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ACKNOWLEDGMENTS

The 'path to MSc' has been a very challenging and would consider myself achieving a milestone in my career. I would like to thank various people who have encouraged and have driven me to this goal. First and the foremost I would like to thank Professor and my supervisor Bill Buchanan for his extensive support and guidance throughout my dissertation. I thank Prof. Jim Jackson for his understanding and providing me his support.

In addition, I would like to thank Aegis Pvt Ltd India for providing me the infrastructure to conduct the experiment. I would like to thank my Manager Mr Sathish M.R and Mr. Jagadish Aradhya without which I could not have conducted this experiment.

Finally, I would like to thanks my friends, parents and my sisters who have contributed to my success.

ABSTRACT

The aim of this project is to identify and analyse different queuing mechanism and mark the traffic flows in real-time VoIP network. A prototype design is created to know the effect of each queuing technique on voice traffic. Voice traffic is marked using DSCP especially Expedited Forwarding (EF) PHB. Using Network monitoring tool (VQ manager) the voice traffic stream is monitored and QoS parameters are measured. QoS parameters are delay, jitter and packet loss. By analysing these QoS parameters, efficiency of each queuing technique is identified.

Experiments are performed on data and voice converged IP network. Voice is being sensitive to jitter, delay and packet loss, the voice packets are marked and queued to analyse four different queuing mechanisms such as Priority Queue (PQ), Weighted Fair Queue (WFQ), Class-Based Weighted Fair Queue (CBWFQ) and Low Latency Queue (LLQ). Each queuing mechanism has their own feature, in PQ higher priority queue has strict priority over lower ones [21]. WFQ provides fair queuing, which divides the available bandwidth across queues of traffic flow based on weights [23].CBWFQ is an extended from of WFQ, which guarantees minimum bandwidth based on user-defined traffic classes [19]. LLQ is the combination of PQ and CBWFQ.

The outcome of this project is to understand the effect of queuing mechanisms and classifying of traffic. Results obtained from experiment can be used in determining the efficient queuing technique.

APPENDICES

1. PROJECT PROPOSAL

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1 Introduction

1.1 Introduction

In ten years, the Internet has grown exponentially and has reached almost 3,000,000 hosts. There is a huge demand for integrating voice and data into same network. In 1990, many individuals in research background began to take an intense interest in carrying voice and video over IP network, this turned to be voice over IP (VoIP). Currently many of the organisations use VoIP. In order for VoIP to be a viable alternative to the traditional Public Switched Telephone Network, an evaluation of Quality of Service (QoS) of VoIP is required.

Organisations are switching to VoIP because not only is it cost efficient, but it also converges with data and transmitted in the same IP network. Voice is judged as real-time application on an IP network. This has to be treated with special treatment because of their sensitivity towards jitter, delay and packet loss. The special treatment or priority that is given to achieve high quality voice is called QoS.

This document starts with concise introduction to VoIP and QoS. This is followed by theory of VoIP with its architecture, protocols, VoIP system structure and Voice coding techniques. Later, it leads to a literature review, which includes marking of voice packets and queuing techniques. This is based on current and previous research papers and white papers. The chapter on methodology is described in the next chapter. The following chapter is implementation, which consists of topology and configuration, and the thesis ends with conclusion, critical discussion and further work.

1.2 Aims and objectives

In a network, a number of network impairments can affect quality of voice. Packet loss, delay and jitter are the most important network performance characteristics on IP networks that influence the resultant speech quality. Data and voice converged IP network addresses the issues of packet loss, delay and jitter. Voice is sensitive traffic hence it is treated with special priorities. The aim of this project is to identify and analyse different queuing technique and tagging for the traffic flows.

This includes evaluating each queuing technique by measuring the delay, jitter and packet loss and analysing the graph to prove which is more efficient for VoIP network. To analyse the voice traffic flow has to be marked for discriminating from the data network. This is done by marking of voice packets, using Differentiated Services Code Point (DSCP) and IP Precedence (IPP). In this project, DSCP is been used for marking and measured QoS parameters such as delay, jitter and packet loss.

1.3 Background

VoIP has gained lot of attention as a replacement of traditional telephony especially in business. Most of the contact centres have migrated to VoIP; the reason behind this is not only lower call rates, but its ease of integration of voice and data traffic in the same network and across multiple sites [1]. Implementing VoIP on a data network and making a high quality calls is a challenge that involves large number of factors. These factors are speech, codec, packetization, packet loss, jitter, signalling protocol and QoS [2].

