PERFORMANCE ANALYSIS OF VoIP CODEC SCHEMES AND QUEUING TECHNIQUES AND THEIR IMPACT ON FTP AND VIDEO CONFERENCING

BY

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DECLARATION

I, **Emma Munthali** do declare that this dissertation is entirely my own work except as specified in the acknowledgements and that neither the dissertation nor the original work contained herein has been submitted to this or any other institution for a higher degree.

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APPROVAL

This dissertation by Emma Munthali entitled ‘Performance Analysis of VoIP Codec Schemes and Queuing Techniques and their Impact on FTP and Video Conferencing’ is approved as partially fulfilling the requirements for the award of the degree of Master of Telecommunication Systems of the University of Zambia.

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ABSTRACT

Advancement in internet technology allows for the integration of network traffic 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 Voice over Internet Protocol (VoIP) and video require certain Quality of Service (QoS) from the network which cannot be guaranteed on the internet therefore, maintaining the right QoS parameters becomes all the more important. Studies have been carried out on the effect of congestion management tools while others compare the effects of such tools on the quality of VoIP. However, in reality, these tools are not used in isolation in a network and most networks do not support single traffic only. The goal of this research therefore is to compare the effects of the combinations of some of these tools (i.e queuing techniques and codec schemes) on the quality of VoIP and to determine their influence on the rest of the traffic on a network. Simulation approach using the OPNET Modeler 14.5 tool was 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, G711 and G729 codec schemes and queuing techniques namely; First in First out (FIFO), Priority Queuing (PQ), Custom Queuing (CQ) and Weighted Fair Queuing (WFQ) were tested resulting in different scenarios. Custom Queuing technique showed the best performance overall except for the amount of video conferencing traffic received where it had the worst performance. WFQ and FIFO suffered the highest delay for video and VoIP traffic respectively while FTP traffic suffered from starvation in PQ. 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 produced higher throughputs and slightly lower delays compared to G711. G729 would best be used for low bandwidth networks while G711 would be ideal for networks with higher bandwidths and where VoIP communication is the main priority.

Keywords: Codec schemes, Queuing Techniques, VoIP, Quality of Service, OPNET

DEDICATION

To my late father, Mr Philemon K. Munthali for believing in me and investing in my education.

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LIST OF ABBREVIATIONS

FTP	File Transfer Protocal
ACELP	Algebraic Code Exited Linear Prediction
ADPCM	Adaptive Differential Pulse Code Modulation
CBWFQ	Class Base Weight Fair Queuing
CQ	Custom Queuing
FIFO	First In First Out
GNU	Gnu’s Not Unix
GSM	Global Systems for Mobile Communication
GUI	Graphical User Interface
HTTP	Hypertext Transfer Protocal
IAT	Inter Arrival Time
IP	Internet Protocal
ISDN	Integrated Services Digital Network
ITU	International Telecommunication Union
LAN	Local Area Network
LLQ	Low Latency Queuing
LTE	Long Term Evaluation
MOS	Mean Opinion Score
MPLS	Multiprotocal Label Switching
MWRR	Modified Weighted Round Robin
NAT	Network Access Translation
NS	Network Simulator
OMNET	ObjectiveModular Network Testbed in C++
OPNET	Optimized Network Engineering Tool
OS	Operating System
PBX	Private Branch exchange
PC	Personal Computer
PCM	Pulse Code Modulation
PQ	Priority Queuing

PSTN	Public Switched Telephone network
QoS	Quality of Service
TCP	Transmission Control Protocal
UDP	User Datagram Protocal
UMTS	Universal Mobile Telecommunication
USD	Universal Scene Description
VoIP	Voice over Internet Protocal
VPN	Virtual Private Network
WAN	Wide Area Network
WFQ	Weighted Fair Queuing
WLAN	Wireless Local Area Network

CHAPTER ONE

INTRODUCTION

The integration of Voice over Internet Protocol (VoIP), File Transfer Protocol (FTP), and video conferencing technologies has significantly transformed the landscape of digital communication, offering flexibility and efficiency in transmitting voice, data, and video over the internet. However, the performance of these services is largely influenced by the underlying network protocols, codec schemes, and queuing techniques employed to manage and prioritize digital traffic. As such, the analysis of how different VoIP codec schemes and queuing techniques affect the performance of FTP and video conferencing is critical for optimizing network operations, thereby enhancing user experience. This chapter gives a brief introduction to Voice over Internet Protocol and how its quality is affected by codec schemes and queuing techniques. It also gives the objectives and significance of the research.

1.1 Background

Over recent years, there has been an increasing trend in the use of real-time voice communication over the Internet. 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 (PSTN) [1]. VoIP is a term used for a set of technologies used to control the delivery of voice information over the Internet. Various VoIP communication software products are already available on the internet such as Skype, Google Talk, and Windows Live Messenger [1]. The process of VoIP transmission involves the digitization of voice, the isolation of unwanted noise signals, and then the compression of the voice signal using compression codecs. After the compression, the voice is packetized and sent over an IP network. Each packet needs a destination address, a sequence number, and data for error checking. The signaling protocols are added at this stage to achieve these requirements along with the other call management requirements. When a voice packet arrives at the destination, the sequence number enables the packets to be placed in order and then the decompression algorithms are applied to recover the data from the packets. Synchronization and delay management are then done to make sure there is proper spacing [3]. The efficiency and quality of VoIP systems are influenced by several factors, including queuing techniques and codec schemes, which play important roles in shaping their performance.

VoIP has a set of parameters that defines its Quality of Service (QoS) such as end-to-end delay, jitter, packet loss, Mean Opinion Score (MOS), and throughput. Quality of Service is the measurement of the overall performance of a service. In packet-switched networks, it is the ability to control the combination of bandwidth, delay, jitter, and packet loss in a network to deliver a network service that ensures high voice call quality over VoIP traffic [2]. Jitter is the variation in time delay between consecutive packets over a network while the end-to-end delay is the time interval in which a packet travels from one node to another node in a network. Delay in a network must be controlled and minimized as VoIP is very sensitive to it [6]. Delayed packets may become useless after a pre-specified amount of time and may be dropped [5]. Packet loss is inevitable in IP networks and occurs for various reasons. It can occur when a router or switch works beyond capacity or when queue buffers overflow. Packet loss above a set threshold rate introduces audio distortions that cause a decrease in voice quality [7]. Internet traffic is said to be "best-effort" because it delivers packets as it receives them and does not guarantee delivery [4]. A simple way of reducing delays and jitter for a given packet stream is to prioritize it over all others, but this is not an acceptable solution since most networks are designed to serve all users equally well.

As the internet carries various types of traffic, it requires traffic management systems which include queuing. Each router in the network must implement some queuing discipline that governs how packets are buffered while waiting to be transmitted. These queuing disciplines can affect the performance of the applications and utilization of network resources [11]. They also affect the packet latency by determining how long the packets wait to be transmitted. Several papers have shown comparative studies of the different queuing mechanisms which are; Priority Queuing, Custom Queuing, Weighted Fair Queuing, First in First out, and Class-Based Weighted Fair Queuing [4, 12, 14, 15]. In this research, however, only the first four are considered. In priority queuing, traffic with high priority is served before that with low priority [15]. Custom queuing discipline

(CQ) reserves a percentage of an interface's available bandwidth for each selected traffic type. If a particular type of traffic is not using its shared bandwidth, then other types may use it [4]. Weighted fair queuing (WFQ) divides bandwidth among queues based on weights ensuring that all traffic is treated fairly according to its weight [17].

Voice codec is one of the most critical components of a VoIP system. Codec is a coder/decoder that converts an audio signal to a digitized version for transmission over the medium and then back into the original uncompressed version on the receiver side [12]. VoIP services are based on this concept [13]. Common VoIP codecs include G711, G729, G723 and G726. G.711 is a high-bitrate (64 Kbps) codec according to the International Telecommunication Union (ITU) standard. It gives the best call quality for VoIP on the basis that it uses no compression at all. As a result, its call quality is almost like that of a regular ISDN phone [19]. G.729 offers a good level of call quality at a low bit rate of 8Kbps (kilobits per second), which enables more calls for the same bandwidth as that of the G.711 Codec.

The study extends its scope to comprehend how variations in queuing techniques and codec selections affect network functionalities. Assessing their impact on FTP (a fundamental data transfer protocol), and video conferencing applications, it provides comprehensive insights into the broader implications of VoIP optimizations on other network-dependent services.

In this research, a network model is simulated where two of the codecs (G711 and G729) are used together with four of the queuing techniques (FIFO, CQ, PQ, and WFQ) to compare the performance differences of the QOS parameters (throughput and delay) in a network.

1.2 Statement of the problem

The growth of Internet technology together with the relatively low deployment cost of IP networks has pushed for an integrated single network for data, video, and voice access [1,2]. Voice over IP (VoIP) was established with the integration of Internet and communication technologies to reduce the cost of communication [3]. However, the integration of all types of traffic onto a single IP network created various technical challenges for the Internet community to address [40].

Real-time traffic services such as VoIP and video require a certain Quality of Service (QoS) from the network [4] which cannot be guaranteed in an IP network [40]. For this reason, research has been done on the deployment and maintenance of VoIP with different technologies applied to improve its quality on networks [6,7,21,9,40,42,44], which has led to a steady rise in the number of VoIP subscribers [18]. Studies have also been carried out to compare the effects of such methods /techniques on the quality of VoIP [2,11,13,41,43]. These studies however concentrated on singular technologies only and how they impact VoIP. Furthermore, the networks being investigated in the previous studies support different types of network traffic rather than being dedicated to VoIP only, it is therefore imperative to know how applying VoIP quality improvement methods affects the rest of the traffic in the network.

This study seeks to analyze and identify the optimal combinations of queuing techniques and codec schemes that enhance the quality and efficiency of VoIP while considering their influence on FTP and video conferencing, addressing the need for a comprehensive understanding of how these factors interact and affect overall network performance in multi-application environments.

1.3 Aim

To investigate how the choice of the VoIP codec schemes and queuing techniques used in a network affects the performance of VoIP, FTP, and video conferencing traffic through analysis of the quality of service parameters.

1.4 Research Objectives

1. Assess the performance of popular VoIP codecs (G.711, G.729) under various queuing techniques in the network
2. Assess how the selected codec schemes and queuing techniques affect File Transfer Protocol (FTP) and video conferencing performance in the network.

3. To propose optimization strategies for queuing technique configurations to enhance FTP and Video Conferencing performance.

1.5 Research Questions

1. How does each codec scheme (G.711, G.729) perform under varying queuing techniques concerning throughput and delay in VoIP transmissions?
2. How does the implementation of various VoIP codec schemes and queuing techniques impact the performance of FTP transfers and video conferencing applications over the same network?
3. What configurations of queuing techniques optimize the performance of FTP transfers and video conferencing?

1.6 Significance of the study

The findings of this study can have practical implications for network administrators, telecom companies, and businesses using VoIP services. They can be used as a guide in selecting the most suitable codec schemes and queuing techniques based on their specific requirements, balancing voice quality, bandwidth usage, and overall network performance. Eventually, the goal is to enhance the user experience in VoIP communications while maintaining the quality of other network services.

1.7 Research Structure

This research consists of six chapters. Chapter One introduces the research topic, outlines the research problem, states the objectives and the questions that guide the direction of the research, and also shows the importance of the study. Chapter Two highlights some literature of related works that gives necessary background information about the research area to gain a better understanding of the topic. It describes VoIP and shows how it is affected by quality of service parameters such as jitter, delay, and packet loss. Coding/decoding schemes and queuing techniques are also discussed.

Chapter Three discusses the methodology adopted for this research. It starts by first discussing the research approach i.e. simulation, the tool used, and then the methods.

Chapter Four presents the results obtained from each scenario of the simulation and analyses them in relation to the existing body of knowledge. Lastly, chapter Five provides the conclusion and recommendations of the research based on the literature review, findings, and analysis in sync with the research questions and objectives.

CHAPTER TWO

LITERATURE REVIEW

2.1 Introduction

Over the years, Voice over Internet Protocol (VoIP) has attracted the attention of the network engineering research and operation communities. This has also resulted in several studies being carried out on the performance of VoIP networks [8,10,20,23,37,38,4]. Voice over IP has become one of the most important technologies today with the growth in its usage as can be seen in Figure 2.1. With the benefits such as a significant reduction in communication costs, more and more organizations are adopting VoIP applications. However, the quality of the performance in IP networks is still not guaranteed. This chapter explains in detail some codecs as well as the queuing techniques used in VoIP communication. This literature review aims to ensure a thorough understanding of the topic and identify similar work done within the area. It is also to compare and critique previous findings. It covers literature of research done by other researchers which includes; the selection of appropriate voice compression and decompression (codec) schemes depending on the Quality of Service (QoS) of VoIP in different networks, comparative analysis of queuing systems, and how the choice of a queuing discipline can affect the applications and utilization of the network resources inrouters as well as the benefits of VoIP over the traditional PSTN.

