University of Alberta

Performance Evaluation for VoIP with Codec, QoS, and OPNET

by

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Abstract

Voice over Internet Protocol (VoIP) is a popular topic for the networking technologies. It is a very attractive and cost effective technology that merges both data and voice networks into providing several benefits including cost savings, flexibility, advance features, and low bandwidth requirements**.**

However, VoIP delivers real-time voice packet across networks using Internet Protocol (IP) instead of traditional Public Switched Telephone Network (PSTN) and it is difficult to guarantee voice quality when VoIP is implemented on the real networks because voice quality is affected by several factors such as delay, jitter, packet loss, and etc. IP traffic also is naturally treated as "best-effort" and transmitted on a first-come, first-served basis. Therefore, voice codec schemes and QoS are carefully chosen to guarantee voice quality before deploying VoIP to the real networks. Codec schemes define voice compression mechanism and have different characteristics. QoS is one of network congestion managements and each queuing has different characteristics.

In this paper, three voice codec schemes such as G.711, G.723.1, and G.729 and three QoS queuing schemes such as FIFO, Priority Queuing (PQ), and Weighted Fair Queuing (WFQ) are used on simulations using OPNET. Voice quality parameters such as voice packet end-to-end delay, voice packet delay variation, and packet loss are collected, analyzed, and compared for performance evaluation.

Keywords: Voice over Internet Protocol (VoIP), Codec, Quality of Service (QoS), First-In First-Out (FIFO), Priority Queuing (PQ), Weighted Fair Queuing (WFQ), Optimized Network Engineering Tool (OPNET)

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Table of Contents

Abstract……………………………………………………………………………ii
Acknowledgement………………………………………………………………..iii
Table of Contents…………………………………………………………………iv
List of Figures……………………………………………………………………vi
List of Tables……………………………………………………………………vii
Acronyms………………………………………………………………………..viii
Introduction……………………………………………..……………………………1
Chapter 1 Background…………………………………….…………………..2
1.1 Voice over Internet Protocol…………………………………..……2
1.1.1 Characteristics and Requirement of VoIP……………………....4
1.2 VoIP Architecture and Operation……………………………..…5
1.2.1 VoIP System…………………………………………………….5
1.2.2 VoIP Protocols…………………………………………………..6
Chapter 2 Coder-decoders (CODECs)………………………………………10
2.1 Overview of CODECs…………………………..…………10
2.2 G.711……………………….………………………………………10
2.3 G.723.1…………………………………………………..………..11
2.4 G.729………………………………………...……………………11
Chapter 3 Quality of Service and Quality Parameter in VoIP…………………..…13
3.1 Overview of QoS technologies……………………………………....13
3.2 QoS Mechanism………………………………...…………………15
3.3 Congestion Management……………………...…………..………16
3.3.1 FIFO Queuing…………………...…………………………..16
3.3.2 Priority Queuing……………………………………………..16
3.3.3 Weighted Fair Queuing……………………………………...17
3.4 Voice Quality Parameters in VoIP…………………………………...19
3.4.1 Delay…………………………………………………………...19
3.4.2 Loss……………………………………………………………19
3.4.3 Delay Variation (Jitter)……………………………………...19

3.4.4 ITU-T's Voice Quality Measurement……………..…………21
Chapter 4 Network Modeling……………………………………………………23
4.1 Simulation Tools…………………………………………………..23
4.2 Simulation Environment…………………………………………..24
4.3 Network Design…………………………………………………...24
4.4 VoIP Traffic Settings……………………………………………...25
Chapter 5 Simulation Results and Data Analysis…………………………………...29
5.1 Comparison of Performance Metrics………………………………...29
5.2 VoIP Traffic…………………………………………………………34
Conclusion……………….....……...……………………………………………………35
References…………………………………………………………………………………..36

List of Figures

Figure 1.1: VoIP Network Topology……………………………………………………….. 2
Figure 1.2: VoIP System………………………………………………………………..6
Figure 1.3: VoIP Protocol Stack…………………………………………………………..6
Figure 1.4: IP Packet for Voice Call: RTP………………………………………………..7
Figure 1.5: H.323 Call Flow………………………………………………………………7
Figure 1.6: H.323 Components………………………………………………………....7
Figure 1.7: SIP Call Flow…………………………………………………………………9
Figure 3.1: IP Precedence values……………………………………………………...14
Figure 3.2: DiffServ IP packet header……………………………………………….....14
Figure 3.3: QoS Mechanism……………………………………………………………..15
Figure 3.4: FIFO Queuing……………………………………………………………….16
Figure 3.5: Priority Queuing……………………………………………………………..17
Figure 3.6: Weighted Fair Queuing……………………………………………………...18
Figure 3.7: MOS Subjective Analysis …………………………………………………..22
Figure 4.1: Simulation Topology………………………………………………………...25
Figure 4.2: Configuration of Voice Encoder Scheme Parameters……………………….26
Figure 4.3: Profile Definition…………………………………………………………..…28
Figure 4.4: QoS Configuration…………………………………………………………..28
Figure 5.1: Figure 5.1: Link utilization between routers – G.729 + WFQ………………30
Figure 5.2: Voice Packet Dropped, Sent, and Received ……………………………….31
Figure 5.3: Voice Packet End-to-End Delay………..…………………………………..32
Figure 5.4: Voice Packet End-to-End Delay with PQ……………………….………..32
Figure 5.5: Voice Packet Jitter – PQ………..…………………………………………..33
Figure 5.6: Voice Packet Jitter – WFQ………………..……………………………….33

List of Tables

Table 2.1: Characteristics of the Codec Schemes……………………………………..12
Table 3.1: Pros and Cons of WFQ……………………………………………………18
Table 3.2: ITU-T Voice Quality Levels………………………………………………21
Table 3.3: Mean Opinion Score (MOS)………………………………………………22
Table 3.4: ITU-T MOS Ratings of Different codec schemes…………………………22
Table 4.1: Codec Parameters for Simulation…………………………………………25
Table 5.1: Summary of simulation results……………………………………………29

Acronyms

VoIP	Voice over Internet Protocol
PSTN	Public Switched Telephone Network
IP	Internet Protocol
PBX	Private Branch Exchange
TDM	Time-Division Multiplexing
RTP	Real Time Protocol
RTCP	Real-Time Transport Control Protocol
UDP	User Datagram Protocol
MCU	Multipoint Control Unit
ISDN	Integrated Services Digital Network
UAC	User Agent Client
UAS	User Agent Server
MMUSIC	Multi-party Multimedia Session control
ITU	International Telecommunication Union
VAD	Voice Activity Detection
SIP	Session Initiation Protocol
IETF	Internet Engineering Task Force
TCP	Transmission Control Protocol
CODEC	Coder-Decoders
PCM	Pulse Code Modulation
MP-MLQ	Multipulse LPC with Maximum Likelihood Quantization
ACELP	Algebraic-Code-Excited Linear Prediction
CS-ACELP	Conjugate-Structure Algebraic-Code-Excited Linear Prediction
MOS	Mean Opinion Score
QoS	Quality of Service
FIFO	First-In First-Out
PQ	Priority Queuing
WFQ	Weighted Fair Queuing

Introduction

In recent years, Voice over Internet Protocol (VoIP) has become a popular topic for networking technologies. VoIP refers to real-time delivery of packet voice across networks using Internet Protocol (IP) instead of traditional Public Switched Telephone Network (PSTN). It is very attractive and cost effective to merge both data and voice networks into one technology and provides benefits including cost savings, flexibility, advance features, and low bandwidth requirements. Therefore, more and more people and companies are adapting a VoIP system.

However, it is difficult to guarantee voice quality when VoIP is implemented on the networks. Voice quality is affected by several factors such as delay, jitter, packet loss, and others. IP traffic is naturally treated as "best-effort" and transmitted on a first-come, first-served basis. These characteristics affect voice quality because these cause large delays, large delay variations, and packet losses in packet delivery. Therefore, voice codec schemes and QoS are considered to guarantee voice quality before deploying VoIP to real networks.

This paper evaluates the performance of VoIP traffic and provides what codec scheme and QoS are outperformed. Simulations using OPNET are carried out. In this paper, I focus on three voice codec schemes such as G.711, G.723.1, and G.729 and two QoS schemes such as Priority Queuing (PQ) and Weighted Fair Queuing (WFQ) because First-In First-Out (FIFO) queuing just provides a means to hold packets while they are waiting to exit an interface. FIFO is also simulated and results are obtained to compare with QoS-enabled results. Voice quality parameters such as voice packet end-to-end delay, voice packet delay variation, and packet loss are collected, analyzed, and compared to evaluate performance.

This paper will be helpful to choose a codec scheme and QoS before deploying VoIP to the real networks.

Chapter 1

Background

1.1 Voice over Internet Protocol

Voice over Internet Protocol (VoIP), known as Internet Telephony, is a technology for delivery of voice calls over Internet Protocol networks such as the Internet instead of the traditional circuit-committed protocols of the Public Switched Telephone Network (PSTN). VoIP is the real-time data and it is transported in the Internet by using the set of Real Time Protocol (RTP)/User Datagram Protocol (UDP)/Internet Protocol (IP) protocols [7]. TCP/IP is a reliable communication suite but it's not suitable for real-time communication like VoIP because it uses acknowledgement/retransmission feature which makes excessive delay [22].

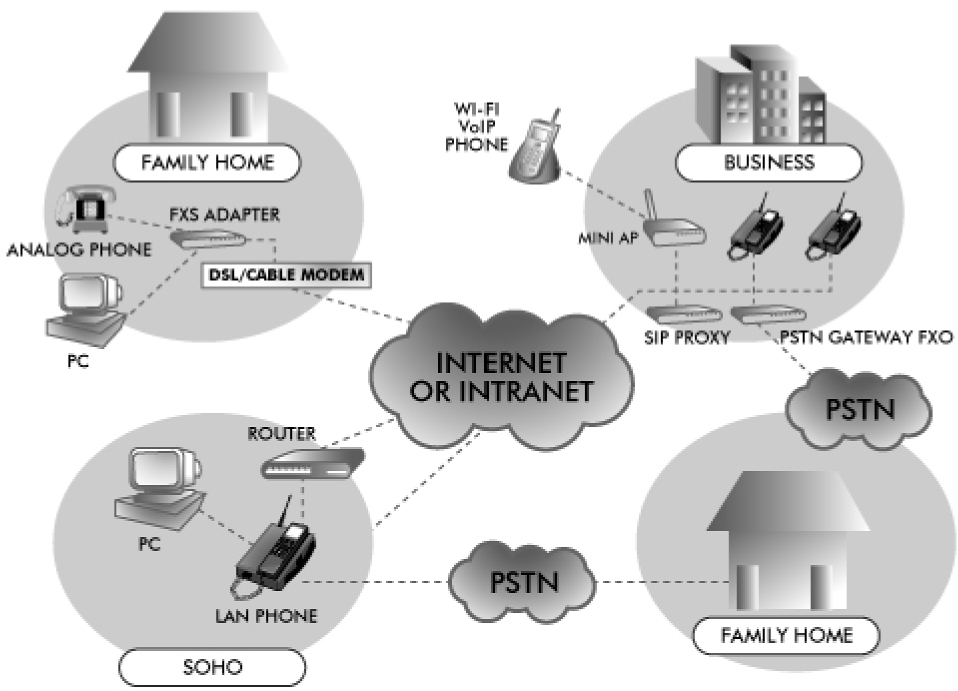


Figure 1.1: VoIP Network Topology [23]

VoIP is providing several benefits:

- Cost savings – Public Switched Telephone Network (PSTN) Private Branch Exchanges (PBXs) cost millions of dollars. VoIP shares bandwidth among multiple logical connections and data traffic, which makes more efficient use of bandwidth whereas traditional time-division multiplexing (TDM) dedicates 64kbps bandwidth per voice channels. It results in substantial savings on capital equipment and operation costs.

