it also experienced a higher throughput even though it was still the lowest among the queuing techniques. The rest of the queuing techniques also had higher throughputs under this codec scheme than G711. These observations led to a conclusion that G729 performs better than G711 for VoIP supporting the observations made in [23] where a comparative analysis of these codec schemes was done.

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