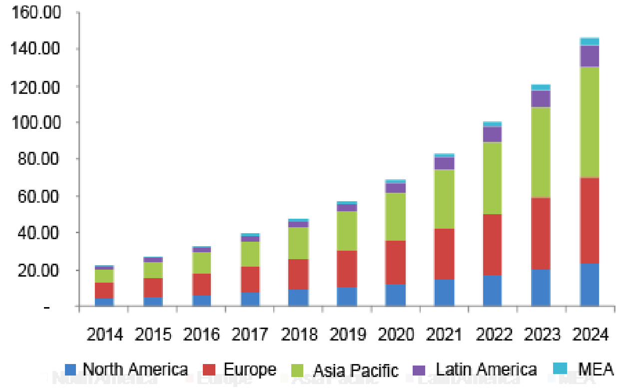


Figure 2.1: Global mobile VoIP market by region, 2014 -2024 (USD Billion) [18]

2.2 Overview of VoIP

VoIP is a telephone connection over the Internet. The data is sent digitally using the Internet Protocol (IP) instead of analog telephone lines, this allows people to talk to one another over long distances and around the world without having to pay long-distance or international phone charges [24]. A large number of factors are involved in making a high-quality VoIP call. These factors include the speech codec, packetization delay, packet loss, delay (coding, transmission, propagation, and queuing), delay variation, and the network architecture to provide Quality of Service. Other factors involved in making a successful VoIP call include the call setup signaling protocol, call admission control, security concerns, and the ability to traverse Network Access Translation (NAT) [38].

2.2.1 Benefits of VoIP over PSTN

Some of the advantages VoIP offers over PSTN include the following;

- With VoIP, calls are free no matter the distance if both parties use the same VoIP system, and Personal Computer to Phone call charges are nominal [24]. This is the biggest advantage to most companies. Typically, companies pay their normal high-speed Internet provider and in return receive VoIP free [31].
- VoIP accounts can easily be accessed from anywhere as long as there is internet connectivity. This makes it easy for those who travel frequently to make calls frequently to those back at home at local call rates, no matter where they are [24].
- Networking equipment is more scalable than equipment used for the PSTN since the latter consists of proprietary equipment (line cards, trunk cards, circuit switch cards, processor cards, etc) [26].

2.2.2 Challenges associated with VoIP

As with any new technology, VoIP introduces both opportunities and problems. Some of the problems are as follows:

- Although it offers lower costs and greater flexibility, VoIP also faces significant risks and vulnerabilities. Unfortunately, many of the tools used to safeguard today’s computer networks such as firewalls, network address translation (NAT) and encryption don’t work “as is” in a VoIP network [51]

- VoIP is dependent on broadband connection [53]. In a data network, congestion can cause latency, jitter, and even packet loss as packets are queued awaiting transmission. Bandwidth must therefore be properly reserved and allocated to ensure VoIP quality [1].
- Because so many nodes in a VoIP network have dynamically configurable parameters, intruders have as wide an array of potentially vulnerable points to attack as they have with data networks [51]
- VoIP units share physical network connections with the data network where VoIP and data are on the same logical portion of the network. Tools to monitor and control packet networks are widely available. Attaching a packet sniffer to the VoIP network segment makes it easy to intercept voice traffic making eavesdropping accessible to anyone with a PC and an Internet connection [51].
- VoIP is dependent on power: Traditional telephones operate on 48 volts and are supplied by the telephone line itself without any external power supply [1].

2.3 Queuing techniques

In a network, packets can accumulate and be queued into memory buffers of routers and switches. Packets may arrive at routers in bursts from multiple devices and a node may sometimes receive more packets than it can process. If this occurs, router buffers fill up and overflow mostly resulting in what is called tail drop which is simply the dropping of new incoming packets that do not fit in the buffer [27]. Queuing is a congestion-management mechanism used by routers to ensure the control of congestion by determining the order in which packets are transmitted based on priorities assigned to those packets [29]. In [35], queuing is defined as the main mechanism or method for planning how routers give priority to different incoming traffic. As a congestion management mechanism, it involves forming queues, assigning packets to those queues based on the classification of the packet, and then scheduling the packets in a queue for transmission [29]. Congestion may occur anywhere within a network mainly due to speed mismatches and accumulation that is caused by traffic movement from a high-speed LAN to lower-speed WAN and whenever multiple remote sites feed back into a central service site consecutively [35]. A queuing discipline (technique) manages access to a fixed amount of output port bandwidth by selecting the next packet to be transmitted on a port [28]. It plays an important role in a network's performance as it is a solution to the fair sharing of the available resources in a network [45].

There are two types of queues; hardware and software queues. A hardware queue holds the traffic and ready-to-send packets that it sends over a media interface while a software queue defines the memory of each interface. First In First Out (FIFO) is always a hardware queue [36] sometimes referred to as the transmit queue [32]. If the hardware queue is not congested or full, the packets are not kept in the software queue. They are switched directly to the hardware queue, where they are transferred quickly to the medium using a FIFO order. In a case where the hardware queue is full, the packets are held in the software queue, processed, and released to the hardware queue based on the software queuing discipline. The software queuing discipline could be Custom Queuing (CQ), Priority Queuing (PQ), Weighted Fair Queuing (WFQ), Class-Based Weighted Fair Queuing (CBWFQ), and Low Latency Queuing (LLQ).

A software queuing system schedules packets into the hardware queue based on the quality of service (QoS) requirements [29]. Only when the hardware queue is full does the software queue handle packets. Therefore, queuing configuration only takes effect during periods of interface congestion, when the hardware queue has overflowed. Several of the difficulties faced in networks including congestion are related to a limited amount of shared resources such as routers to competing users, applications, and service classes. Common causes for congestion include:

i. The speed of an ingress (inbound) interface is higher than that of an egress (outbound) interface. Congestion can occur whenever traffic moves from a high-speed LAN environment to lower-speed WAN links for example.

ii. The combined traffic of multiple ingress interfaces exceeds the capacity of a single egress interface. This happens in WANs when several remote sites feed into a central site. Congestion often occurs at the distribution layer of a LAN where the access layer devices feed traffic to the distribution layer switches.

iii. The switch or router's central processing unit is insufficient to handle the size of the forwarding table also results in congestion.

It is important to reduce congestion in a network as it reduces packet throughput, causes jitter, increases end-to-end delay, and can also lead to packet loss [28].

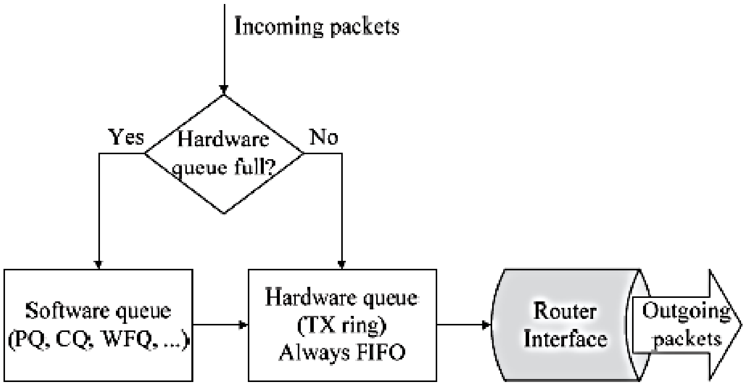


Figure 2.2: Queuing components [30]

2.3.1 First In First Out

First In First Out queuing popularly known as FIFO is one of the easiest queuing schemes. In FIFO queuing, the packet that comes first in the buffer is treated first [37]. FIFO queuing discipline places all packets it receives in a single queue, transmitting them as bandwidth becomes available. [45]. If the queue becomes saturated, new packets will be dropped. FIFO queuing is generally supported on an output port when there is no other queuing discipline configured [33]. Packets from different flows arrive at the router which multiplexes them into the same FIFO queue in the same order they arrived in [27]. The FIFO technique is extremely simple when compared to other queuing disciplines. It does not carry out packet reordering and the maximum queue delay can easily be calculated by taking into account the maximum length of a queue [27]**.** As long as the queue remains short, FIFO provides simple contention resolution for network resources without significantly increasing queuing delays at each hop [28]. It is not possible however to offer different services for different packet classes as all packets are put into the same queue and queuing delay increases as congestion increases which affects the performance of real time applications such as voice. A sudden burst in an incoming flow may cause an entire buffer space to be filled by this single flow not allowing other flows to be serviced until the buffer is emptied. During network congestion, FIFO benefits non congestion-responsive flows over congestion-responsive flows resulting in bandwidth starvation for the congestion-responsive flow as it decreases its sending rate once packet loss is detected. UDP on the other hand is unaffected by such an occurrence and continues to transmit at its normal sending rate [27].

Figure 2.3: First in First out Queuing

2.3.2 Priority Queuing

Priority Queuing (PQ) is a type of queuing mechanism that prioritizes traffic on output interfaces based on packet characteristics. This causes the router to place traffic into a different number of queues [29]. In [45], it is defined as a discipline that classifies all packets by the system placing them into different priority queues. Packets are transmitted in decreasing order of priority, meaning that the highest priority queue is emptied first, and then data on the next highest priority queue is transmitted, and so on [4]. Within each queue, packets are arranged in FIFO order [27]. Traffic is classified based on user-specified criteria as four output sub-queues; high, medium, normal, and low [4]. Packets without classification fall into the normal queue (no priority). Packets can be classified by the following packet characteristics which include; packet size, protocol type, or an access control list [29]. PQ uses tail-drop logic where new packets arriving in a particular queue that is full are dropped [30]. One benefit of PQ is that it places a relatively low computational load on the system when compared with other queuing disciplines [28]. The other benefit is that packets of different classes can be managed using different queues and thus a certain class of traffic can be handled differently from another [27].

Priority queuing seems suitable for time-sensitive services like VoIP however, choosing it as a queuing mechanism on a network device can affect low-priority traffic in a negative way [29].No packets from any other queue are processed for as long as there are packets in the high-priority queue which can result in complete starvation of low-priority traffic [32].

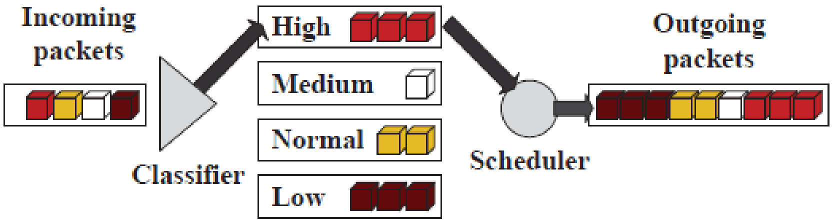


Figure 2.4: Priority Queuing

2.3.3 Weighted Fair Queuing

Weighted Fair Queuing (WFQ**)** dynamically creates queues based on traffic flows [32]. In Class-Based Weighted Fair Queuing, or CBWFQ, a form of WFQ, packets are reordered and latency is controlled at the edge and in the core. A switch or router manages buffering and bandwidth for each service class by assigning different weights to different service classes [17]. The primary goal of WFQ is to protect each service class by ensuring a fixed amount of output port bandwidth that is independent of the performance of other services [46]. In WFQ, all queues are served so that there is no bandwidth starvation as it ensures that a weight is assigned to each queue. However, some queues are given higher weights and therefore receive more service [27]. This weight determines the percentage of the link’s bandwidth that each flow gets. Weighted fair queuing can be thought of as round-robin mechanism starting with small packets [29]. In this technique, a finish time is assigned to each packet taking into account the link bandwidth, the number of queues, the weight of queues, and the packet length. Afterward, the scheduler serves the queue consisting of the packet with the minimum finish time. The finish time is used to build the order in which packets are to be transmitted through the link [27]. Because WFQ puts packets of different flows in different queues, it has a greater number of queues than all of the non-flow-based queuing techniques. The number of existing queues in the system is based on the number of active flows. By default, a maximum of 256 queues can exist, though this can be increased to 4096 [32]. Its scheduler uses different logic from that of other queuing tools to deal with the larger number of queues. It is complex as it requires a per-service class state and checking of all states for each packet arrival and departure.

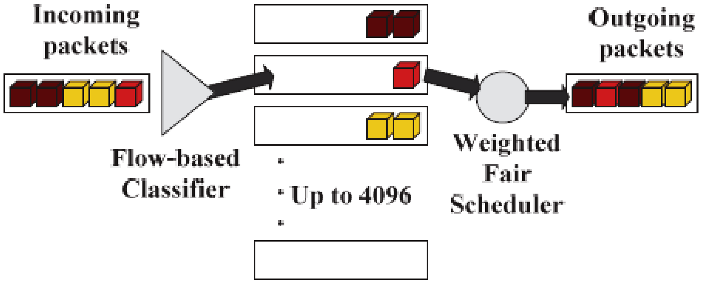


Figure 2.5: Weighted Fair Queuing

2.3.4 Custom Queuing

Custom queuing is a tool that allows classifying traffic into various queues based on the types of information that can be selected by any matching tool. These properties include transport or application protocol, port numbers, differentiated services code point or IP Precedence markings [29]. It is a less strict form of queuing which employs a weighed round-robin queuing methodology [32]. CQ addresses the biggest shortcoming of PQ ensuring a guaranteed minimum bandwidth to each queue, thereby queue starvation is avoided [30]. The queues are emptied one by one in a round-robin fashion, starting with the first queue. It assigns byte counters to every queue and takes packets from the queue until the total byte count that was specified for the queue has been met or exceeded [30]. Each queue is processed in order but each queue can have a different weight or size (measured either in bytes or the number of packets). The queuing technique supports a maximum of 16 queues [32].