- Flexibility – Service providers can easily segment customers, which help the service providers provide different applications, customer services, and rates that depend on the traffic volume needs of the customers and other factors.
- Advanced features – VoIP applications provide advanced features such as advanced call routing, unified messaging, integrated information systems, long-distance toll bypass, encryption, and others.
- Lower bandwidth requirement – PSTN uses line switching technology, so a dedicated bandwidth is required between the two ends and doesn't share the bandwidth during idle time. However, compression technologies are developed and VoIP needs less than 7kbps without a noticeable loss of voice quality.

VoIP also has some disadvantages:

- Bandwidth: VoIP is time-sensitive application and requires properly reserved or allocated bandwidth to ensure VoIP quality. For example, poor quality of VoIP is provided over a dial-up connection even though VoIP consumes low bandwidth because it has 64kbps connection and is difficult to allocate bandwidth for VoIP.
- Security: In VoIP application, incoming and outgoing phone numbers can be intercepted, voice mails can be broke in, and confidential conversation over IP networks can be listened by intruders [27].
- Emergency calls (911): Each traditional telephone is tied to a physical location, thus emergency service providers can easily track callers' location. However, VoIP users are able to use the same phone number anywhere, so it's difficult to track callers when they make emergency calls.
- Power dependency: On traditional telephone system, the power is providing from Central Offices (C.Os) of telephone companies. Thus telephone system works during the power failure. However, VoIP is dependent on the electric power supply, thus VoIP phones don't work without power.

1.1.1 Characteristics and Requirements of VoIP

There are several factors to determine the quality of voice: codec, packet loss, delay, and jitter. The one-way end-to-end delay can accumulate up to 150ms in order that voice quality is acceptable [17]. However, 150 to 400ms can still be considered acceptable, as long as both the speaker and the listener understand that there is a delay, and that both are able to tolerate the delay. At 400ms one-way delay, people start to notice [31].

Delay can be broken down into at least three different components [20]:

- Encoding, compression, and packetization delay at the sender. The delay at the sender is approximately to a fixed delay of 25ms.
- Propagation, transmission, and queuing delay in the network. The network delay should not exceed 80ms in order to establish acceptable quality.
- Buffering, decompression, depacketization, decoding, and playback delay at the receiver. Total delay is approximately to a fixed delay of 45ms.

  The bandwidth required for VoIP is 64kbps. Encoder refers to the conversion of A/D signal into samples. Packetization refers to encapsulation of samples into IP packet. VoIP packets are time sensitive and packet loss can significantly affect VoIP quality because it causes voice breaks and voice skips. Voice delay variation (jitter) can be reduced by using buffer at the receiver.

The required bandwidth for a single VoIP call, one direction, is 64 kbps for G.711, 6.3 / 5.3 kbps for G.723.1, and 8 kbps for G.729. G.711 and G.729 sample 20ms of voice per packet and G.723.1 samples 30ms of voice per packet. Therefore, G.711 and G.729 have 50 pps and G.723.1 (5.3 / 6.3 kbps) has 22.08 / 26.25 pps as the following calculations.

- G.711 PPS = (codec bit rate) / (voice payload size) = 64kbps / (160 * 8bits) = 50 pps
- G.729 PPS = 8 kbps / (20 * 8 bits) = 50 pps
- G.723.1 (5.3 kbps) PPS = 5.3 kbps / (30 * 8bits) = 22.08 pps
- G.723.1 (6.3 kbps) PPS = 6.3 kbps / (30 * 8bits) = 26.25 pps

Total packet size is calculated as Total packet size = (L2 header) + (IP/UDP/RTP header) + voice payload size.

- Total packet size of G.711 = 18 bytes (Ethernet header) + 20 bytes (IP) + 8 bytes (UDP) + 12 bytes (RTP) + 160 bytes = 218 bytes = 218 bytes * 8 bits/byte = 1,744 bits

- Total packet size of G.723.1 (5.3 kbps) = 18 bytes (Ethernet header) + 20 bytes (IP) + 8 bytes (UDP) + 12 bytes (RTP) + 20 bytes = 78 bytes = 78 bytes * 8 bits/byte = 624 bits
- Total packet size of G.723.1 (6.3 kbps) = 18 bytes (Ethernet header) + 20 bytes (IP) + 8 bytes (UDP) + 12 bytes (RTP) + 24 bytes = 82 bytes = 82 bytes * 8 bits/byte = 656 bits
- Total packet size of G.729 = 18 bytes (Ethernet header) + 20 bytes (IP) + 8 bytes (UDP) + 12 bytes (RTP) + 20 bytes = 78 bytes = 78 bytes * 8 bits/byte = 624 bits

Total Bandwidth per call (one direction) is calculated as bandwidth per call = voice packet size * PPS.

- Bandwidth per call of G.711 = 1,744 bits * 50 pps = 87.2 kbps
- Bandwidth per call of G.723.1 (5.3 kbps) = 624 bits * 22.08 pps = 13.78 kbps
- Bandwidth per call of G.723.1 (6.3 kbps) = 656 bits * 26.25 pps = 17.22 kbps
- Bandwidth per call of G.729 = 624 bits * 50 pps = 31.2 kbps

For both directions, the required bandwidth for a single call of G.723.1 (5.3 kbps) is 44.16 pps or 27.56 kbps and G.723.1 consumes the lowest bandwidth.

1.2 VoIP Architecture and Operation

1.2.1 VoIP System

Voice over IP system in Figure 1.2 is composed of several components. In reference [21], the first component is the encoder which periodically samples the original voice signal and assigns a (usually fixed) number of bits to each sample. Further reduction in data rate can be achieved if no signal is encoded during silence periods, a technique known as Voice Activity Detection (VAD). Speech can be modeled as a process that alternates between talkspurts and silences that follow exponential distribution with a mean of 1.2 and 1.8 seconds respectively.

The packetizer follows the encoder, encapsulates a certain number of speech samples or a certain number of frames into packets of equal size, and adds the RTP header. The UDP, IP and Data Link headers are also added. The voice packets are sent over an IP network with delays and drops.

At the receiving end, the playback buffer is an important component to absorb variations in delay and provides a smooth playout. It holds arriving packets until a later playout time in order to ensure that there are enough packets buffered to be played out continuously. Any packets arriving after its scheduled playout time are discarded.
The playout buffer delivers a continuous stream of packets to the depacketizer and eventually to the decoder which reconstructs the speech signal.

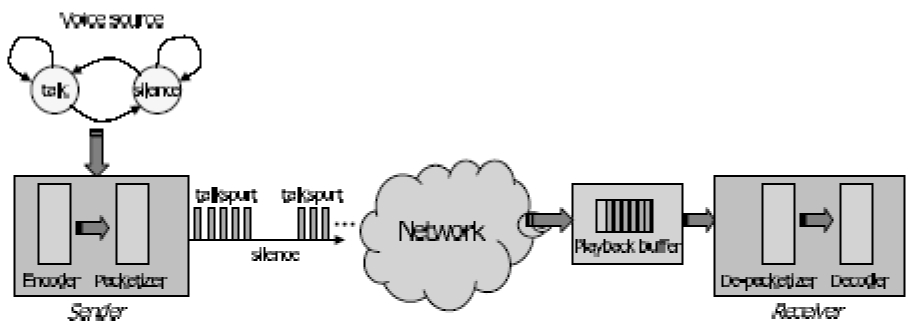


Figure 1.2: VoIP System [21]

1.2.2 VoIP Protocols

Figure 1.3 shows VoIP Protocol Stack. There are two standard VoIP protocols: H.323 and Session Initiation Protocol (SIP).

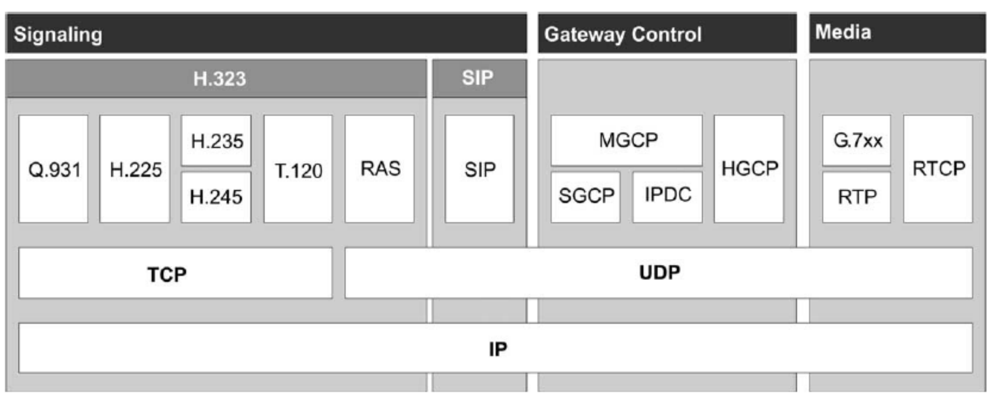


Figure 1.3: VoIP Protocol Stack [1]

H.323, which was adopted in 1996 by the International Telecommunication Union (ITU), was the first call control standard developed for VoIP. The H.323 offers solutions for audio, video, and multipoint data transfers. The H.323 is known for quite complex signalling, high connection setup latencies, and implementation difficulties. H.323 is based on RTP (Real-Time Transport Protocol). RTP is located between UDP and Voice Payload as shown in Figure 1.4.

20 Bytes	8 Bytes	12 Bytes	Variable
IP	UDP	RTP	Voice Payload

Figure 1.4: IP Packet for Voice Call: RTP

RTP consists of two parts: data and control. RTP carries the media stream. RTCP (Real-Time Transport Control Protocol) is the control part of RTP. RTCP monitors transmission statistics and quality of service and aids synchronization of multiple media streams. Figure 1.5 shows H.323 call flow.

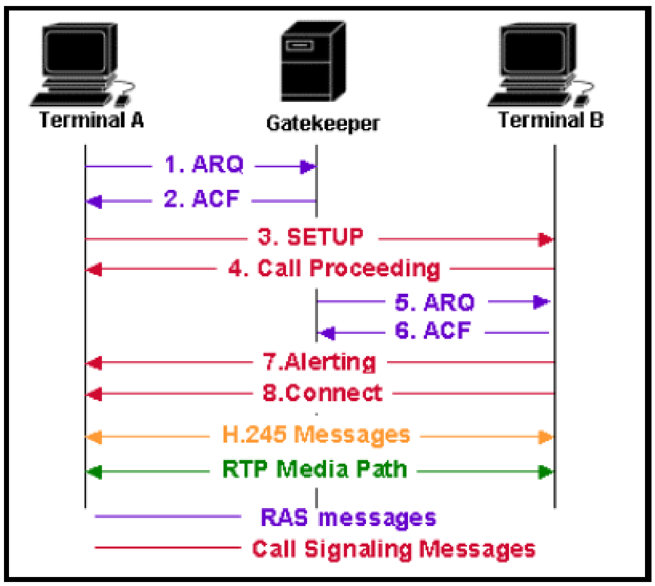


Figure 1.5: H.323 Call Flow [2]

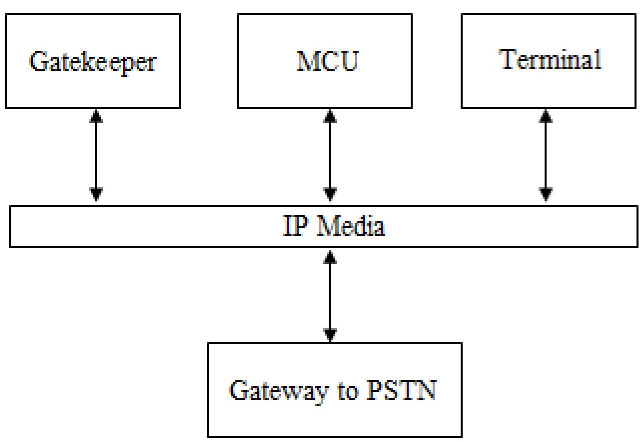


Figure 1.6: H.323 Components

A default H.323 network is composed of four components as shown in Figure 1.6: terminals, gateways, gatekeepers, and multipoint control units (MCUs).
IP telephone and a multimedia PC can be a terminal. A gateway enables communication between H.323 networks and other networks, such as PSTN or ISDN (Integrated Services Digital Network) networks. A MCU is responsible for managing multipoint conferences. A gatekeeper is an optional component providing a number of services to terminals, gateways, and MCUs. Services by gatekeepers are endpoint registration, address resolution, admission control, user authentication and others.

The **Session Initiation Protocol (SIP)** is an ASCII-based peer-to-peer application layer protocol developed by the Internet Engineering Task Force (IETF) Multi-Party Multimedia Session Control (MMUSIC) Working Group and was published as RFC 2543 in April 1999 [2]. It defines initiation, modification, termination of interactive and multimedia communication between users. SIP incorporates elements of two widely used Internet protocols: Hyper Text Transport Protocol (HTTP) used for Web browsing and Simple Mail Transport Protocol (SMTP) used for e-mail. From HTTP, SIP borrowed a client-server design and the use of URLs and URIs. From SMTP, SIP borrowed a text encoding scheme and header style. For example, SIP reuses SMTP headers such as To, From, Date, and Subject [33]. SIP is not dependent on TCP but handles its own acknowledgement and handshaking [2]. SIP provides a suite of security services, which include denial-of-service prevention, authentication, integrity protection, encryption, and privacy services [28].
SIP is a client-server protocol and has two components: SIP user agents and SIP servers. User agents are peers in a SIP network and can be an agent client or an agent server. A user agent client (UAC) initiates a call by sending a SIP request and thereby manages a SIP session. A user agent server (UAS) can accept, terminate, or redirect the request as responses to the SIP request. A SIP server handles requests, e.g. request transfer, security, authentication, and call routing, from user agents. These roles of UAC and UAS only last for the duration of a SIP transaction. SIP call flow is shown in Figure 1.7.
SIP is widely used in VoIP applications and instant messaging such as Microsoft MSN Messenger and Apple iChat.

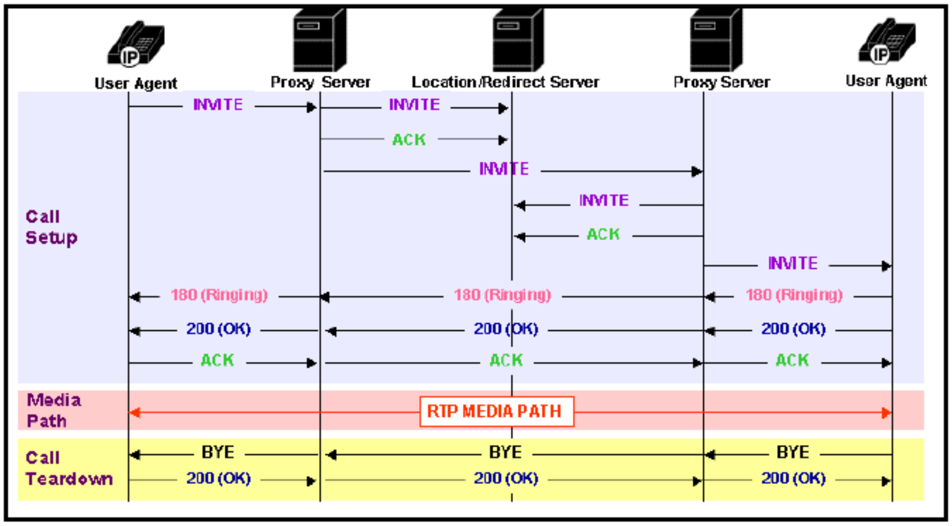


Figure 1.7: SIP Call Flow [2]

SIP provides a number of core primitives that are used as request commands for various functions in the establishment and management of multimedia sessions [34].

- INVITE: The most important primitive defined in the core SIP protocol. It is used to initiate and update a multimedia session with another client using the Session Description Protocol [34].
- ACK: The ACK primitive is used in conjunction with the INVITE primitive. It forms the final part of the three-way handshake that is involved in establishing a multimedia session.
- CANCEL: The CANCEL request primitive provides the originator of the call the option to cleanly terminate the INVITE transaction before the call answered.
- BYE: The BYE primitive request allows either of the clients to terminate INVITE-initiated session.
- REGISTER: The REGISTER primitive is used by the client to convey its current location information.
- OPTIONS: The OPTIONS primitive allows for the probing of both servers and endpoints to obtain important information such as what SIP extension are supported and possible media types that are supported.

Chapter 2

Coder-decoders (CODECs)

2.1 Overview of CODECs

Coder-decoders (CODECs) provide the coding and decoding translation between analog and digital facilities. Each CODEC type defines the method of voice coding and the compression mechanism to save network bandwidth. The actual binary values used to represent the voice vary based on which codec is used. The most significant feature of each codec is the amount of bandwidth required to send the voice payload created by codec. Currently, there are many different audio codec schemes available for voice applications. The simplest and most widely used codec schemes are G.711, G.723.1, and G.729. Characteristics of the codec schemes are shown on Table 2.1.

2.2 G.711

G.711 is an ITU-T standard and its formal name is Pulse Code Modulation (PCM) of voice frequencies. It was first released for usage in 1972 and was profusely used in telephony. The nominal value recommended for the sampling rate is 8000 samples per seconds. The tolerance on that rate should be ± 50 parts per million (ppm). Non-uniform (logarithmic) with 8 bits is used to represent each sample, resulting in a 64 kbps constant bit rate. Two encoding laws are recommended that these are commonly referred to as A-law and μ-law.

It has the following features:

- Sampling frequency 8 kHz
- Bit rate 64 kbps
- Typical algorithm delay is 0.125 ms with no look-ahead delay
- Waveform speech coder

2.3 G.723.1

Its official name is Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbps. G.723.1 compresses voice in 30 ms frame and is mostly used in Voice over IP application due to its low bandwidth requirement.
There are two bit rates at which G.723.1 can operate:
6.3 kbps (using 24 byte frames) using a MP-MLQ algorithm
5.3 kbps (using 20 byte frames) using an ACELP algorithm
This codec has the following features:

- Sampling frequency 8 kHz / 16-bit (240 samples for 30 ms frame)
- Fixed bit rate: 5.3 kbps with 20 byte 30ms frame and 6.3 kbps with 24 byte 30 ms frame
- Algorithm delay is 37.5 ms per frame with 7.5 ms look-ahead delay
- Hybrid speech coder with high bit rate using MP-MLQ and low bit rate using ACELP

2.4 G.729

It's officially described as Coding of speech at 8kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP) and became a standard in 1996. G.729 compresses digital voice in packets in 10 ms duration and is also mostly used in Voice over IP application such as Skype due to its low bandwidth requirement. The bit rate operates at 8kbps. The output frame size of the encoder is 10 bytes and the frame and the frame duration is 10 milliseconds. The output bandwidth of the encoder is 8 kbps. There are extensions (6.4kbps and 11.8kbps) for worse and better speech quality.
Features for G.729 are:

- Sampling frequency 8 kHz / 16-bit (80 samples for 10 ms frames)
- Fixed bit rate 8 kbps / 10 ms frames
- Fixed frame size 10 bytes for 10 ms frames
- Algorithm delay is 15 ms per frame with 5 ms look-ahead delay
- Hybrid speech coder using ACELP

Codecs	Bit rate (bps)	Sampling rate (kHz)	Frame size / Lookahead (ms)	Payload (byte)	MOS
G.711	64	8	0.125 / 0	160	4.1
G.723.1	6.3 / 5.3	8	30 / 7.5	20 / 24	3.9 / 3.62
G.729	8	8	10 / 5	20	3.92

Table 2.1: Characteristics of the Codec Schemes

The best candidate for use in the Internet telephony service is G.729 because user perception of G.729 service is not operationally different from that of the PSTN service. Since the PSTN uses G.711 codec without VAD/VAC for digital transport, this means that G.729 is a viable substitute for G.711 in any context in which the handling of acoustic data signals is not an issue. G.729 will also be a more cost-effective alternative because G.711 service would have to use VAD/VAC to achieve any bandwidth reductions [35].

The G.723.1 is a viable candidate for use in a voice service only in an environment in which it is critical to reduce bandwidth requirement and there is inherent resiliency to delays to reduce bandwidth oral stimuli and aural responses. It reduces signal data rates to 5.3 kbps or 6.3 kbps from 8 kbps required for G.729, but at a cost of using 30-ms samples instead of 10-ms samples. With the smallest jitter buffer, this difference will add more than 110 ms to the round-trip delay, while substantially reducing resiliency to jitter and dropped packet rates over G.729 carrying 10-ms voice samples in each packet. The G.723.1 can achieve a reduction in capacity requirements over G.729 of about 60 percent when full headers are used. When header compression is used, a reduction in capacity requirements over G.729 can be achieved about 34 percent [35].

The only code that is viable for the hybrid transport service is G.711 without VAD/VAC because the CELP coding used in G.723.1 and G.729 preserves phase and produces too much amplitude jitter. When there are no provisions for differential handling of fax and data modem signals, application of voice activity detection (VAD) will result in premature disconnects of acoustic modem transmission [35].

Chapter 3

Quality of Service and Quality Parameters in VoIP

3.1 Overview of QoS technologies

There are three types of services on IP networks: Best-Effort, Integrated Services (IntServ), and Differentiated Services (DiffServ).
IP traffic naturally is treated as "best-effort" and transmitted on a first-come, first-served basis. The network delivers packets in the shortest path without guaranteeing delivery and quality. Quality of Service is the ability of network to provide improving service to selected network traffic over various underlying technologies including Frame Relay, ATM, IP-routed network, and others [26]. There are many methods to implement QoS on the network, but only three queuing algorithm among congestion management features are considered on this report. IEFT initially defined IntServ [RFC1633] and then DiffServ [RFC2475] as two architecture models for QoS [29].

Integrated Services (IntServ): IntServ defines a fine-grained (flow-based) QoS system and can be used to deliver video and audio to the receiver without interruption. The Resource Reservation Protocol (RSVP) [RFC2205] is used to signal resource reservation across the network.

Differentiated Services (DiffServ): DiffServ defines a simple, scalable, and coarse-grained (class-based) mechanism for classifying, managing network traffic, and providing QoS. The 3-bit IP precedence in the Type of Service (ToS) byte of the IP header as shown in Figure 3.1 is re-defined into the DHCP field. DiffServ uses the 6-bit Differentiated Services Code Point (DSCP) field in the header of IP packets as shown in Figure 3.2 for packet classification purposes. DiffServ defines four Per-Hop Behaviours (PHBs): Default PHB, Expedited Forwarding (EF) PHB, Assured Forwarding (AF) PHB, and Class Selector PHB.