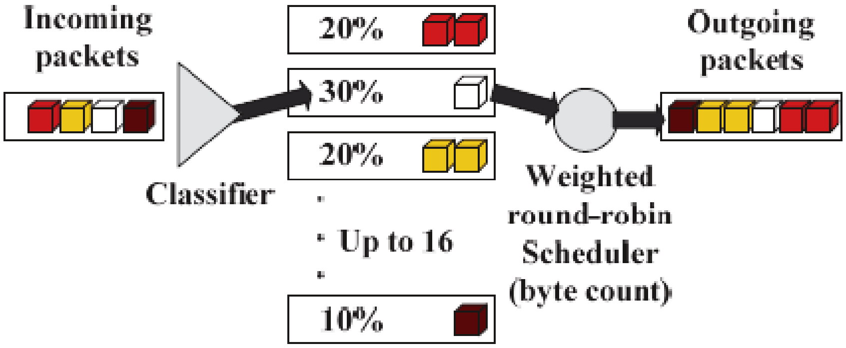


Figure 2.6: Custom queuing

Table 2.1 below shows a summary of the different features of the queuing techniques being studied in this research.

Table 2.1: Queuing protocol comparison [30]

Feature	FIFO	PQ	CQ	WFQ
Includes a strict priority queue		Yes		
Reserves bandwidth per queue			Yes	
Classifies based on flows				Yes
Maximum number of queues	1	4	16	4096

2.4 Codecs

A codec is a device or computer program for encoding or decoding a digital data stream or signal. Codec is a portmanteau of coder-decoder [16]. It converts the audio signal to a digitized version for transmission over the medium and then back into the original uncompressed version on the receiver's side. There are several codecs used for VoIP communication each having its bandwidth and characteristics such as sampling rate, bit rate, encoding algorithm, coding delay, sound quality e.t.c [13].

2.4.1 Classification of codec schemes

Codecs can be classified into audio, video, or text**.** The only basic difference between these is that they are designed to compress and decompress audio, video, and text files respectively.

2.4.2 Audio codecs

Audio codecs determine the sound quality of a call. High levels of compression cause higher compression delays compared with codecs having no or low compression. In addition, high compression levels also result in less bandwidth requirements, and accordingly a better execution during a system blockage [67]. Below are some of the International Telecommunication Union (ITU-T) standards for audio compression and de-commpression.

G.711 is an ITU-T standard for audio companding also called Pulse code modulation (PCM) of voice frequencies. The three main steps involved in the PCM are sampling, quantization, and coding.It is a required standard in many technologies such as in the H.320 and H.323 standards [21]. G.711 uses no compression and therefore takes up more bandwidth than other audio codecs of 64 kbit/s [47]. G711 has two versions called U-law (US, Japan) and A-law (Europe). U-law is concerning the T1 standard used in North America and Japan. The A-law is related to the E1 standard used in the rest of the world [25].

G.723.1 codec can compress voice signals at a low bit rate and is a very low-bandwidth codec as compared to others. This uses only 5.3 Kbps (based on ACELP) to 6.3 Kbps (based on MP-MLQ) of bandwidth. The voice quality is less than G.711 and is not recommended for fax services.

G.726 is an Adaptive Differential Pulse Code Modulation (ADPCM) based low bandwidth codec that uses 16, 24, 32, and 40 Kbps. Service providers can interchange ADPCM voice between packet voice and public phone or packet voice and PBX networks. The lower settings enable VoIP over dial-up links but the voice quality is still less than G.711 codec.

G.729 is an ITU-T recommendation for the coding of speech signals at an 8kbps data rate using Conjugate Structure-Algebraic Code Excited Linear Prediction (CS-ACELP), a speech coding algorithm in which a limited set of pulses is distributed as excitation to a linear prediction filter. It is a low bit rate and higher compression speech codec that is mostly used in applications when bandwidth is an issue for the VoIP implementation [37]. Data compression delay is 10ms for G.729, and the lower bandwidth of 8kbps leads to its use in VoIP applications easily. Other variants of G.729 are G.729.1, G.729A, and G.729B. G.729.1 enables scalable data rates between 8 and 32 kbps. It is a wideband speed and audio coding algorithm that is interoperable with G.729, G.729A, and G.729B codecs [25].

Table 2.2: Audio codecs [21]

ITU No.	Rate (kbit/s)	Algorithm	Frame Length
G711	64	Pulse Code Modulation	0.125ms
G723.1	6.4/5.3	Algebraic code-excited linear prediction	30ms
G726	40/32/16	Adaptive Differential PCM	0.125ms
G729	8	Conjugate Structure-Algebraic Code Excited Linear Prediction	10ms

2.5 Quality of Service

Quality of Service (QoS) is the ability to guarantee a certain level of performance to traffic flow in a network by providing different priorities to different applications, users or data flows [48]. It enables the control of the quality of data transmitted in networks and improves the organization of data traffic flows that go through many different network technologies. Network equipment companies integrate QoS mechanisms into routers and switches which provide priorities to one or more data streams simultaneously and also ensure the existence of all remaining (lower-priority) data streams [33]. QoS also provides a mechanism to drop lower-priority traffic before higher-priority traffic, during periods of congestion [32]. VoIP QoS is measured according to ITU recommendations based on different parameters like (delay, jitter, and packet loss), which can be changed and controlled within the acceptable range to improve VoIP quality [34].

2.5.1 QoS Parameters

VoIP QoS is measured according to ITU recommendations based on different parameters like delay, jitter, and packet loss. Voice packets should not be dropped, excessively delayed, or have a high jitter for a good-quality call as VoIP is extremely sensitive to these factors [49].

a) Jitter

Jitter is defined as the inconsistency in packet arrival at the destination [50]. In [49]**,** it is defined as the difference between the time a packet is expected to arrive and the actual arrival time. Voice media is transported by Real-Time Transport Protocol developed based on User Datagram Protocol therefore packets received out of order can't be reassembled at the transport level. Reordering is done at the application level, introducing significant overhead. Even when packets arrive in order, high jitter causes them to arrive at their destination in spurts [51]. When voice packets are transmitted from source to destination over IP networks, packets may experience variable delay, called jitter. If the packet inter-arrival time (IAT) on the receiver side is not constant even if the packet inter-departure time (IDT) on the sender side is constant, the result is packets arriving at the destination with varying delays (between packets) referred to as jitter [52].

$Jitter_n = |\Delta Arrival_n - \Delta Arrival_{n-1}|$, [36]

where n is the current packet

b) Delay/ latency

Latency or delay is defined as the average "travel" time taken by a packet to pass through a network from source to destination [50]. In [69], Packet end-to-end delay is defined as a measure of the time difference between packet departure from the source and arrival at the preferred destination. Delay is said to be a major source of voice quality degradation as the network grows and the load on a network increases [36]. Latency turns traditional security measures into double-edged swords for VoIP in that tools such as encryption and firewall protection can help secure the network while also introducing significant delay [51]. Packet end-to-end delay is measured by calculating the delay in the arrival of voice packets from the speaker to the receiver including the compression and decompression delays along the media path and signal path [68]. It can be calculated as follows;

$D_{E2E} = D_n + D_e + D_d + D_c + D_{de}$ [68]

Where Dn, De, D_d, Dc, and D_{de} represent the network, encoding, decoding, compression and decompression end-to-end delay respectively.

Table 2.3 shows different classifications of Delay and Jitter in milisconds.

Table 2.3: QoS Parameters [35]

Network parameter	Good	Acceptable	Poor
Delay (ms)	0-150	150-300	>300
Jitter (ms)	0-20	20-50	>50

c) Packet loss

Packet loss is the percentage of undelivered packets in the data network [50]. It describes the packets that don't reach the required destination [69]. This happens when any incoming data is dropped at any time by a device such as a router, switch, or link [35]. Packet loss can occur in the network or at the receiver side for example, due to excessive network delay in case of network congestion [52]. VoIP is sensitive to packet loss; it is therefore very important not to have lost packets while transmitting voice signals to have reasonable voice quality [49].

$$Packet\ Loss = \frac{Sent\ packets - Received\ packets}{Sent\ packets} \times 100 \quad [38]$$

2.6 FTP and Video conferencing traffic

File Transfer Protocol (FTP) is a standard network protocol used to transfer computer files from one host to another host over a TCP-based network, such as the Internet [63]. Two important things that exist in FTP are an FTP server and FTP Client. An FTP server is software used to exchange files and is always ready to give a service whenever a request is made from an FTP client. An FTP client is a computer that requests a connection to an FTP server to exchange files (uploading or downloading files) [64]. The protocol employs acknowledgments that guarantee the delivery of data at a destination. In case acknowledgments are not received till the timeout period, retransmissions are made to ensure the delivery of data to the receiver [63].

Video conferencing is the transmission of live video images and audio between two or more different participants. Conversations may be one-to-one (point-to-point) or one-to-many (multipoint), in simplex (one-way only), half-duplex (one way at a time, taking turns), or full-duplex (all parties are seen and heard simultaneously) [65]. Video conferencing needs considerable amounts of bandwidth, a need which can be balanced by codecs through the increase in the compression level. However, even for the latest generation of codecs, the minimum level required for an acceptable quality can produce a rather huge stream of bytes that must be transmitted through the network [66].

2.7 Simulation Tools Review

Researchers have used the simulation technique to develop new networks or test, modify, and optimize existing ones. The simulation process creates a theoretical representation of an already existing or proposed system to identify and understand the controlling factors of the system and to predict the behavior of the system. 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 [8]. Sometimes networks may contain a large number of network nodes and services making it time-consuming and costly to establish physical networks. Network simulation is one of the leading evaluation methodologies in the study of telecommunication/computer networks. Musa et al [60] noted that network simulation is critical during the development of new network protocols and communication architectures.

Different simulation tools are available and can be used for simulating networks. Some of the popular tools are;

2.7.1 Network Simulator (ns-3)

This is an open-source discrete-event network simulator licensed under the GNU GPLv2 license, released in 2006 as a replacement for the previous NS-2 [61]. The ns-3 project's goal is to build a solid simulation core that is well-documented, easy to use, and debug [62]. It provides a user-friendly graphical user interface that has been coded in C++ and Python language and can run under most modern operating systems [61]. The ns-3 simulation supports both IP and non-IP-based networks with the large majority of its users focusing on wireless/IP simulations such as models for Wi-Fi, WiMAX, or LTE [62].

2.7.2 OMNET++:

OMNeT++ is an extensible, modular, component-based C++ simulation library and framework, primarily for building network simulators. It stands for Objective Modular Network Testbed in C++. OMNeT++ is not a network simulator itself however it has gained popularity as a network simulation platform within the scientific community. The OMNeT++ simulation kernel runs on all platforms where a modern C++ compiler is available.

2.7.3 Optimized Network Engineering Tool (OPNET) Modeler:

The Optimized Network Engineering Tool (OPNET) simulator is an authoritative communication system simulator developed by OPNET Technologies [9]. It has several models for network elements, and it has many different real-life network configuration capabilities. The OPNET also includes features such as a comprehensive library of network protocols and models, a user-friendly Graphical User Interface (GUI), and data collection and analysis (graphical results and statistics) [1]. The simulator has gained considerable popularity in academia as it is being offered free of charge to academic institutions [6].

2.8 Related literature

S. Dhanalakshmi et al [10], investigated the performance of VoIP over Ethernet LAN in a Campus Network. The purpose of this project was to conduct several test cases in VoIP by constructing different simulation scenarios under OPNET 14 Software. A successful implementation of the project would reflect the advantage of VoIP over the traditional PSTN thus proving that VoIP would be an ideal candidate for modern technology in network communication. It should however be mentioned that traditional PSTN offers stability and reliability in connectivity and might still be used alongside VoIP.

S. Jalendry [21] undertook a detailed review of VoIP by giving an introduction to the technology: the network structure, protocols, echo and delay, jitter, and packet loss in VoIP network. In [6], a detailed simulation approach for deploying VoIP successfully was presented using the OPNET simulation tool. The paper described the modeling and representation of background and VoIP traffic, as well as various simulation configurations. It was necessary to help network researchers and designers determine how well VoIP would perform on a network before its deployment.

K. Balasundaram [12] gives a comparative analysis of three queuing techniques namely; FIFO, PQ, and WFQ. Three simulation models of network topologies were tested to evaluate 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. The simulation results showed that the WFQ technique has a superior quality over the other techniques.

In [4], two VoIP service classes, one being excellent effort and the other interactive voice were studied for various scheduling disciplines including First in First out (FIFO), Priority Queue (PQ), Custom Queue (CQ), and Modified Weighted Round Robin (MWRR). The parameters considered for evaluation were packet end-to-end delay (sec) and packet delay variation. It was concluded that PQ performed the best with interactive voice while CQ gave the best results for excellent effort service.

M. Gospodinov [14] researched the effects of different queuing disciplines on packet delivery for three applications: FTP, Video, and VoIP using OPNET. This paper presented how the choice of the queuing discipline can affect the applications and utilization of the network resources in the routers. PQ and WFQ were found to be most appropriate for voice, and WFQ for video while the results for FTP were not presented.

S. Gurrapu [2], evaluated the performance analysis of VoIP using various codecs such as G.711, G.723, and G.729 over Wi-Fi networks while increasing the number of nodes.

Performance metrics such as Mean Opinion Score (MOS), average end-to-end latency and disconcert were evaluated and discussed. In this research, the G.711 codec was shown to be 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.

M.A. Mohamed [3] evaluated the performance of different VoIP codecs over the WiMAX network. The performance of these codecs namely; G711, G723, G726, G728, and G729 were evaluated using the 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.

In [13], the focus was on the selection of an appropriate voice codec scheme depending on the Quality of Service (QoS) of VoIP in different networks. Wired, Wireless Local Area Networks (WLAN), Worldwide Interoperability for Microwave Access (WiMAX), and Universal Mobile Telecommunication System (UMTS) networks were implemented in OPNET Modeler. The quality was compared using different QoS parameters like end-to-end delay, MOS, throughput, and jitter. The VoIP codecs used in the measurements of QoS were: GSM-FR, G.711, G.723.1, and G.729A. Simulations results showed that G.711 and GSM-FR were the best schemes to provide high-quality voice in Wireless Local Area Network (WLAN) while WiMAX performed well with G.729A concerning all the parameters. G.723.1 performed well in WiMAX and UMTS networks depending. The wired network model gave the best results regardless of the codec being used.

A.M. Alsahlany in [22] evaluated the QoS performance for VoIP traffic under various voice codecs (G711, G723.1, G729) when integrating a wireless local area network with a wide area network. G729 was seen to perform better compared to the other two as it gave lower values of jitter and delay.

A. Ahmed [20] studied the performance of different scheduling schemes like FIFO, PQ, and WFQ for different codec formats namely; G711, G729, and G723 in a wide area network. All three scheduling schemes 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. However, the results produced in the graph for end-to-end delay were in seconds while the researcher recorded them to be in milliseconds without converting them.

2.9 Gap in the literature

Previous studies have focused on either codec schemes or queuing techniques only and their effects on the quality of VoIP. The study carried out in [20] investigated the effects of applying both technologies onto VoIP traffic only. Data networks usually carry different types of traffic therefore introducing technologies to alter the performance of one type of traffic might affect the other types of traffic too. The purpose of this study is to investigate the effects of applying both technologies (codec schemes and queuing techniques) to a wired data network on VoIP quality and the rest of the traffic on the network i.e. FTP and video conferencing traffic in this case. This presents a network operator with more thorough information to assist with decision-making depending on the priority usage of the network.

2.10 Summary

The literature reviewed shows how Voice over Internet Protocol is affected by the choice of codec schemes and queuing techniques through the measurement of different performance metrics. The chapter also delved into the codecs schemes that are used for VoIP, the queuing techniques as well as the popular simulation tools available for simulating networks.

CHAPTER THREE

RESEARCH METHODOLOGY

3.1 Introduction

Research methodology is a plan of how knowledge would be known and involves specific methods [56]. In the previous chapter, literature was reviewed aimed at explaining in detail some of the codecs as well as the queuing techniques used in VoIP communication. In this chapter, a description of the research process and an explanation of the methods used to gather and evaluate the collected data is given. The results were collected in the form of graphs and summarised in tabular forms.

3.2 Research design

A research design is a procedural plan, structure, and strategy used by a researcher to answer research questions objectively [54]. In this research, the comparative experimental design was adopted as it seeks to compare the effects (QoS parameters) of different treatment modalities (VoIP codec schemes and queuing techniques) on the study population (network).

3.3 Research Approach

This research involves an investigation of a network supporting both real-time and non-real-time network traffic. Emphasis is put on VoIP and how efforts to improve its quality affect the other traffic in the network. Traffic management techniques will be applied to VoIP only while maintaining the initial settings of the rest of the traffic. In reality, however, traffic management practices can be applied to the rest of the traffic in the network too.

Experimenting on or creating a real (existing) network would be challenging, expensive, and time-consuming as it implies the following;

a) It would disturb the normal operation of a network

b) Would require acquiring permission from the network

c) Would involve too many variables resulting in difficulties in understanding the contribution of each factor

For these reasons, a study in a controlled environment is necessary. A quantitative research methodology was therefore adopted and simulation was used to provide the controlled environment needed. The results collected were presented in graphical form.

3.4 Simulation

In this research, the OPNET Modeler 14.5 was used to simulate a model of the network under investigation. This was due to it being readily available and the researcher being already familiar with it.

3.4.1 Network Model Description

The type of network chosen for this research is a wired local area network supporting three different types of network traffic namely: FTP traffic, Video conferencing traffic, and VoIP traffic. The network was set up as shown in Figure 3.1. It consisted of two routers connected with each router connected to three end-user nodes. One of the routers is connected to VoIP, Video conferencing, and FTP clients while the other is linked to VoIP and video conferencing clients and an FTP server.

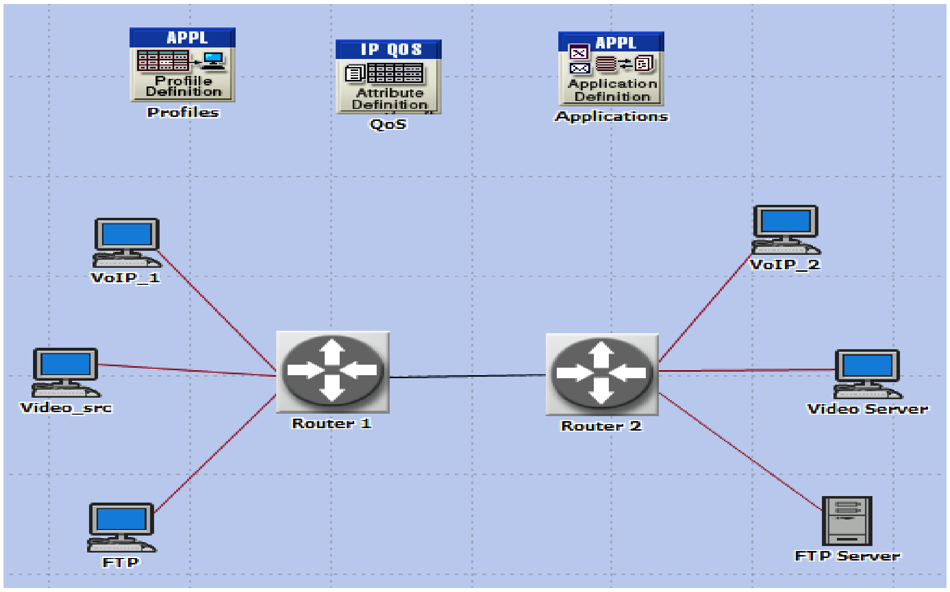


Figure 3.1: Network model

3.4.2 Network scenarios

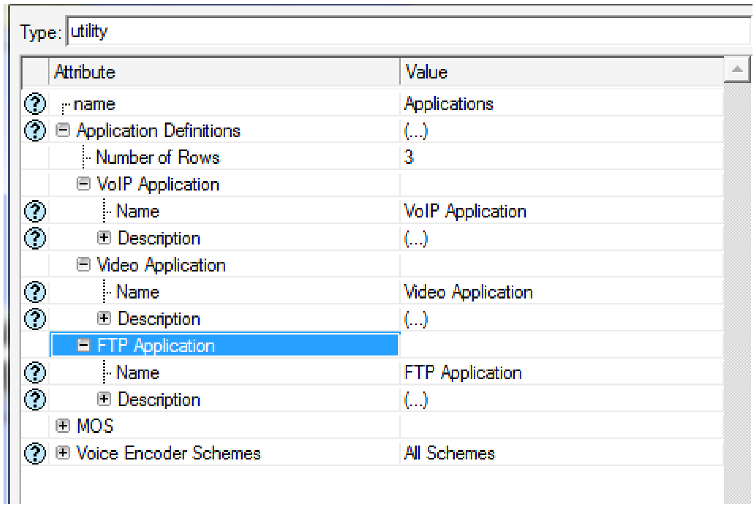
While maintaining the same topology and traffic of the network, different types of codec schemes and queuing techniques were tested resulting in different scenarios. Two codec schemes have been used with four different queuing techniques creating eight scenarios in the simulator. These scenarios can be grouped into two according to the two schemes as shown below.

$$\text{G711} + \left\{ \begin{array}{l} \text{FIFO} \\ \text{PQ} \\ \text{WFQ} \\ \text{CQ} \end{array} \right. \qquad \text{G729} + \left\{ \begin{array}{l} \text{FIFO} \\ \text{PQ} \\ \text{WFQ} \\ \text{CQ} \end{array} \right.$$

3.4.3 Network Configuration

Some important objects in OPNET are used to simplify working with it. In this research, the objects that were used are; Application Configuration, Profile Configuration, and QoS Attribute Configuration as shown in Figures 3.2, 3.3 and 3.4 respectively. Their functions are as follows:

a) Application Configuration Object: This is an object used to define and configure all Applications in the network according to the user requirements. OPNET has most of the common applications like HTTP, E-mail, video, File transfer, Voice, and database. It also specifies encoder Parameters for each of the encoder schemes used for generating voice traffic in the network.

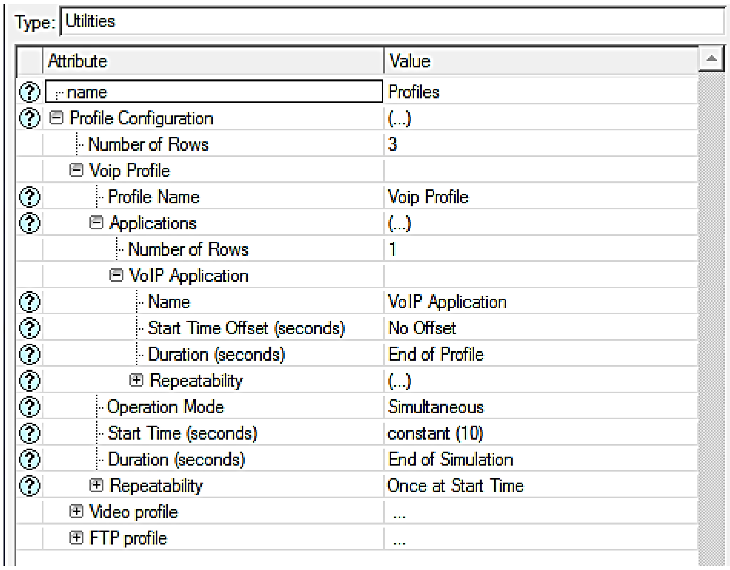


Type: utility

Attribute	Value
name	Applications
⊟ Application Definitions	(...)
Number of Rows	3
⊟ VoIP Application	
Name	VoIP Application
⊞ Description	(...)
⊟ Video Application	
Name	Video Application
⊞ Description	(...)
⊟ FTP Application	
Name	FTP Application
⊞ Description	(...)
⊞ MOS	
⊞ Voice Encoder Schemes	All Schemes

Figure 3.2: Configuration of the Applications

b) Profile Configuration Object: The "Profile Config" node can be used to create user profiles. These user profiles can then be specified on different nodes in the network to generate application layer traffic. The application defined in the "Application Config" objects are used by this object to configure profiles. Therefore, you must create applications using the "Application Config" object before using this object.

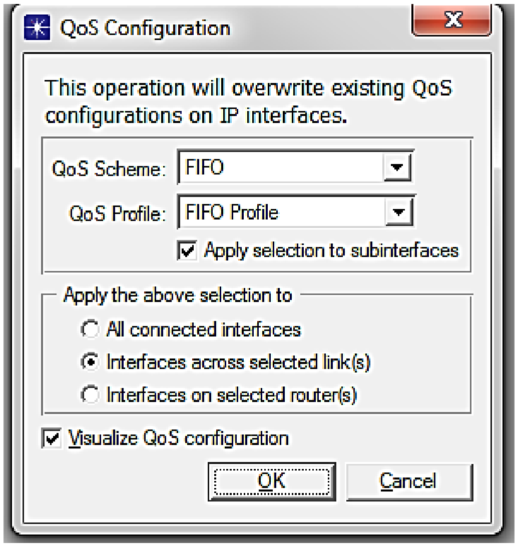


Type: Utilities

Attribute	Value
name	Profiles
⊟ Profile Configuration	(...)
Number of Rows	3
⊟ Voip Profile	
Profile Name	Voip Profile
⊟ Applications	(...)
Number of Rows	1
⊟ VoIP Application	
Name	VoIP Application
Start Time Offset (seconds)	No Offset
Duration (seconds)	End of Profile
⊞ Repeatability	(...)
Operation Mode	Simultaneous
Start Time (seconds)	constant (10)
Duration (seconds)	End of Simulation
⊞ Repeatability	Once at Start Time
⊞ Video profile	...
⊞ FTP profile	...

Figure 3.3: Profile configurations

c) The QoS Attribute node: A node that defines attribute configuration details for protocols supported at the IP layer. These specifications can be refrenced by the individual nodes using symbolic names.


QoS Configuration

This operation will overwrite existing QoS configurations on IP interfaces.

QoS Scheme: FIFO

QoS Profile: FIFO Profile

☑ Apply selection to subinterfaces

Apply the above selection to

○ All connected interfaces

● Interfaces across selected link(s)

○ Interfaces on selected router(s)

☑ Visualize QoS configuration

Figure 3.4: QoS configuration in OPNET

3.4.3.1 Application and Profile Configuration

Three applications were configured namely; the FTP application, the the Video application, and the Voice application as shown in Table 3.1. Three profiles were each set to run simultaneously to represent a busy period in the network.