- Default PHB: for typical best-effort traffic
- Expedited Forwarding (EF) PHB: for low-latency, low-loss, and low-jitter traffic

- Assured Forwarding (AF) PHB: for assurance of delivery
- Class Selector PHB: to maintain backward compatibility with the IP precedence

The Expedited Forwarding (EF) PHB [RFC3246] has the low-latency, low loss, and low jitter. Thus it is suitable for real-time applications like VoIP.

IP Precedence Value	Binary Value	Priority
0	000	Routine
1	001	Priority
2	010	Immediate
3	011	Flash
4	100	Flash Override
5	101	Critical
6	110	Internetwork Control
7	111	Network Control

Figure 3.1: IP Precedence values [29]

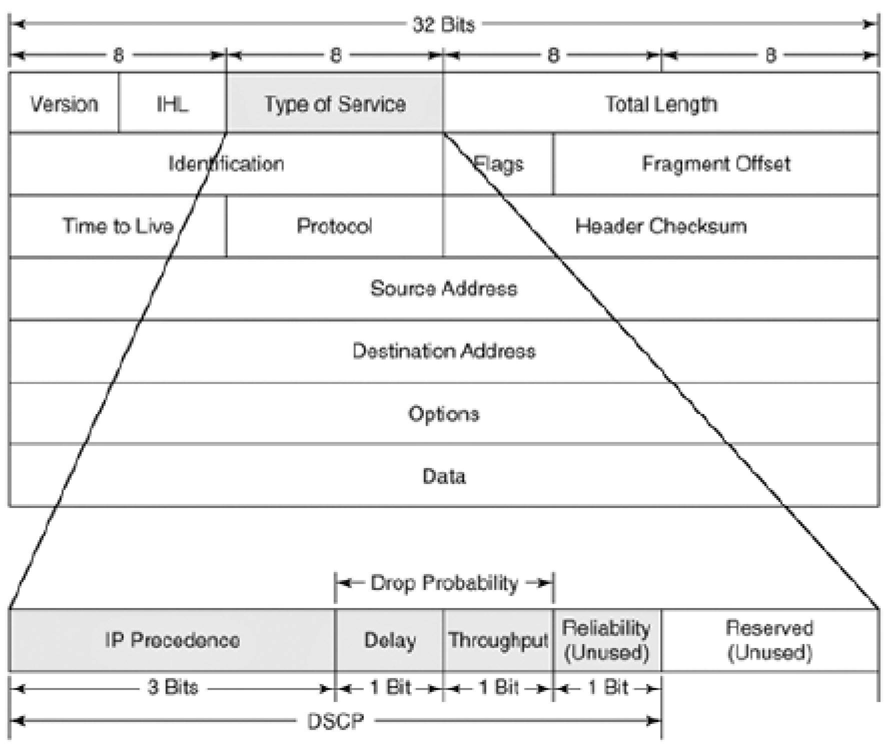


Figure 3.2: DiffServ IP packet header [29]

3.2 QoS Mechanism

As shown in Figure 3.3, QoS mechanism at the router level includes

- Traffic classification
- Traffic marking
- **Congestion management** such as Priority Queuing and Weighted Fair Queuing
- Congestion avoidance such as Random Early Detection (RED) and Weighted Random Early Detection (WRED)
- Traffic policing
- Traffic shaping such as token bucket, leaky bucket, and TCP rate control

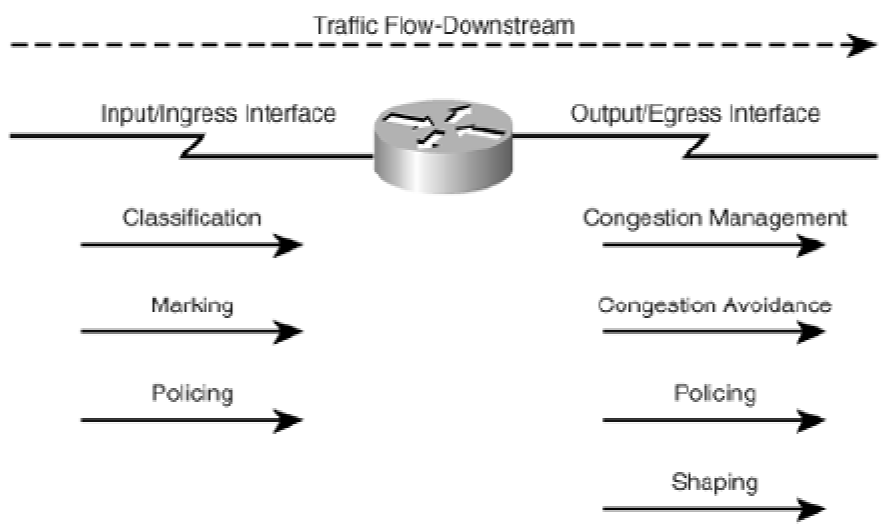


Figure 3.3: QoS Mechanism [29]

In this paper, only congestion management is considered. Other QoS mechanisms are out of the scope of the project and remove the future works.

3.3 Congestion Management

3.3.1 FIFO Queuing

As show in Figure 3.4, First-In First-Out (FIFO) queuing just provides a means to hold packets while they are waiting to exit an interface. FIFO Queuing has only one queue and treats all packets equally because it has no concept of priority or classes of traffic. Consequently it doesn't need for classification to decide the queue into which packet should be placed. It also doesn't need for scheduling logic to pick which queue from which to take the next packet. The really interesting parts of FIFO Queuing are the queue length, which is configurable, and how the queue length affects delay and loss. Ill-behaved sources can consume all the bandwidth, bursty sources can cause delay in time-sensitive or important traffic, and important traffic can be dropped because less important traffic fills the queue [25]. FIFO queuing is good for large queues and fast-switching environments with predictable outcomes [30].

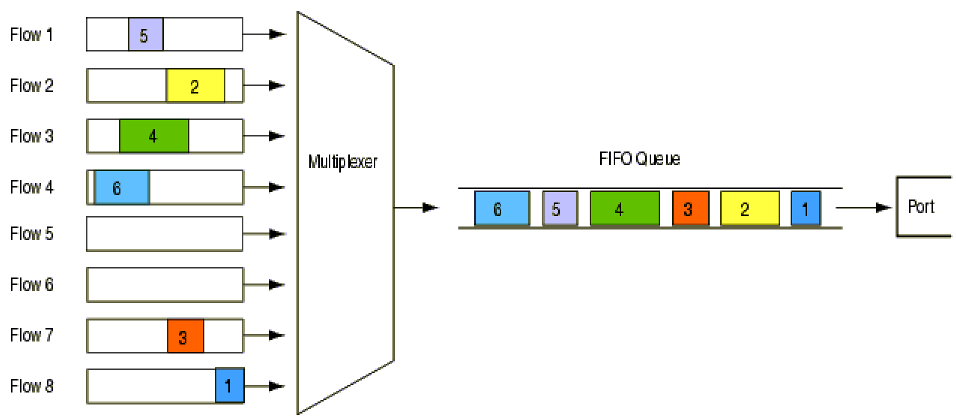


Figure 3.4: FIFO Queuing

3.3.2 Priority Queuing

Priority Queuing is based on the classification of the packet into a service class and the packet is placed into the appropriate queue based on the classification as shown in Figure 3.5. It can configure four queues, named High, Medium, Normal, and Low. Priority Queuing (PQ) schedules traffic such that higher priority queues always get serviced with the side effect of starving bandwidth for the lower-priority queues. This mechanism is good for important traffic such as real-time applications, but can lead to queue starvation. Packets in the high queue can claim 100 percent of the link bandwidth with minimal delay and minimal jitter, but the lower queues suffer bandwidth. Thus administrators need to be careful while using this queuing because a small error in judgement can cause very poor deployment of this scheme. It is a common practice to keep the higher queues shorter and the low priority queue longer [30].

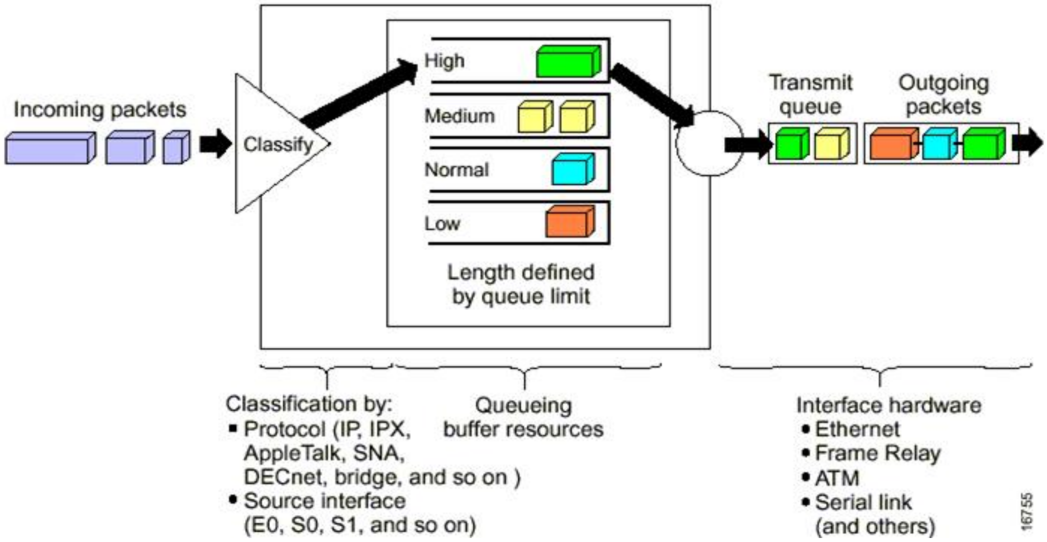


Figure 3.5: Priority Queuing

3.3.3 Weighted Fair Queuing

Weighted Fair Queuing (WFQ) classifies packets based on flows and offers dynamic allocation of resources to all queues based on the weights configured as shown in Figure 3.6. A flow consists of all packets that have the same source and destination IP address, and the same source and destination port numbers. Each flow uses a different queue and an interface can have up to 4,096 queues. It schedules interactive traffic to the front of the queue to reduce response time and fairly shares the remaining bandwidth between high bandwidth flows [24]. This prevents any single uncontrolled application from bursting traffic into the network [30]. For voice, WFQ can calculate the bandwidth required by identifying the voice flows using ToS or precedence bits. The calculation made by WFQ does not reserve or guarantee bandwidth, but provides to figure out how much bandwidth each flow needs. How much bandwidth each flow gets is a function of the total number of flows and their associated ToS values [31].

BW of a flow = Circuit BW * (1+ IP Precedence)/Sum of All Flows

For example, a circuit bandwidth is 56kbps. Two VoIP flows at 24 kbps and two FTP flows at 56 kbps each are serviced. If IP precedence is not configured, the bandwidth available for one VoIP flow can have 14 kbps. BW = 56 kbps * (1/4) = 14 kbps

This result is not acceptable because one VoIP flow needs 24 kbps to perform properly. If the precedence value is set higher for voice at a value of 5, the resulting bandwidth available for the voice flow is 24 kbps. BW = 56 kbps * (1+5)/14 = 24 kbps

Now WFQ realizes that voice flow needs 24 kbps but its full bandwidth still may not be met, depending on the demands of other flows and their respective precedence.

Pros and Cons of WFQ is in Table 3.1.