Table 3.1: Application Definition

Application/ Profile	Load Description	Type of service
FTP	High Load	Best Effort
Video conferencing	Low Resolution	Streaming Multimedia
Voice	PCM quality speech	Interactive Voice

3.4.3.2 QoS Attribute Node

The queuing profiles that are defined in the QoS node were kept in their default settings as shown in Table 3.2. The type of queuing technique used for each scenario was configured by carrying out a QoS configuration on the link connecting the routers.

Table 3.2: Queuing profiles in the QoS node

Queue Profile	No. of queues available in OPNET	Maximum queue size (pkts)	Determination of the No. of packets transmitted
FIFO	1	500	First in first out basis
PQ	4	80(Low),60(Normal), 40(Medium),20(High)	Based on the priority of the queue
CQ	8	20 per queue	Based on the byte count of the queue
WFQ	8	500 per queue	Based on the weight given to each queue

Table 3.3: Summary of the simulation process

Steps	**Details**
Create and Configure the Network	The Application Config, Profile Config, QoS Attribute Config, five ethernet_wkstn, one ethernet_server, and two ethernet4_slip8_gtwy routers were selected from the object palette and added to the workspace. Network devices were connected as shown in Fig 3.1. The network had 3 applications (VoIP App, Video App, FTP app) and 3 profiles (one profile for each application) configured. Configuration of all nodes was done according to the desired settings.
Choose the Statistics	The QoS parameters to be tested were selected. These were packet end-to-end delay, traffic sent, and traffic received.
Configure the Simulation	After configuring the network, the project was saved and the simulation duration was set to run for 300 seconds.

Duplicate the Scenario	To analyze the effect of different queuing disciplines and codecs, more scenarios were created by duplicating the existing ones and making appropriate changes.
View the Results	After running the simulation and saving the project, the results were viewed and exported to Microsoft Excel for analysis.

3.5 Assumptions

The following assumptions were made

1. The 3 traffic flows represent the many different network traffics in real networks
2. In reality, it is possible to have periods where some of the traffic is not being sent. This experiment however focuses on the 5 minutes when all traffic is active.
3. Only codecs and queuing techniques are under consideration although in reality network performance is affected by many more factors such as bandwidth availability, type of signaling being used, number of users etc.

3.6 Data Analysis

The data analysis process involved the following:

a) Generating graphs in OPNET and exporting to Microsoft Excel Simulation involves the generation of results which were in graphical form.
b) Calculating averages (mean values) using Microsoft Excel
c) Calculating the throughput in percentage for each scenario

The throughput percentage was calculated using the formula below:

$$Throughput(\%) = \frac{\#\ of\ Received\ pkts}{\#\ of\ Sent\ pkts} \times 100$$

Where '# of Received pkts' means the average number of packets received and '# of Sent pkts' means the average number of packets sent

d) Comparing average measured values to maximum acceptable values

   The mean (average) of the collected results were compared with the maximum acceptable values by the International Telecommunication Union using a table.

e) Also comparing results to those in the literature reviewed

f) Stating and explanation of main observations made

3.8 Summary

This is an experimental research where a simulation method was used to collect results. Simulations depict real-life networks allowing the study to proceed without the costs of acquiring actual equipment. The results collected were exported to Microsoft Excel before the analysis.

CHAPTER FOUR

RESULTS AND ANALYSIS

The previous chapter discussed the research methodology adopted for this study to satisfy the research objectives as set out in chapter one, the research approach and research design. This chapter presents the outcomes of the study in line with the research questions. The findings have mainly been presented in the form of graphs and tables.

The queuing techniques chosen for this study are the four most common queuing techniques studied in the literature reviewed [4,11,12,14,30,33,34,45] and are all found in the OPNET modeler. Two codecs namely; G711 and G729 were selected representing two different sides of the spectrum. The no-compression G711 codec scheme requires a high bandwidth of 64kbit/s [47] and G729 a low bit rate, higher compression codec with a bandwidth of 8kbit/s [37].

4.1 Performance of Popular Voip Codecs (G.711, G.729) Under Various Queuing Techniques in a Network.

4.1.1 VoIP Traffic under G711

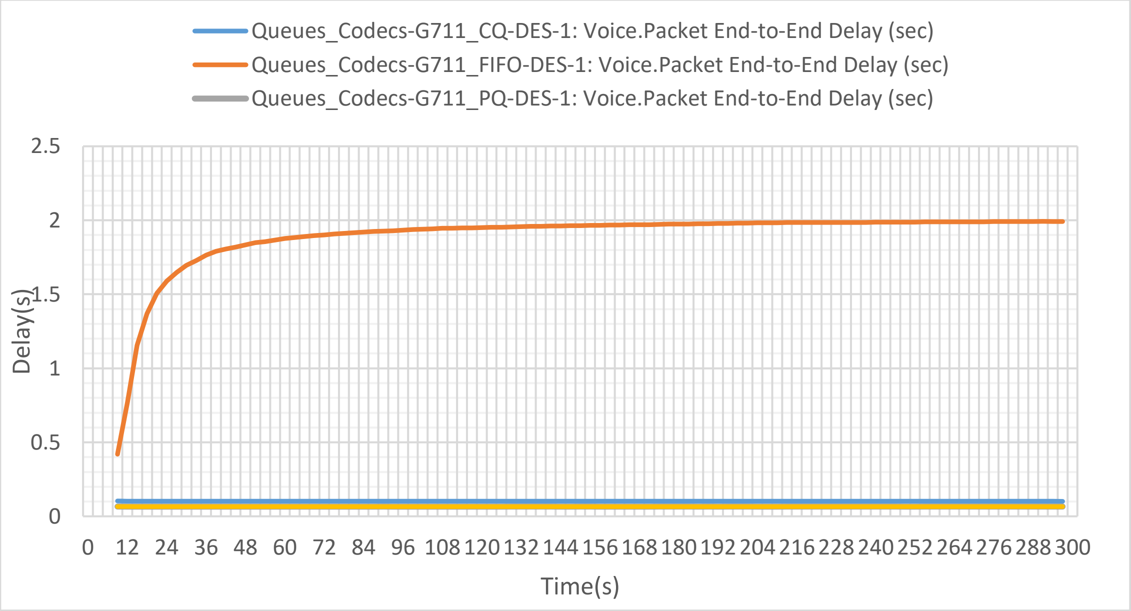


Figure 4.1: Voice packet End-to-End Delay

PQ and WFQ were observed to have suffered the least voice packet end-to-end delay while FIFO experienced the highest as seen in Figure 4.1. 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 [29,32]. This significantly reduces the delay between the transmitter and the receiver. In FIFO, all packets are put into the same queue, and 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 [12] where the delay for both PQ and WFQ was seen to be almost zero with FIFO delay being the highest.

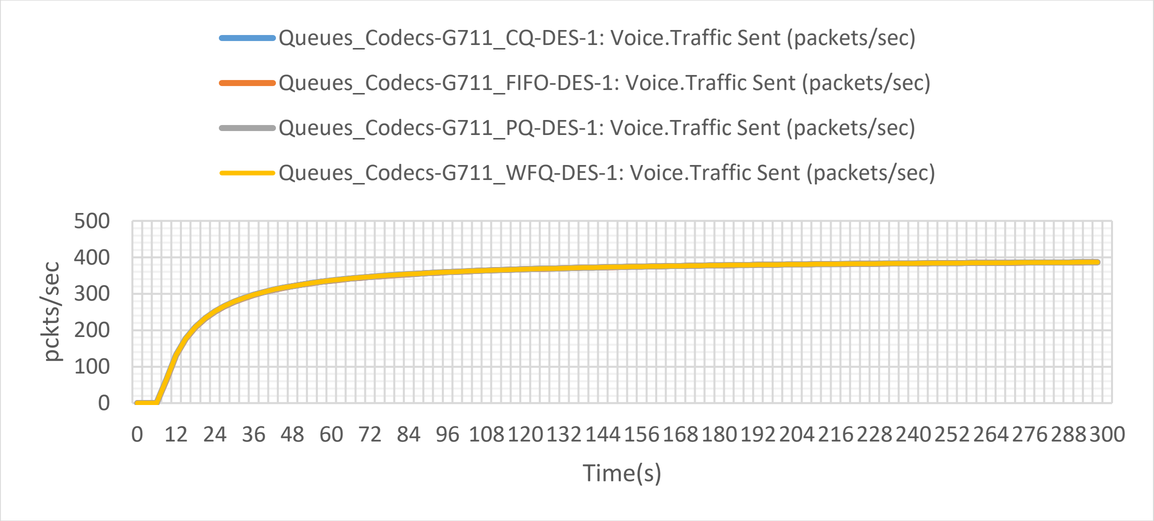


Figure 4.2: Voice traffic sent

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 Figure 4.2. This is because the graph is in an 'overlaid statistics' presentation.

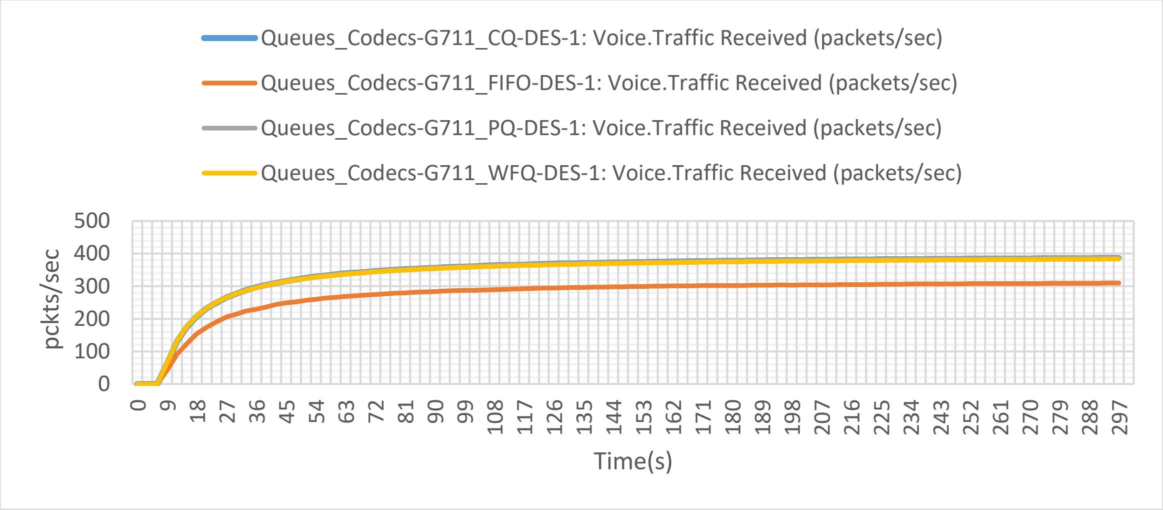


Figure 4.3: Voice traffic received

In Figure 4.3 CQ, PQ, and WFQ all received the same number of voice packets per second and therefore have their curves overlaid with one another. FIFO received the lowest amount of voice traffic in this scenario.

Table 4.1: Summary of Voice traffic parameters (Average) under G711

Type of queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. Traffic sent	386.596	386.56	386.633	386.633
Av. Traffic received	386.566	309.903	386.62	383.403
Throughput	99.90%	80.20%	99.90%	99.20%
Delay	0.101	1.893	0.066	0.066

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 as can be seen in Table 4.1. This may be since 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 [12] and [14], both scholars reported observing similar results except that both papers did not include custom queuing.

4.1.2 Voice Traffic under G729

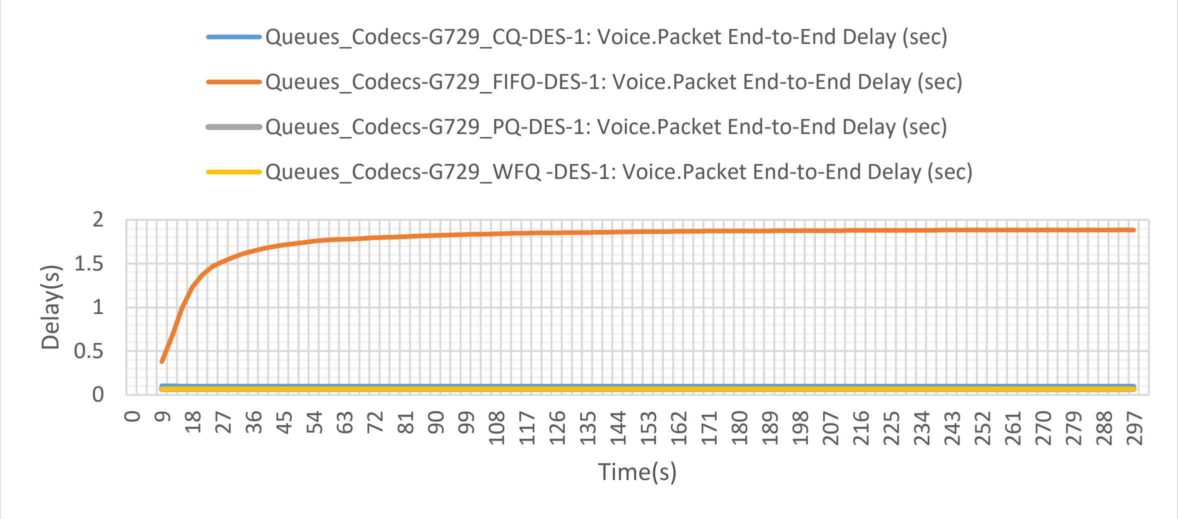


Figure 4.4: Voice packet end-to-end delay

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 as shown in Figure 4.4. 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 [47] as compared to G729 with a compression delay of 10ms [37], 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.

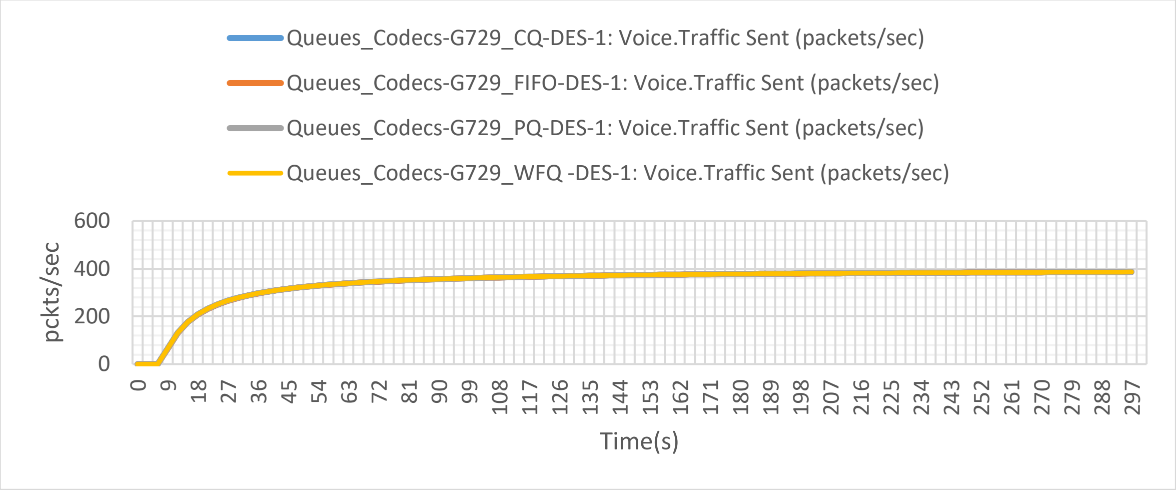


Figure 4.5: Voice traffic sent

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.

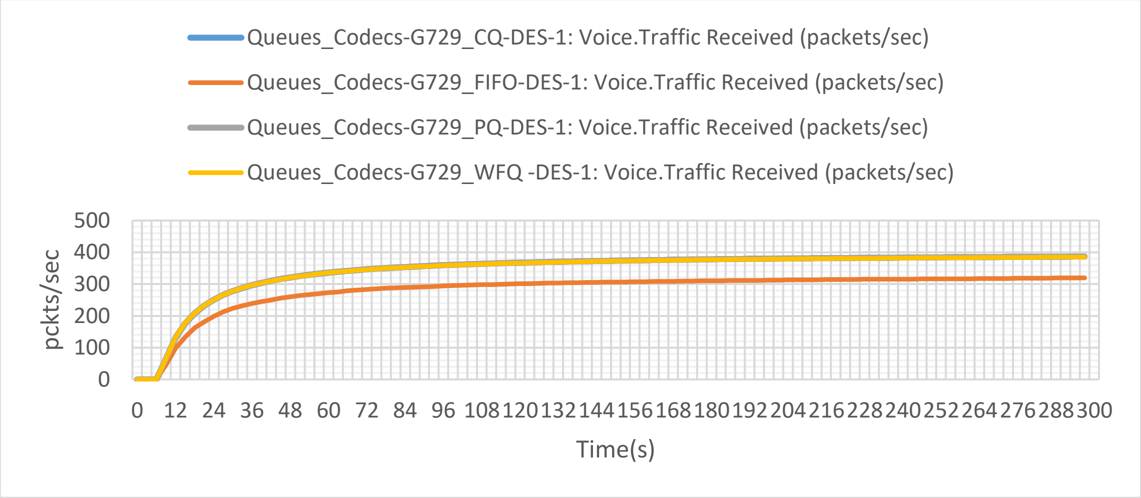


Figure 4.6: Voice traffic received

The same amount of voice traffic was received in CQ, PQ, and WFQ while FIFO received less as shown in Figure 4.6.

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 [13] where a comparative analysis of these codec schemes was done.

Table 4.2: Summary of Voice traffic parameters (Average) under G729

Type of queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. Traffic sent	386.59	386.55	386.62	386.62
Av. Traffic received	386.51	319.45	386.61	384.99
Throughput	99.98%	82.60%	99.90%	99.60%
Delay(s)	0.1004	1.787	0.066	0.065

4.2 Effects of Selected Codec Schemes and Queuing Techniques on File Transfer Protocol (FTP) And Video Conferencing Performance in the Network.

4.2.1 FTP Traffic under G711

Figures below show the amount of File transfer protocol (FTP) traffic that was sent and received under G711 for FIFO, PQ, WFQ, and CQ respectively.

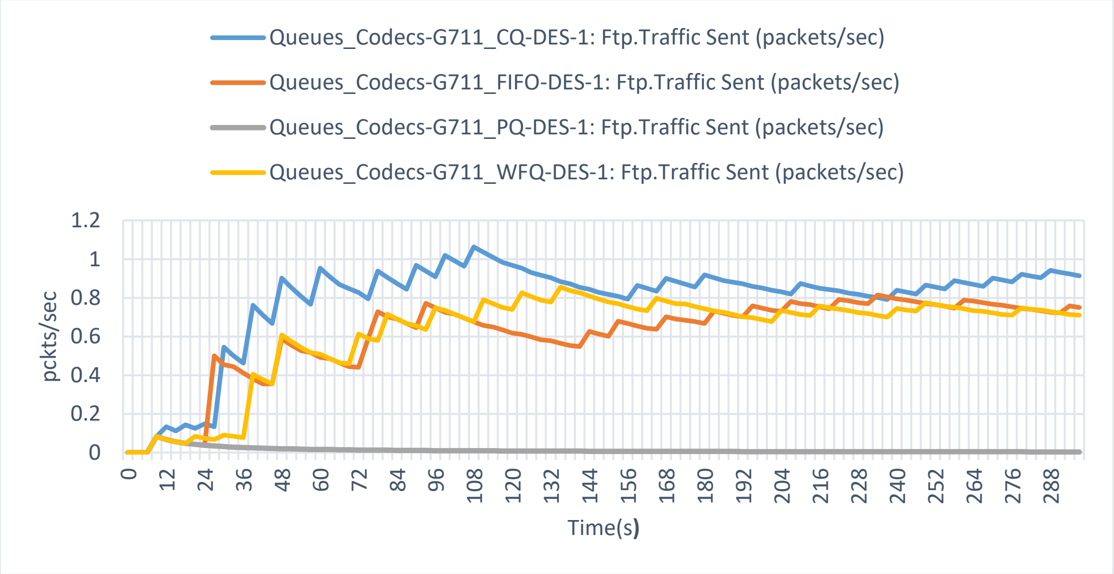


Figure 4.7: FTP traffic sent

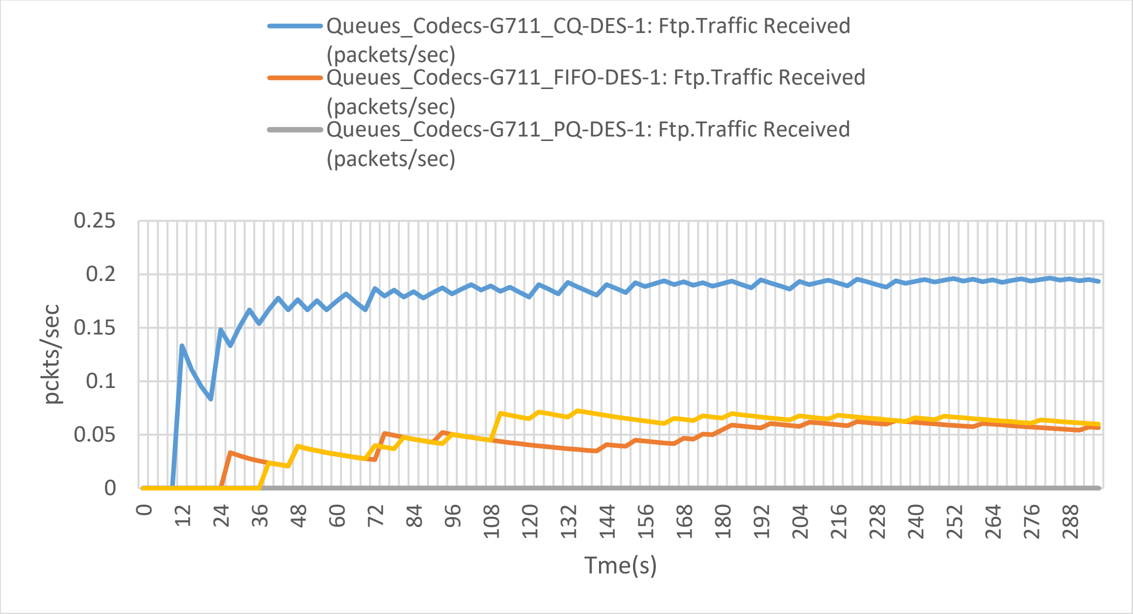


Figure 4.8: FTP traffic Received

Figures 4.7 and 4.8 show that under the G711 codec scheme, the highest amount of FTP traffic was sent and received under the custom queuing discipline while priority queuing recorded the least. The performance of FIFO and WFQ was observed to be close with WFQ performing slightly better. This can also be seen in Table 4.3 where the throughput for WFQ was 8% compared to 7.5% for FIFO.

Table 4.3: Summary of FTP traffic parameters (Average) under G711

Type Of Queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. Traffic Sent	0.913	0.75	0.00333	0.71
Av.Traffic Received	0.193	0.056667	0	0.06
Throughput	21%	7.5%	0%	8%

4.2.2 FTP traffic under G729

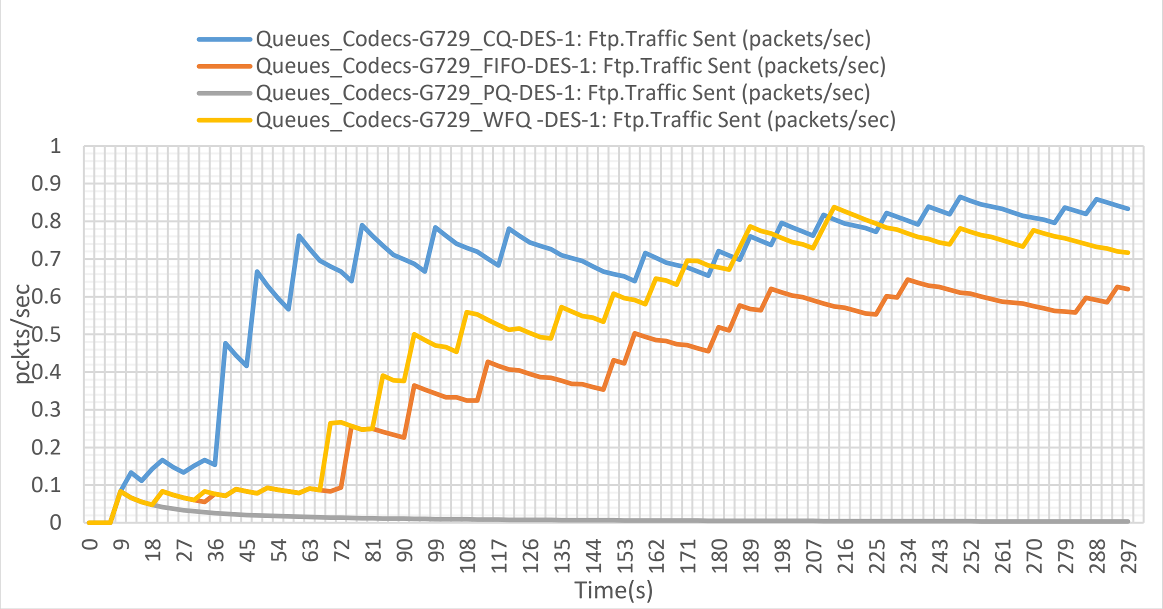


Figure 4.9: FTP traffic sent

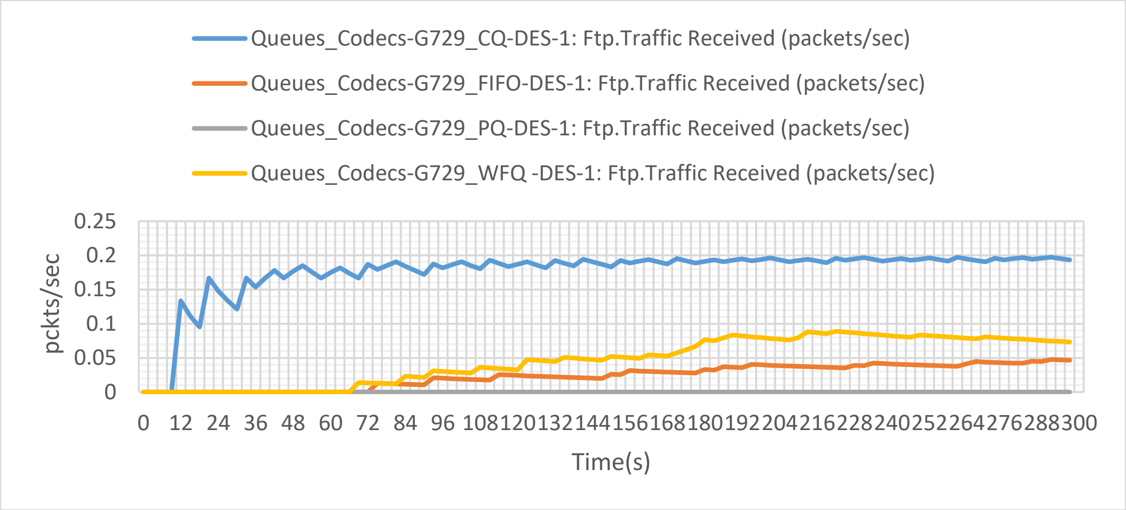


Figure 4.10: FTP traffic received

CQ performed the best in both the amount of FTP traffic that was sent and received across the network. The lowest amount of traffic sent and received was under PQ discipline as shown in Figure 4.9 and Figure 4.10. It was also observed that WFQ and FIFO received no packets for the first sixty (60) seconds despite traffic being sent.