Pros	Cons
Simple configuration	Multiple flows can end up in one queue
Guarantees throughputs to all flows	Does not support the classifications
Drop packets of most aggressive flows	Can't provide fixed bandwidth guarantees

Table 3.1 Pros and Cons of WFQ

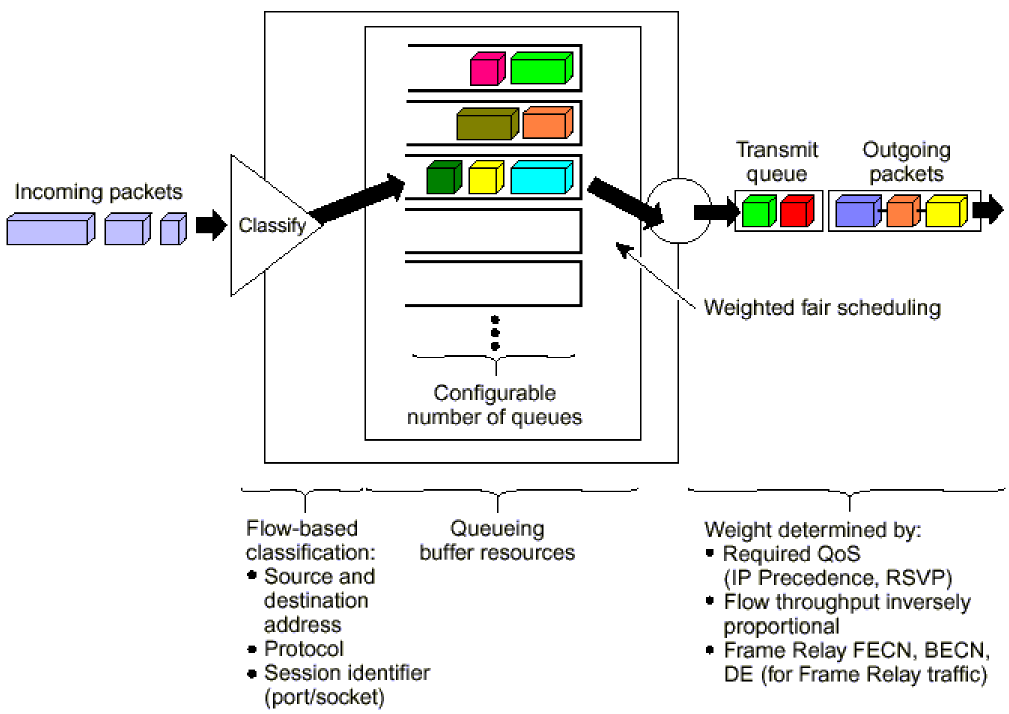


Figure 3.6: Weighted Fair Queuing

3.4 Voice quality parameters in VoIP

Voice over IP (VoIP) refers to real-time delivery of voice packet across IP networks. Voice quality is affected by delay, jitter, and unreliable packet delivery that are typical characteristics of the basic IP-network service.

3.4.1 Delay

All packets in a network experience some delay between source and destination.
The International Telecommunication Union – Telecommunication Standardization Sector (ITU-T) Recommendation G.114 advises that one-way delay can accumulate up to 150 ms without effect, but beyond that point negative consequences begin to gradually accrue [17]. **In [17] a delay of up to 200ms was considered to be acceptable.** Delay can break down into at least three different components [20]: (1) encoding, compression, and packetization delay at the sender (2) propagation, transmission and queuing delay in the network and (3) buffering, decompression, depacketization, decoding, and playback delay at the receiver. Thus, a codec algorithm and queuing algorithm need to be carefully chosen to minimize the voice traffic delay.

3.4.2 Loss

Packets can be lost, dropped, and discarded for many reasons in a network. Packet loss on VoIP network introduces audio distortions that cause voice quality to decrease as the rate of packet loss increases. VoIP packets are also time sensitive, so packet loss can significantly affect VoIP quality. The tolerable loss rates are within 1-3% and the quality becomes intolerable when more than 3% of the voice packets are lost [19].

3.4.3 Delay Variation (Jitter)

Jitter is the variation in timing of some event against a clock. However, with IP packets there is no clock to directly compare the packet arrival times to, so we need to consider differences in delay, as worked out from packet time stamps. RFC 3393 defines this packet jitter as the Instantaneous Packet Delay Variation (IPDV) and deprecate the use of the term jitter. The IPDV is defined as the difference in one way delay between successive packets, ignoring any lost packets, and with the one way delay being received at the destination. If a part of the packet switching process always takes the same time, then obviously its effect will be cancelled out when taking the difference in delay. In the discrete event simulation tool OPNET, the definition of jitter is the time differences between the instances when successive packets are received at the destination minus the time difference between instances when these packets are sent at the source.

Packet Delay Variation (PDV) is defined by the IETF as the difference in one way delay between selected packets ignoring any lost packets. However, the IETF doesn't define what the selection criteria is for PDV in general, and it could be randomly selected packets in a sliding window or the packets which give the maximum and minimum delay in a sequence. In OPNET, the PDV is defined as the variance of the delay and is always $\geq$ 0. Jitter has signed (+ or -) values [37].

Delay variation defines as the different value between the delays of two queuing packets. Delay variation also is called Jitter because of describing the variation of IP packet arrival times. Jitter is a natural result of buffering in packet switched networks. Generally the jitter increases when the traffic becomes more bursty [16]. The acceptable value of jitter is between 0 ms and 50 ms [6].

3.4.4 ITU-T's Voice Quality Measurement

The ITU-T defines five categories of voice quality levels as shown in Table 3.2.

Grade	Quality Level	Description
5	Toll quality	Toll quality emulates and sounds like a copper wire.
4	Transparent quality	This is very similar to toll quality with some tolerable distortion and almost imperceptible distractions
3	Conversational quality	Conversional quality has perceptible distortions and annoying distractions.
2	Synthenic quality	Synthetic quality has tolerable but has very annoying distractions and poor reproduction of the speaker's voice fidelity.
1	Unsatisfactory	A simple exchange between the speaker and listener is strained to the point of the speaker repeating what's been said again and again.

Table 3.2: ITU-T Voice Quality Levels [31]

- **Mean Opinion Score (MOS)** [31]

The ITU-T recommends measurement of voice quality using Table 3.2 by subjective methodology called the Mean Opinion Score (MOS). Test subjects are gathered into a lab environment and asked to rate voice quality through varying methods of compression. The participants listen to a recorded message that is chosen based on its varying fricatives. As they listen to the message, they realize that there are hard sounds and soft sounds, long vowels and short vowels. As the listeners rate the different compression methods, the average, or mean, of all the participants is calculated after the tests. Mean Opinion Score is in Table 3.3.

MOS	Listening Quality	Listening Effort	Loudness Preferrence
5	Excellent	No effort required	Much louder than preferred
4	Good	No appreciable effort required	Louder than preferred
3	Fair	Moderate effort required	Perferred
2	Poor	Considerable effort required	Quieter than preferred
1	Bad	No meaning understood	Much quieter than preferred

Table 3.3: Mean Opinion Score (MOS) [32]

Table 3.4 lists the MOSs of selected compression schemes as published by the ITU-T. G.711 had very high score because it has a short framing size or sampling interval, and require very little of the processor. G.729 demands more processing power and has the low data rate.

Codec	Compression Method	Date rate [bps]	Mean Opinion Score
G.711	PCM	64	4.1
G.723.1	MPC-MLQ	6.3	3.9
	ACELP	5.3	3.62
G.729	CS-ACELP	8	3.92

Table 3.4 ITU-T MOS Ratings of Different codec schemes [31]

Figure 3.7 shows the MOS results on a comparative graph. Vocoders always tend to be low score no matter what their bit rate. Wave form coders have better score at high bit rates but has significant quality drop at lower bit rate. The hybrid coders have high score even at lower bit rates.

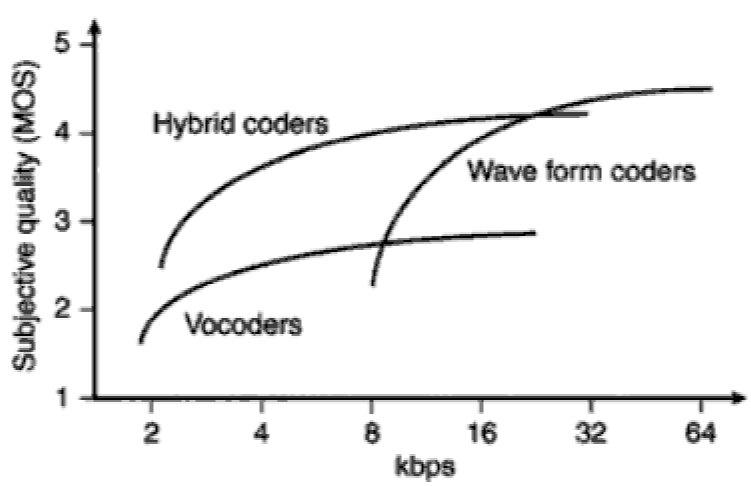


Figure 3.7: MOS Subjective Analysis [31]

Chapter 4

Network modelling

4.1 Simulation Tools

Simulation is the process of testing a designed model on platform which imitates the real environment. It helps the user to predict its strength and weakness before implementing the model in real world. The popular simulation tools used for the data networks are:

- OMNeT++: OMNeT++ is an extensible, modular, component-based C++ simulation library, primarily for building network simulators. It is a discrete event simulation environment and is free for academic and non-profit use [11]. The INET Framework is an open-source communication networks simulation package for the OMNeT++ simulation environment and supports voice protocols [12]. However, it is not straightforward to use OMNeT++ because it requires in learning of number of tutorials, demos, and large web-based documentation.

- NS-2: NS-2 is discrete event simulator targeted at networking research. The knowledge of Otcl and C++ languages is required in order to build the simulation topologies on NS-2 [13]. The documentation is not available for all modules and the source code is required to read and results are not generated automatically.

- OPNET Modeler: OPNET is an object-oriented simulation tool for planning, modelling, and analyzing performance of simulation of network communication, network devices and protocols. OPNET supports GUI (Graphic User Interface), comprehensive library of network protocols and models, graphical interface for results viewing, availability of documentation for users to develop the network models. The users don't need to have deep programming knowledge to use OPNET [7].

4.2 Simulation Environments

The main task for simulations is to evaluate the performance of VoIP traffic by using performance metrics such as voice packet end-to-end delay, voice packet delay variation, packet loss, voice packet sent, and voice packet received. The simulation results obtained are analyzed to determine what codec and QoS combination has the best performance.

Assumptions:

- Two routers are connected by a DS1 link.
- Ethernet workstations and an Ethernet server are connected with routers by 10BaseT links.
- FTP and Video applications are used for generating background traffic in the simulations.
- Each simulation experiment considers 5 minutes of simulation time.
- Only peer-to-peer voice calls are considered without conference calls.

4.3 Network Design

The nine simulation scenarios are performed in the OPNET Modeler 14.5 – Education version. The simulation topology is shown on Figure 4.1.