Table 4.4: Summary of FTP traffic parameters (Average) under G729

Type of queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. Traffic sent	0.833	0.62	0.0033	0.717
Av. Traffic received	0.193	0.047	0	0.073
Throughput	23%	7.53%	0%	10%

FTP traffic was seen to perform the best under custom queuing for both codec schemes with a slight improvement in the throughput for G729. Since custom queuing assigns a percentage of available bandwidth to each queue and works in a round-robin fashion, moving from queue to queue, ensures that each is allocated their apportioned bandwidth. This ensured that FTP traffic did not suffer from starvation despite it not being a real-time application. It has already been established that G729 utilizes bandwidth more efficiently as compared to G711, hence the improvement in throughput.

Priority queuing performed the worst under both codec schemes as no packets arrived at the receiver's end despite traffic being sent. This is what was referred to as starvation in [32]. For FIFO and WFQ, it was observed that despite traffic being sent no packets arrived at the receiver's end for some time (about 27 sec in G711 and 60 sec in G729). The reason for this delay could be that both queuing techniques suffered from packet losses and since FTP uses TCP, retransmission was requested causing a delay in the receipt of packets at the receiving node [63]. Transmission under G729 involves compression delay unlike in G711 which extends to the total delay. Hence the longer period of 60 seconds is observed in G729.

4.2.3 Video Conferencing Traffic under G711

Video conferencing packet end-to-end delay was measured with WFQ experiencing the highest delay while CQ performed the best by having the lowest delay as shown in Figure 4.11.

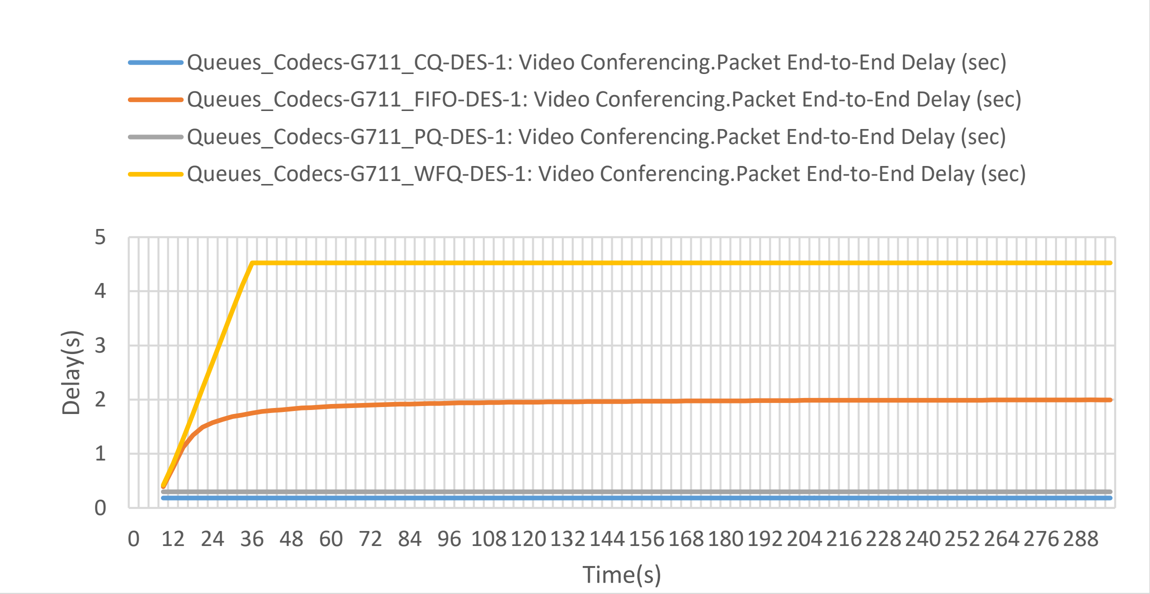


Figure 4.11: Video conferencing end-to-end delay

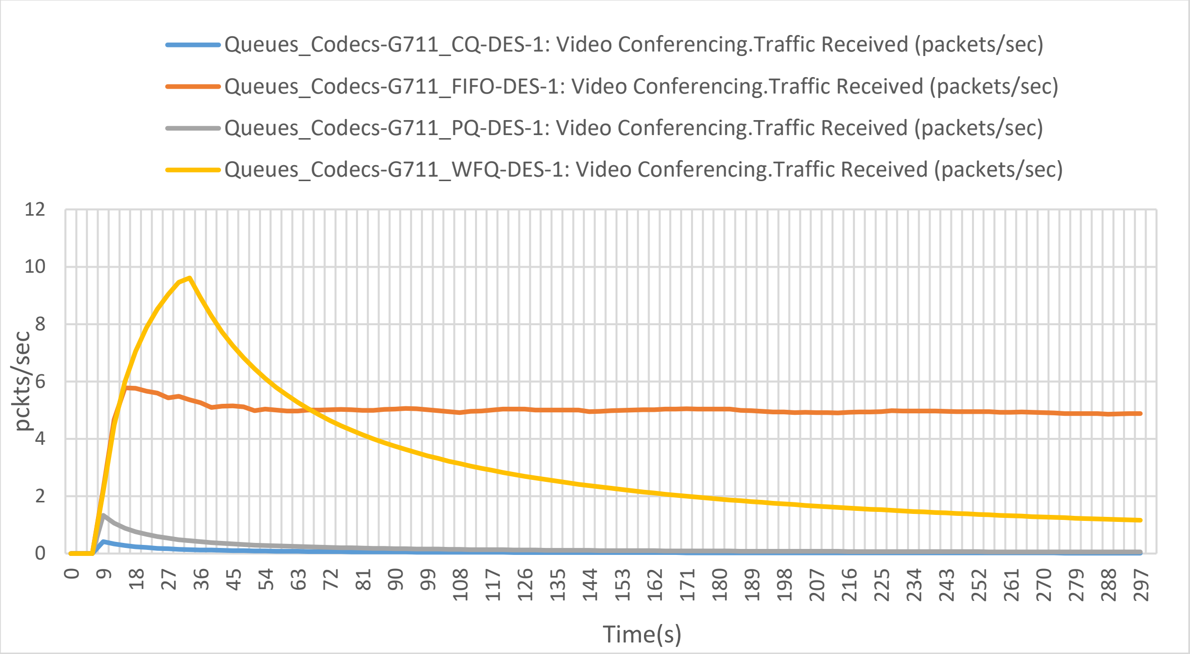


Figure 4.12: Video conferencing traffic received

It can be observed from Figure 4.12 that WFQ received the highest number of packets in the first thirty (30) seconds of the simulation and thereafter the amount dropped to be below that of FIFO. Both PQ and CQ recorded poor results with CQ having the lowest number of packets received.

Table 4.5: Summary of Video conferencing traffic parameters (Average) under G711

Type of queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. Traffic sent	19.337	19.337	19.337	19.337
Av. Traffic received	0.017	4.88	0.0533	1.16
Throughput	0.09%	25.20%	0.28%	6.00%
Delay(s)	0.183	1.892	0.298	4.312

4.2.4 Video conferencing Traffic under G729

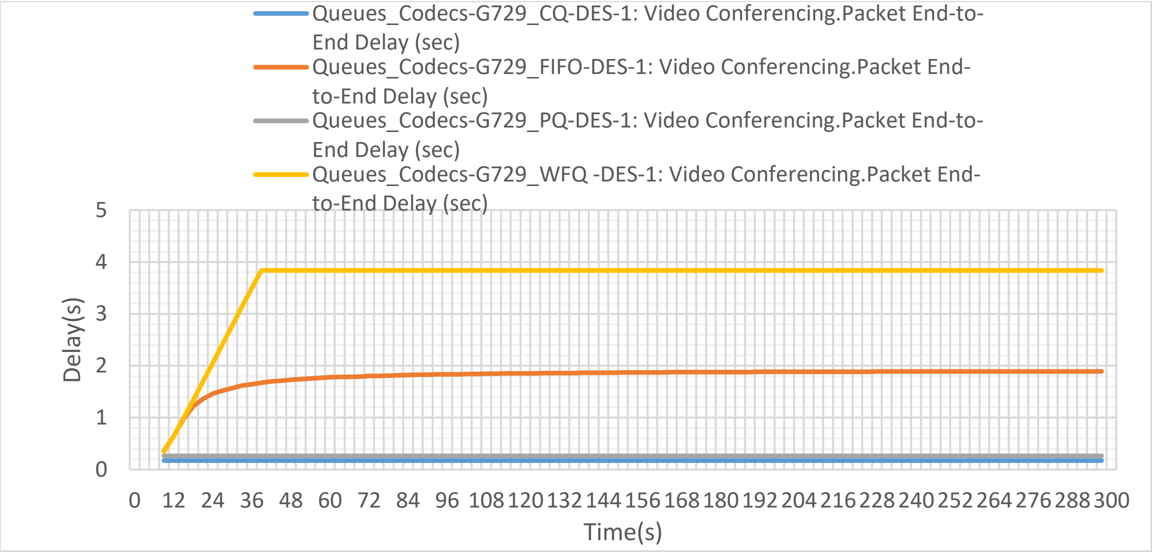


Figure 4.13: Video conferencing End-to-End Delay

Figure 4.13 shows that WFQ suffered the highest delay followed by FIFO queuing. Both PQ and CQ recorded low delays of not more than 0.3 seconds on average. CQ experienced the least delay.

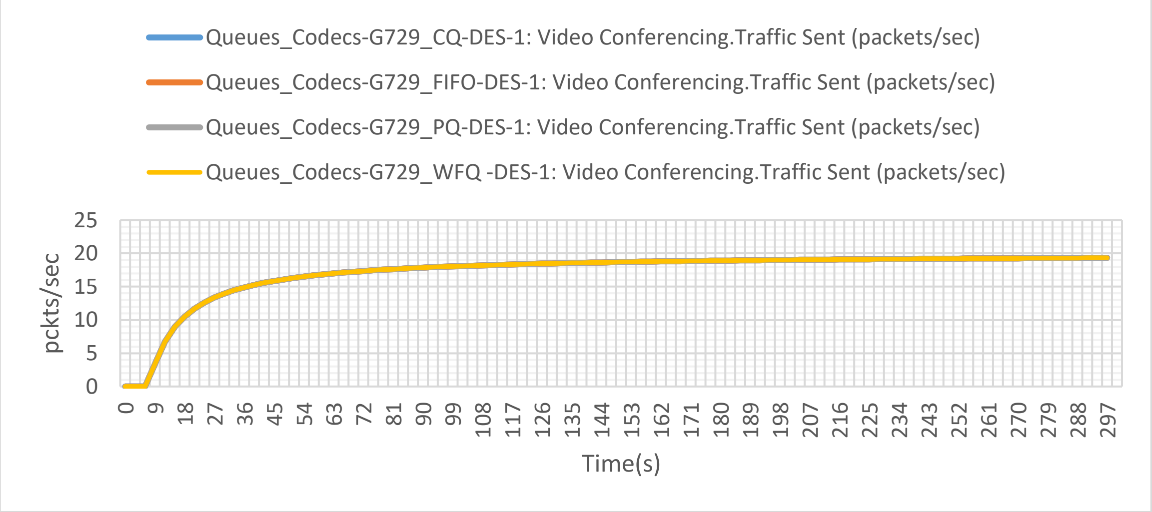


Figure 4.14: Video conferencing traffic sent

The number of video conferencing packets sent per second was the same for all queuing disciplines therefore, the graph in Figure 4.14 appears to have a single line only.

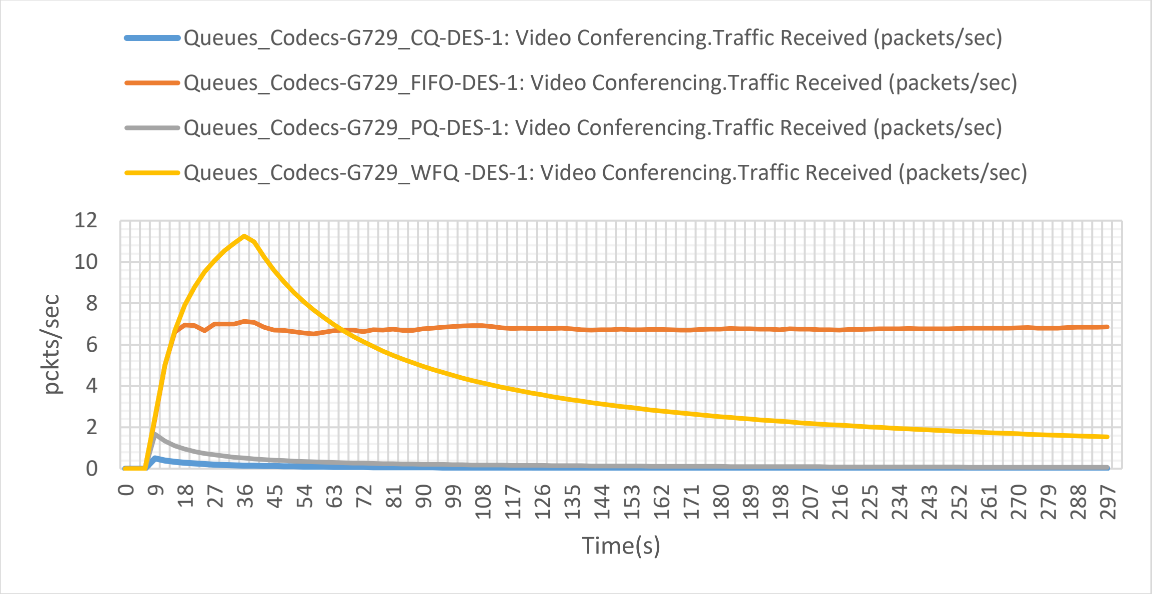


Figure 4.15: Video conferencing traffic received

According to Figure 4.15, WFQ received the highest number of packets in the first thirty (30) seconds of the simulation, and thereafter the amount dropped to be below that of FIFO. Both PQ and CQ recorded poor results with CQ having the poorest performance.

Table 4.6: Summary of Video conferencing traffic parameters (Average) under G729

Type of queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. Traffic sent	19.336	19.337	19.337	19.337
Av. Traffic received	0.02	6.853	0.067	1.537
Throughput	0.10%	35.40%	0.34%	7.95%
Delay(s)	0.174	1.792	0.264	3.637

Custom queuing received the least amount of video conferencing traffic but also suffered the least delay. It was expected that since the queuing technique experienced the least amount of delay, it would receive higher amounts of traffic than the rest of the queuing techniques. The explanation for the poor throughput may be that most packets were dropped because of inadequate bandwidth assigned to the queue. The high end-to-end delay observed in WFQ corresponds with the results recorded in [46] and may have been due to the waiting time that packets experience while other queues are being serviced. The higher the weights assigned the longer the wait (delay). WFQ has been reported to receive the highest video traffic in [14] unlike what has been observed in the results where FIFO has recorded higher than WFQ.

4.3 Optimization Strategies for Queuing Technique Configurations to Enhance FTP and Video Conferencing Performance.

The table below shows the changes made in the queuing profiles in the QoS node in an attempt to optimise the network traffic.

Table 4.7: Changes made in the queuing profiles (I)

Queue Profile	Changes Made
FIFO	Maximum queue size increased from 500 packets to 1000 packets
PQ	FTP traffic configured to the ‘Normal’ priority queue from ‘Low’ priority, Video conferencing traffic set to ‘Medium’ priority while VoIP takes the ‘High’ priority queue.
CQ	FTP traffic byte count increased from 2000 to 10,000 while Video conferencing is given a byte count of 12,000 from 10,000 bytes. VoIP takes a byte count of 14,000.
WFQ	Weight for FTP traffic increased from 1 to 30, Video conferencing allocated a weight of 50 from 40 while VoIP gets a weight of 60.

4.3.1 FTP traffic

Table 4.8: Summary of FTP traffic parameters (Average)

Type of queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. traffic sent	3.831501	2.717991	0.01118	7.524144
Av. traffic received	0.123992	0.021151	0	0.067199
Throughput	3.13%	0.74%	0%	0.93%

After the changes made with the configurations according to Table 4.7, it was observed that the FTP suffered a decrease in throughput for all queuing techniques. This was due to the fact that despite the increase in the amount of traffic sent, FTP also experienced a decrease in traffic received. Despite changing the priority of FTP from Low to Normal under the priority queuing, starvation was still experienced as it still had the lowest priority compared to the others.

4.3.2 Video conferencing Traffic

Table 4.9: Summary of Video conferencing traffic parameters (Average)

Type of queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. Traffic sent	17.17515	17.17515	17.17515	17.17515
Av. Traffic received	0.067081	7.228351	0.223603	2.482961
Throughput	0.4%	42.1%	1.3%	14.4%
Delay	0.170927	3.330772	0.263852	4.349729

Video conferencing traffic experiences an increase in throughput for all queuing techniques with FIFO still having the highest amount.

4.3.3 Voice Traffic

Table 4.10: Summary of Video conferencing traffic parameters (Average)

Type of queuing	**CQ**	**FIFO**	**PQ**	**WFQ**
Av. Traffic sent	342.9887	342.8993	343.1229	343.1229
Av. Traffic received	259.5127	280.4012	343.0894	341.812
Throughput	75.7%	81.8%	99.8%	99.6%
delay	0.14104	3.331793	0.06552	0.065486

Voice traffic is seen to experience a decrease in throughput under CQ and FIFO but experiences almost the same amount for both PQ and WFQ. It generally performs well with all queuing techniques experiencing throughputs ofabove seventy percent.

4.3 Calculations of throughput

An example of how throughput was calculated for VoIP traffic under custom queuing according to Table 4.1.

$$Throughput(\%) = \frac{\#\ of\ Received\ pkts}{\#\ of\ Sent\ pkts} \times 100$$

$$= \frac{386.51}{386.59} \times 100$$

$$= 99.98\%$$

4.4 Summary

Regarding VoIP traffic, the G729 codec scheme produces slightly higher amounts of throughput and lower values of delay compared to the G711 regardless of the queuing technique used. File Transfer Protocol traffic experienced the highest throughput under custom queuing for both codec schemes with a slight improvement under G729. Video conferencing traffic generally experienced poor throughput for all queuing techniques. It can be concluded that Queuing techniques are insufficient for network optimization and require additional network management tools.

CHAPTER FIVE

CONCLUSION AND RECOMMENDATIONS

5.1 Introduction

The previous Chapter discussed the results and findings of the study. This Chapter gives the conclusions and recommendations which were established from the study. This study aimed to show how the choice of the VoIP codec schemes and queuing techniques used in a network affects the performance of VoIP, FTP, and video conferencing traffic by carrying out a performance analysis of the quality of service parameters. Various literature sources have been reviewed focusing on the different queuing techniques and codec schemes available and how they operate. Related literature on how these technologies affect the performance of VoIP traffic in various networks was also covered. This final chapter aims to outline the conclusions from the research carried out as well as make appropriate recommendations based on the results obtained.

5.2 Research questions Review

1. How does each codec scheme (G.711, G.729) perform under varying queuing techniques concerning throughput and delay in VoIP transmissions?

   The G729 codec scheme produces slightly higher amounts of throughput and lower values of delay compared to the G711 regardless of the queuing technique used. It is therefore seen to be a better codec scheme for the quality of VoIP, affecting it positively.
2. How does the implementation of various VoIP codec schemes and queuing techniques impact the throughput, delay, and reliability of FTP transfers and video conferencing applications over the same network?

   FTP traffic experienced the highest throughput under custom queuing for both codec schemes with a slight improvement under G729. It was observed that FIFO and WFQ experienced delay in packets arriving at the receiver's end due to retransmission with G711 performing better.

Video conferencing traffic generally experienced poor throughput (below average) for all queuing techniques. The packet end-to-end delay was observed to be higher than the maximum acceptable value for each of the four queuing techniques with the G729 codec scheme recording lower delay values as compared to G711.

3. What configurations of queuing techniques improve the performance of FTP transfers and video conferencing?

   Increasing the maximum FIFO queue size improves video conferencing traffic throughput at the expense of FTP while assigning VoIP any other priority than the highest in priority queuing results in no throughput of the traffic. Despite the byte count for FTP being significantly increased, its performance is very poor under custom queuing having a throughput of less than five percent. Queuing techniques are insufficient for network optimization and require additional network management tools.

5.3 Summary of Findings

This study used the G711 and G729 codec schemes each requiring a certain bandwidth to represent codecs operating at either high or low bit rates respectively. The queuing techniques used were all available in the simulation tool used and have been covered in the related literature reviewed. From the results obtained, priority queuing, weighted fair queuing, and custom queuing all provided a good performance for VoIP traffic. However, the combination of the G729 and custom queuing resulted in the highest throughput. It was also observed that the use of these technologies also affected the rest of the traffic in the network. For example, while custom queuing provided the best performance for VoIP, it resulted in the lowest amount of video traffic being received. The use of priority queuing resulted in the starvation of FTP traffic regardless of the codec scheme used.

This study contributes to the already existing literature by providing a broader picture of how technologies meant to improve the quality of VoIP also affect the rest of the traffic on the network. This is useful information for network administrators who often need to strike a balance between real-time communication (VoIP), data transfer (FTP), and multimedia streaming (video) while considering available bandwidth and QoS requirements.

5.4 Recommendations

1. Since it was observed that the choice of queuing technique and codec scheme for VoIP can significantly impact the performance of FTP and video either positively or negatively, consideration should be made on how to prevent or minimize the negative impacts.
2. Although FIFO is the simplest and least complex queuing technique, VoIP was observed to have the lowest throughput regardless of the codec scheme used.
3. G729 would best be used for low-bandwidth networks while G711 would be ideal for networks with higher bandwidths and where VoIP communication is the main priority.
4. Queuing techniques can be reconfigured to improve the throughput of various network traffic types. However the improvement is still very minimal and therefore still requires incorporation of more network management tools and technologies.

5.5 Limitations and Further research

Since FIFO is the default queuing technique in a router with the other software queuing techniques only taking over whenever congestion occurs in a network, it would be interesting to see details such as the switch between two different queuing techniques on the same graph. This was however not possible as the researcher was unable to make such configurations in OPNET simulation tool. This because it overwrites the previous queuing technique whenever a QoS configuration is made. Newer and improved queuing techniques could be considered for further studies, including the rest of the technologies considered when setting up an IP network. The network utilized codec shemes tailored for VoIP and was therefore biased towards it. Further steps can be taken to incoparate video codec schemes as well in trying to achieve optimisation.

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APPENDICES

Appendix 1: Other configurations done on queuing Profiles

Table 4.11: Changes made in the queuing profiles (II)

Queue Profile	Changes Made
FIFO	Maximum queue size decreased to 200 packets
PQ	FTP traffic configured to the 'Normal' priority queue from 'Low' priority, VoIP traffic set to 'Medium' priority while Video conferencing takes the 'High' priority queue.
CQ	FTP traffic byte count increased from 2000 to 10,000 while VoIP is given a byte count of 12,000 bytes. Video conferencing takes a byte count of 14,000.
WFQ	Weight for FTP traffic increased from 1 to 30, VoIP allocated a weight of 50 while Video conferencing gets a weight of 60.

Results

Table 4.12: FTP Traffic parameters measured

Type of Queuing	CQ	FIFO	PQ	WFQ
Av. Traffic sent	4.607793346	4.789134301	0.011180147	4.683882699
Av. Traffic Received	0.126272082	0.037788855	0	0.042447993
Throughput	2.7%	0.78%	0%	0.9%

Table 4.13: Video conferencing Traffic parameters measured

Type of Queuing	CQ	FIFO	PQ	WFQ
Av. Traffic sent	17.17515069	17.17515069	17.17515069	17.17515069
Av. Traffic Received	0.078261031	6.548498292	0.089441178	3.044499283
Throughput	0.45%	38.1%	0.52%	17.7%
Delay	0.183502661	0.763410197	0.153544083	4.181887903

Table 4.14: Voice Traffic parameters measured

Type of Queuing	CQ	FIFO	PQ	WFQ
Av. Traffic sent	342.9440065	342.8992859	171.5949848	343.1228888
Av. Traffic Received	239.9432837	284.5630621	0	341.5794829
Throughput	70.0%	83.0%	0%	99.6%
Delay	0.152951896	0.759449385	--	0.065483348

Appendix 2: Published Articles

1. Munthali, E., Tembo, S., Phiri, L. (2024). “Performance Analysis of G.711 and G.729 Codec Schemes under Various Queuing Techniques in Voice over Internet Protocol Transmissions”. *J Electrical Electron Eng*, 3(2), 01-07.
2. Emma Munthali, Simon Tembo, Lukumba Phiri, “Impact of VoIP Codec Schemes and Queuing Techniques on FTP and Video Conferencing Performance”, *International Journal of Networks and Communications,* Vol. 13 No. 1, 2024, pp. 15-22. doi: 10.5923/j.ijnc.20241301.02.

Appendix 3: Ethical Clearance Certificate