- Scenario 1: G.711 + FIFO
- Scenario 2: G.711 + Priority Queuing
- Scenario 3: G.711 + Weighted Fair Queuing
- Scenario 4: G.723.1 + FIFO
- Scenario 5: G.723.1 + Priority Queuing
- Scenario 6: G.723.1 + Weighted Fair Queuing
- Scenario 7: G.729 + FIFO
- Scenario 8: G.729 + Priority Queuing
- Scenario 9: G.729 + Weighted Fair Queuing

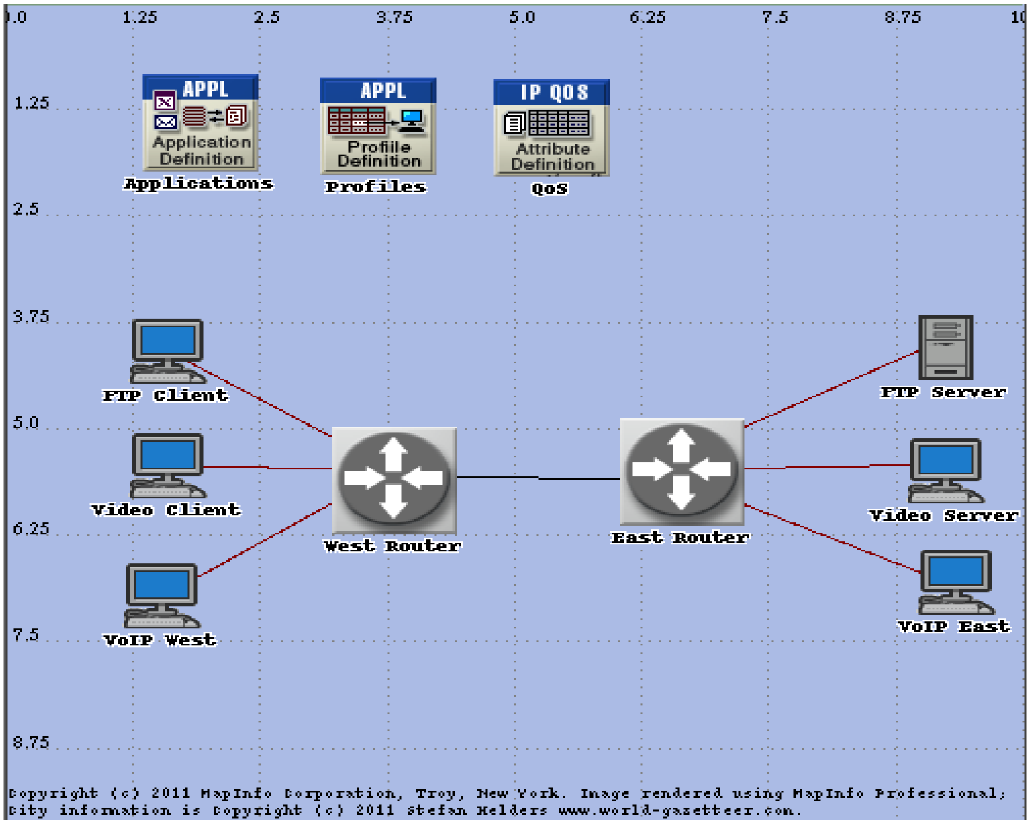


Figure 4.1: Simulation Topology

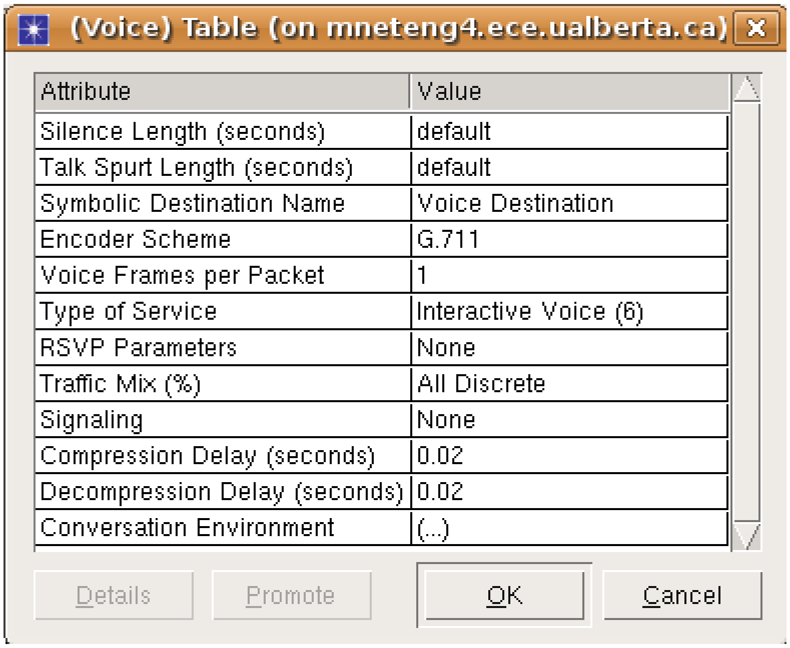
4.4 VoIP Traffic Settings

The simulations of VoIP network are deployed in the OPNET Modeler 14.5. The simulations consist of nine scenarios with considering the same network topology. Three applications (FTP, Video, and VoIP) are modeled in the simulation by using the Applications attributes. A voice application is used to model the VoIP traffic in OPNET.

	G.711	G.723.1	G.729
Silence Length (sec)	exp (0.65)	exp (0.65)	exp (0.65)
Talk Spurt Length (sec)	exp (0.325)	exp (0.325)	exp (0.325)
Voice Frame per Packet	1	1	1
Type of Service	Interactive Voice	Interactive Voice	Interactive Voice
Frame size (msec)	4	30	10
Lookahead Size (msec)	0	7	5
Coding Rate (kbps)	64	5.3	8
Speech Activity Detection	enable	enable	enable

Table 4.1: Codec Parameters for Simulation

Codec parameters for simulations are shown on Table 4.1.

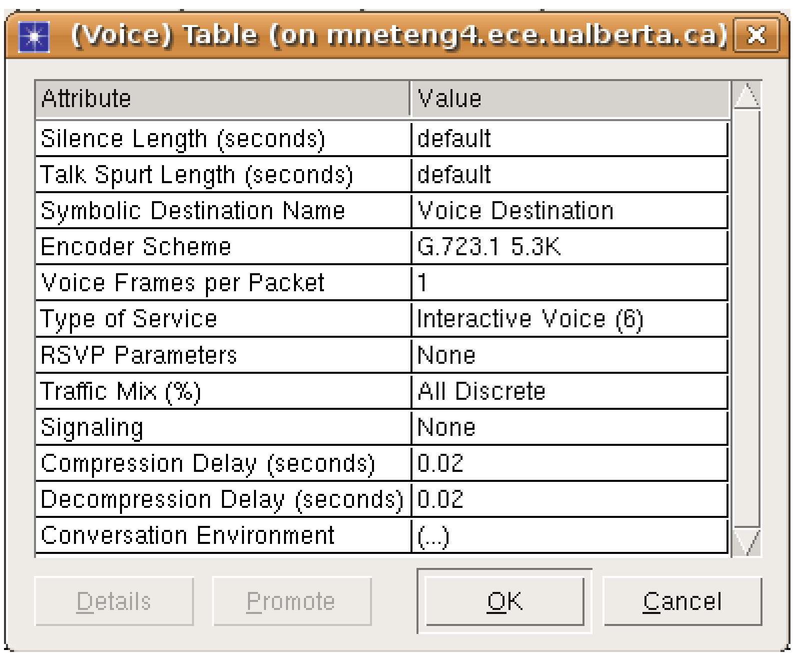


(Voice) Table (on mneteng4.ece.ualberta.ca)

Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.711
Voice Frames per Packet	1
Type of Service	Interactive Voice (6)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02
Conversation Environment	(...)

Details Promote OK Cancel

(a) G.711 Codec Parameters

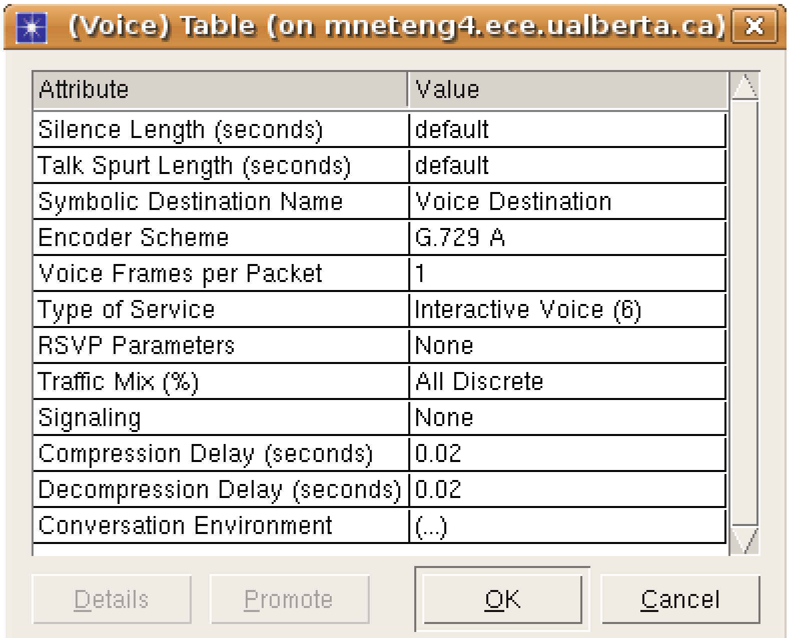


(Voice) Table (on mneteng4.ece.ualberta.ca)

Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.723.1 5.3K
Voice Frames per Packet	1
Type of Service	Interactive Voice (6)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	None
Compression Delay (seconds)	0.02
Decompression Delay (seconds)	0.02
Conversation Environment	(...)

Details Promote OK Cancel

(b) G.723.1 Codec Parameters



(c) G.729 Codec Parameters

Figure 4.2: Configuration of Voice Encoder Scheme Parameters

The “Symbolic Destination Name” is used for defining the destination node for VoIP calls. This attributes is also set to default value that means the destination node of VoIP calls is randomly chosen. In simulation topology, there are only two VoIP nodes, thus VoIP calls are received to each other. After defining and configuring the VoIP application, it is required to configure the way in which workstations are implemented in this application. In general, a profile defines the behaviour of a network workstation and contains one or more applications. These applications are configured by repeatability, start time, end time, and etc. The profile needs to be configured to add calls repeatedly at a fixed rate. In all scenarios, the first VoIP call generates after 120 seconds from the start of the simulation run because the “Start Time Offset” is set to be 60 seconds and the “Start Time” is set to be 60 seconds as shown in Figure 4.3. These start times are adjustable and can be helpful in debugging the simulation by making sure simulated traffic being properly generated at certain times. The “Number of Repetition” of VoIP application is set to be “Unlimited” to keep generating calls and the “Inter-repetition Time” is set to be 5 seconds, so VoIP calls are generating every 5 seconds. This process is repeated till the End of Simulation.

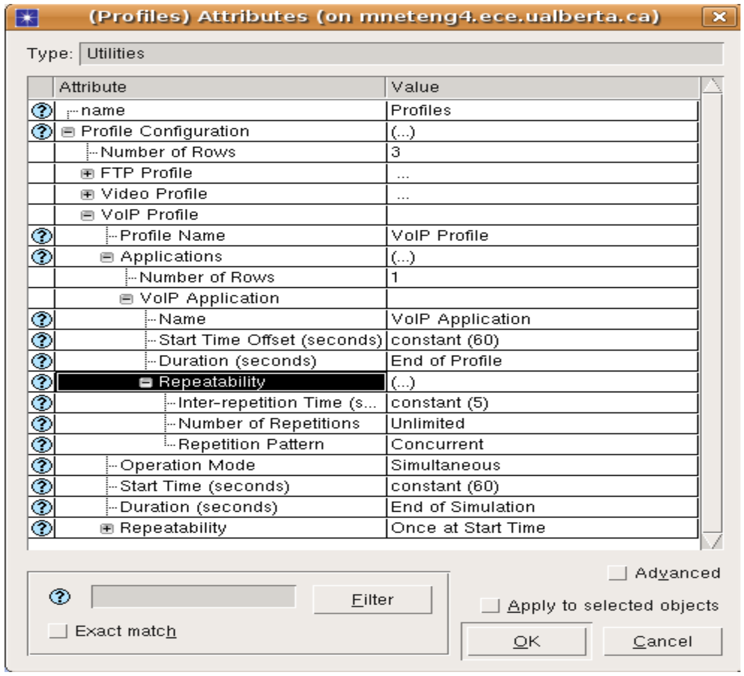
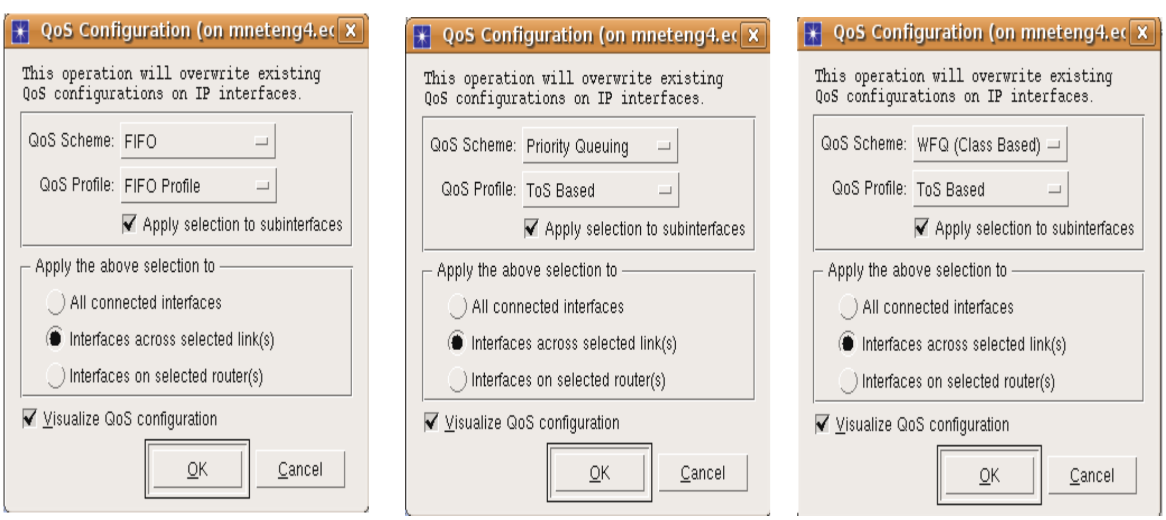


Figure 4.3: Profile Definition

QoS is set on the interface between routers. QoS Profile is set as default value in OPNET. PQ and WFQ are classified by Type of Service (ToS) as shown in Figure 4.4.

ToS classes supported by OPNET are total eight classes: best effort, background, standard, excellent effort, streaming multimedia, interactive multimedia, interactive voice, and reserved. The interactive voice class is configured because it provides highest class for low delay and low loss in comparison with the best effort class.



(a) FIFO (b) Priority Queuing (c) Weighted Fair Queuing

Figure 4.4: QoS Configuration

Chapter 5

Simulation Results and Data Analysis

5.1 Comparison of Performance Metrics

In Chapter 4, simulation modelling is described in detail. A number of statistics in the OPENT is required for VoIP components in order to obtain results before running simulations including VoIP traffic, routers, and links. This chapter presents simulation results for performance evaluation.

For each scenario the duration of OPNET simulation was set to 8 minutes due to memory limitation. The VoIP traffic was generated 120 seconds after the simulation was started. In all scenarios VoIP calls were generated at fixed time interval i.e. for every 5 seconds starting from 120th seconds till 480th seconds. Every simulation stopped at 8 minutes and the statistical and graphical results were generated by the OPNET Modeler 14.5 – Education version.

The summary of performance results obtained for simulations is shown in the Table 5.1.

Codec	QoS	Average Jitter (msec)	Average End-to-End Delay (msec)	Voice Traffic Sent (packet/sec)	Voice Traffic Received (packet/sec)
G.711	FIFO	-0.12657	1013.7973	25628	2534
	PQ	0.24912	172.2654	25715	4021
	WFQ	0.57829	595.7289	26201	2391
G.723.1 (5.3k)	FIFO	-1.00637	2104.5634	3667	494
	PQ	0.00011	105.7893	3779	3615
	WFQ	0.07118	180.0373	3763	2868
G.729	FIFO	-0.20693	1250.2498	10333	1397
	PQ	0.00005	66.1403	10566	5790
	WFQ	0.07236	251.2100	10375	3443

Table 5.1: Summary of simulation results

FIFO is a non-QoS-enabled queuing scheme and has the worst behaviour for real-time packets because it has a mechanism based on first-come, first-serve. As shown in Table 5.1, it has the high packet drops compared with PQ and WFQ. The average value of voice packet end-to-end delay is over 1,000 milliseconds which exceeds the threshold value of 200 milliseconds to maintain the minimum number of VoIP calls with acceptable quality, as explained in section 3.4.1. Therefore the results with FIFO are not considered to evaluate performance. Results with PQ and WFQ are mainly focused, analyzed, and compared to evaluate performance.

Figure 5.2 shows the number of packets that were sent, received, and dropped. Many packet drops are detected at all scenarios because FTP and video traffic were generated as background traffic and link utilization between routers was over 80% as shown in Figure 5. 1. Thus, the VoIP calls were experiencing loss of information due to the packet loss which caused voice breaks and voice skips.

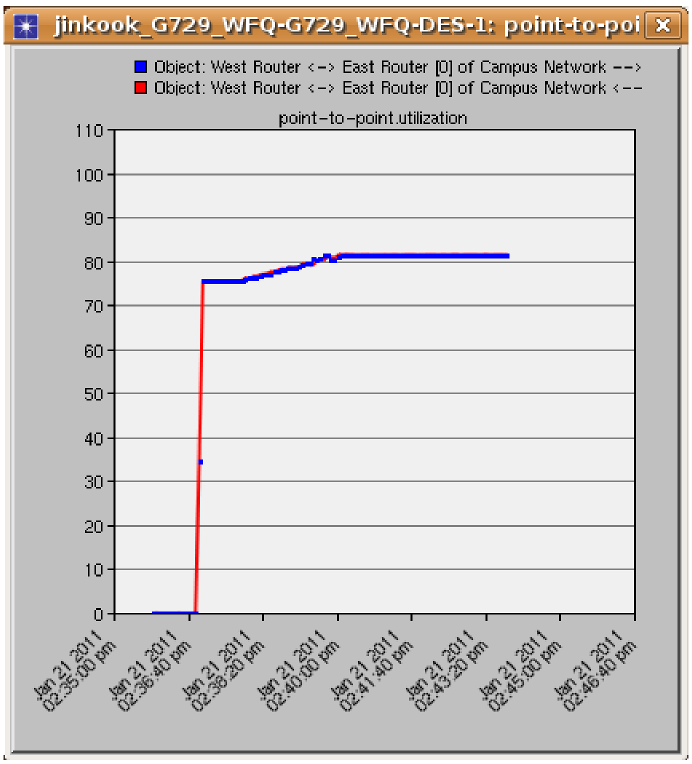
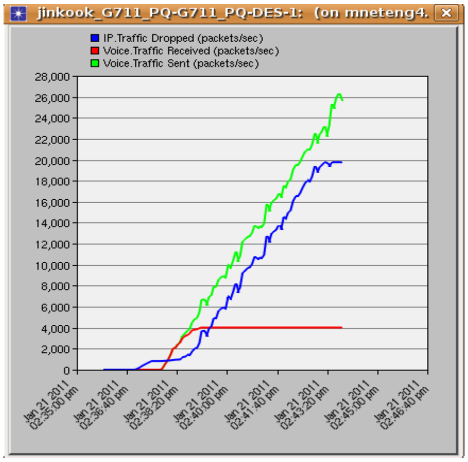


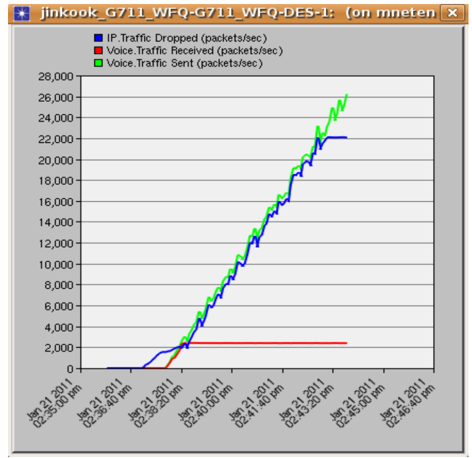
Figure 5.1: Link utilization between routers – G.729 + WFQ

Figure 5.2 shows the total VoIP traffic that was sent, received, and dropped and clearly shows that a receiver didn't receive all VoIP packets sent by a sender; i.e. there are mismatched packets between traffic sent and received. We can determine the number of calls that can be supported by examining the X and Y axes in Figure 5.2. X axis represents the simulation run time and Y axis represents a traffic volume.

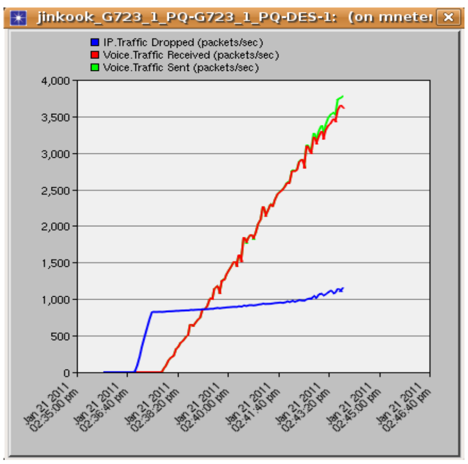
As shown in Figure 5.2 (c), G.723.1 and PQ had the lowest packet drops because G.723.1 (5.3k) consumed the lowest bandwidth per call and voice packets were sent with the highest priority on router interfaces.



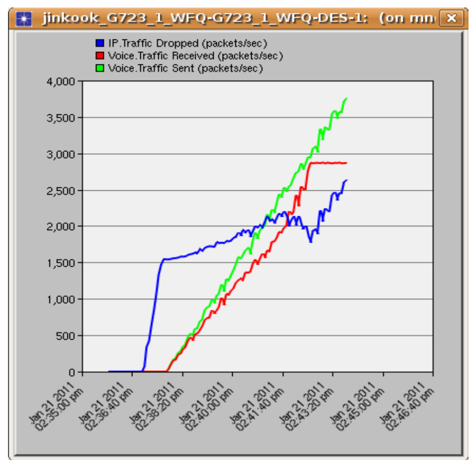
(a) G.711 + PQ



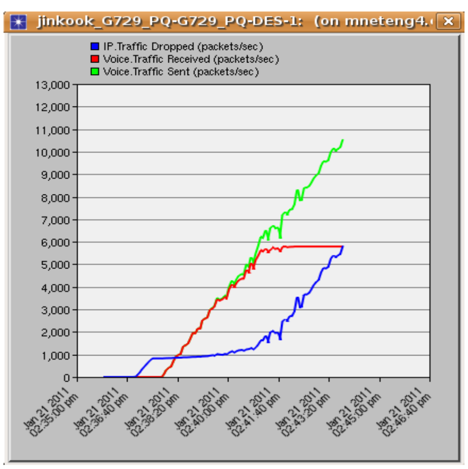
(b) G.711 + WFQ



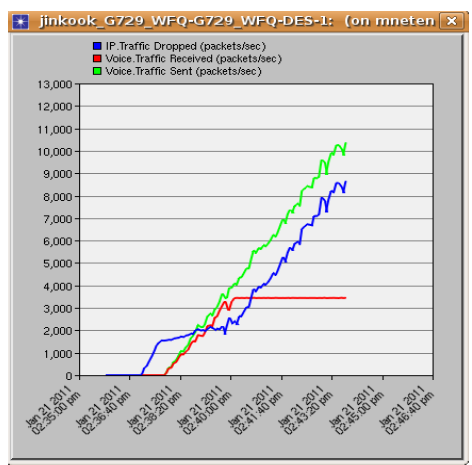
(c) G.723.1 + PQ



(d) G.723.1 +WFQ



(e) G.729 + PQ



(f) G.729 +WFQ

Figure 5.2: Voice Packet Dropped, Sent, and Received

As explained in section 3.4.2, the tolerable loss rates are within 1 -3 % and the quality becomes intolerable when more than 3 % of the voice packets are lost. Therefore, voice quality becomes intolerable. However, only G.723.1 with PQ has tolerable loss rates as shown in Figure 5.2. As a result, G.723.1 + PQ scheme provides the best performance from performance evaluation with voice quality parameters because it has the lowest bandwidth per call, lower end-to-end delay, and lower jitter value.

Figure 5.3 and 5.4 show the VoIP end-to-end delay. Remember, this delay should not exceed 200 milliseconds as discussed in section 3.4.1. All traffic on the network including VoIP, FTP, and video were generated at 120 milliseconds and the delay increased sharply at 120 milliseconds. However, average delay values of codec schemes with PQ have less than 200 milliseconds during simulation as shown in Table 5.1.

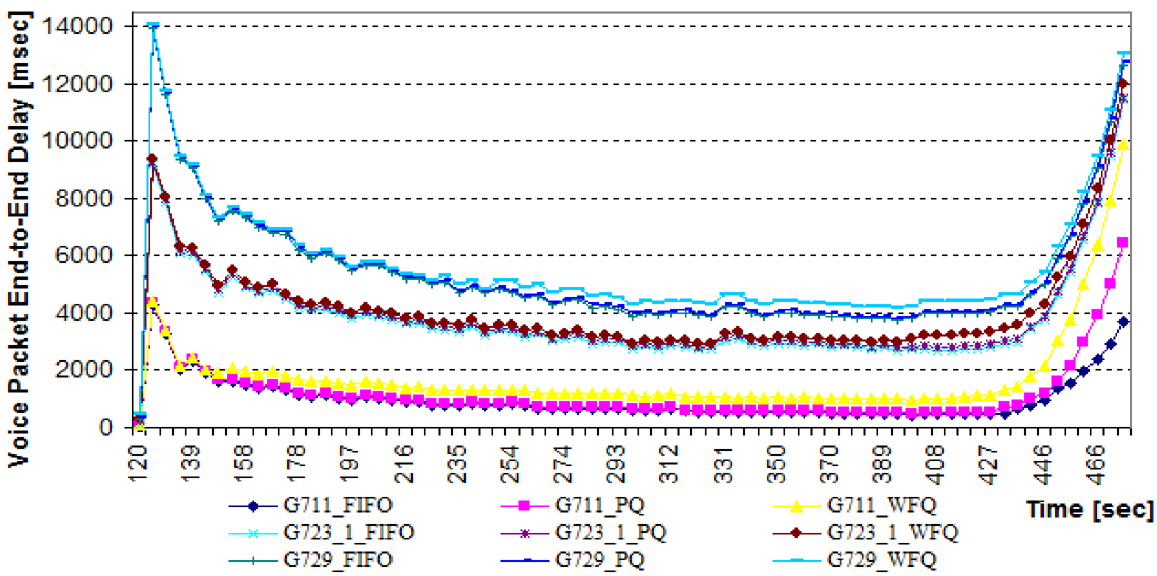


Figure 5.3: Voice Packet End-to-End Delay

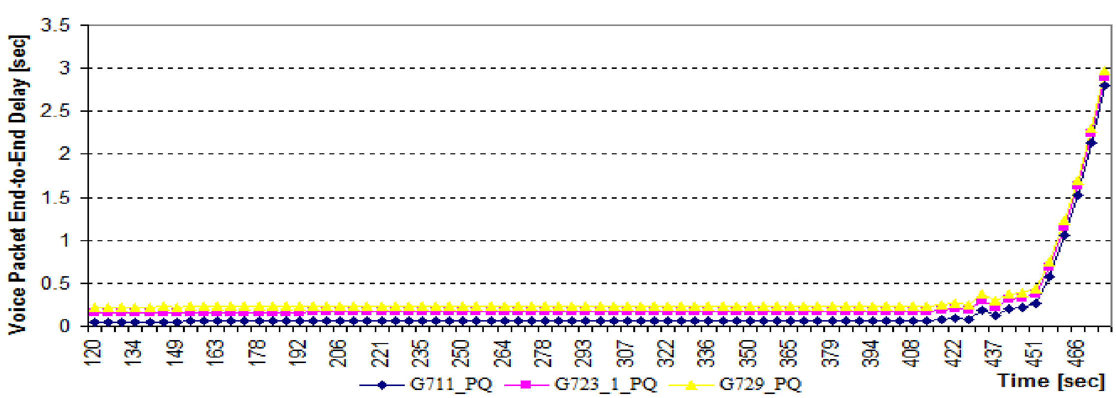


Figure 5.4: Voice Packet End-to-End Delay with PQ

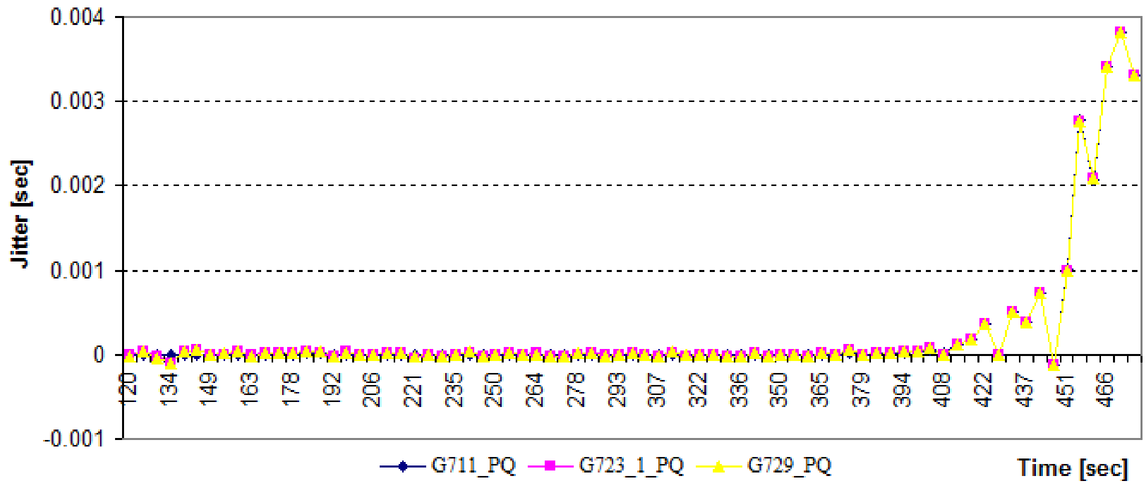


Figure 5.5: Voice Packet Jitter - PQ

Jitter values with PQ and WFQ are shown in Figure 5.4 and 5.5 respectively. All values are under threshold and have acceptable range (0 – 50 msec).

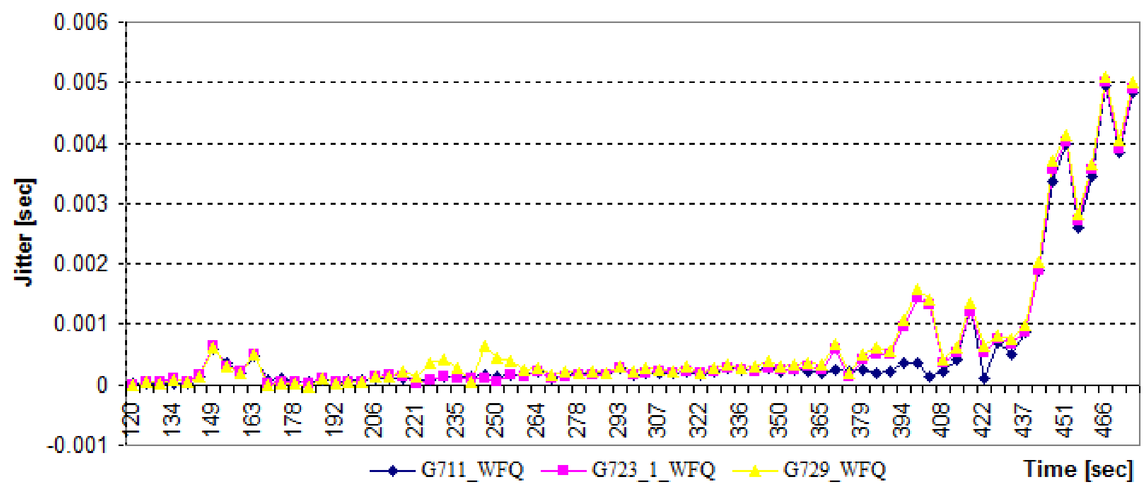


Figure 5.6: Voice Packet Jitter – WFQ

Jitter can have positive and negative values. The transit time between two consecutive packets will not be the same anymore: the second packet will have to go through a longer queue, spending more time, and generating positive jitter. Once burst is over, the queue will progressively reduce, reversing the situation. Out of two consecutive packets, the second one will spend less time in the queues, and will therefore generate negative jitter [36].

5.2 VoIP Traffic

The delay in the network must not exceed the threshold value of 200ms to maintain the minimum number of VoIP calls with acceptable quality explained in section 3.4.1. The numbers of VoIP calls that can be maintained in networks are estimated using the Voice Packet End-to-End Delay graphs.

The VoIP calls in the OPNET are added to the network by configuring the Application Definition and Profile Definition. The duration of the OPNET simulation was configured to run for 8 minutes. In all scenarios VoIP calls were generated at fixed time interval i.e. for every 5 seconds starting from 120^{th} seconds till 475^{th} seconds. The last successful two calls were at 8 minutes. The generation of background traffic by default in the OPNET started at 40 seconds from the start of the simulation run. In each scenario the total number of calls established is given by calculating total simulation time. Since for every 5 seconds one call is added to the network so the total number of calls maintained in the network is ((475 – 120) / 5) * 2 +2 = 144 VoIP calls. The calls calculated in the network models are varied depending on the traffic conditions and the network topology.

Conclusion

The main objective of this paper evaluates the performance analysis of VoIP traffic in respect of codec schemes and QoS using OPNET simulation tool. This report presents statistical and graphical analysis and can help network designers to choose a codec scheme and QoS before deploying VoIP on the real networks.
The performance evaluation is made on focusing on the performance metrics such as Voice Packet End-to-End Delay, Voice Packet Delay Variation, and Packet loss. These voice quality parameters are obtained by simulation.
The literature review was started and helped me how to approach the project topic and how to solve problems encountered during the project.
Based on the simulation results it can be concluded that both of codec scheme and QoS are affecting the voice quality because of the following reasons:

- Many packet drops are detected at all scenarios because FTP and video traffic are generated as background traffic and link utilization between routers is over 80%.
- Voice Packet End-to-End delays with PQ have less than 180 milliseconds and it doesn't exceeds the threshold 200 milliseconds.
- Voice packet can have the highest priority on PQ and claim 100 percent of link bandwidth with minimal delay and minimal jitter as shown in Table 5.1. However, other applications such as FTP and Video can starve the bandwidth.

G.723.1 uses the lowest bandwidth as described on section 1.1.1. Codec schemes with Priority Queuing have better performance than with other queuing. However, it also has the side effect as the above mentioned. Therefore, network engineers carefully select codec schemes and QoS to provide the best performance on the networks.

This paper focuses on the performance evaluation of three codec schemes and congestion management. The performance evaluation with other codec schemes and QoS mechanism such as congestion avoidance, traffic shaping, and etc is the future work.

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