Voice, video and data, requires special treatment due to their sensitivity to delay, jitter when deployed on network. Traditionally circuit switched networks carrying voice provided deterministic where delay, jitter and error rates were constant. However when using packet switched technologies, application such as VoIP have to contend for network resources available, as these networks does not guarantee to voice that are required.

Voice is known as an application on IP network, voice traffic must be given priority, over other applications contending for the same bandwidth. The technique for providing such priority is known as Quality of Service (QoS).

QoS is very important in a VoIP network, which solves lot of issues like, packet loss, jitter and delay. VoIP is less tolerant towards packet loss and jitters. A QoS implementation in IP network is well defined. IEEE 802.1p, a subset of 802.1q (2003) and newly integrated into 802.1D (2004) provides Ethernet switches to prioritise at Layer 2 of the OSI 7-Layer model. In Almquist's (1992) Request for Comment, RFC 1349, the Type of Service (ToS) byte is re-defined for IP packets to provide a similar role at Layer 3. In addition, Differentiated Services (DiffServ), defined by Nichols et al. (1998) in RFC 2474, redefined the entire ToS byte, for smoother network classification.

1.4 Thesis structure

The thesis structure is:

- Chapter 2 (Theory). This provides brief background of VoIP networks, protocol and structure of the network, and closes with review of voice coding technology and codecs. The architecture is described to exhibit the method of converting and carrying an analogue signal (voice) across an IP network. The Real Time Protocol (RTP) is considered with its ability to enable the synchronisation and sequencing of voice while dispatched across protocols such as UDP and IP.
- Chapter 3 (Literature Review). This provides concise description on QoS in VoIP. This is followed by a discussion on QoS parameters such as packet loss, jitter and delay and different types of marking of voice packets such as IPP and DSCP. In this chapter, recent research papers and reviewed which are based on the queuing techniques and their approach to their determining the voice quality. This chapter ends with conclusion.
- Chapter 4 (Methodology). This outlines the experimental design with a technical specification of each device and topology of the network.
- Chapter 5 (Implementation). This deals with the implementation of the network topology and configuration. Experiments are described which are conducted on two baselines but same topology. The first experiment is been done without any QoS applied and the second experiment is sub divided into four experiments. The experiments performed using four different queuing techniques.
- Chapter 6 (Evaluation). This investigates possible approaches to improve the QoS in VoIP network. This chapter deals with result and analysis of the two experiments, which were discussed, in the pervious chapter. First experiment is conducted without implementing QoS on the network and measured jitter, delay and packet loss. Moreover, the second experiment was performed using four different queuing techniques such as PQ, WFQ, CBWFQ and LLQ.

2 Theory

2.1 Introduction

This chapter provides a short background of VoIP networks, protocol and structure of the network, and closes with review of voice coding technology and codec. The convergence of data and voice networks has widened in real-time application like VoIP. Usually voice networks are separated from data networks due to the protocols which is been used and features of voice application is very different from other application. Still, the advantage of implementing a converged network for the support of voice and data has resulted in an increase in the use of IP for the transportation of voice services.

2.2 Voice over IP

Voice over IP (VoIP) converts audio signals into digital data, which can be transmitted over Internet. This is a revolutionary technology, which is replacing the phone system. VoIP has gained a lot of attention from number of organisations and is growing steadily. Traditional PSTN are resource-dedicated, where IP network are resource-shared. VoIP is the convergence of voice on the data network by using IP. This encapsulation of the voice transmission allows the two networks to become a single network. This lowers the cost of the organization by managing voice and data in only one network.

VoIP is a transport mechanism for supporting voice traffic between Private Branch Exchanges (PBXs) via IP. It is been deployed on the trunk side of the PBX and makes use of IP gateways to access the IP network. This makes organization to use existing telephony network components such as PBX, telephone instruments and internal cabling infrastructure.

2.3 VoIP Protocol Architecture

The VoIP protocol architecture is mixture of many interrelated protocols. The Real-time

Transport Protocol (RTP) [25], Real-Time Control Protocol (RTCP) and H.323 [26] or Session Initiation Protocol (SIP), [27] for call signalling.

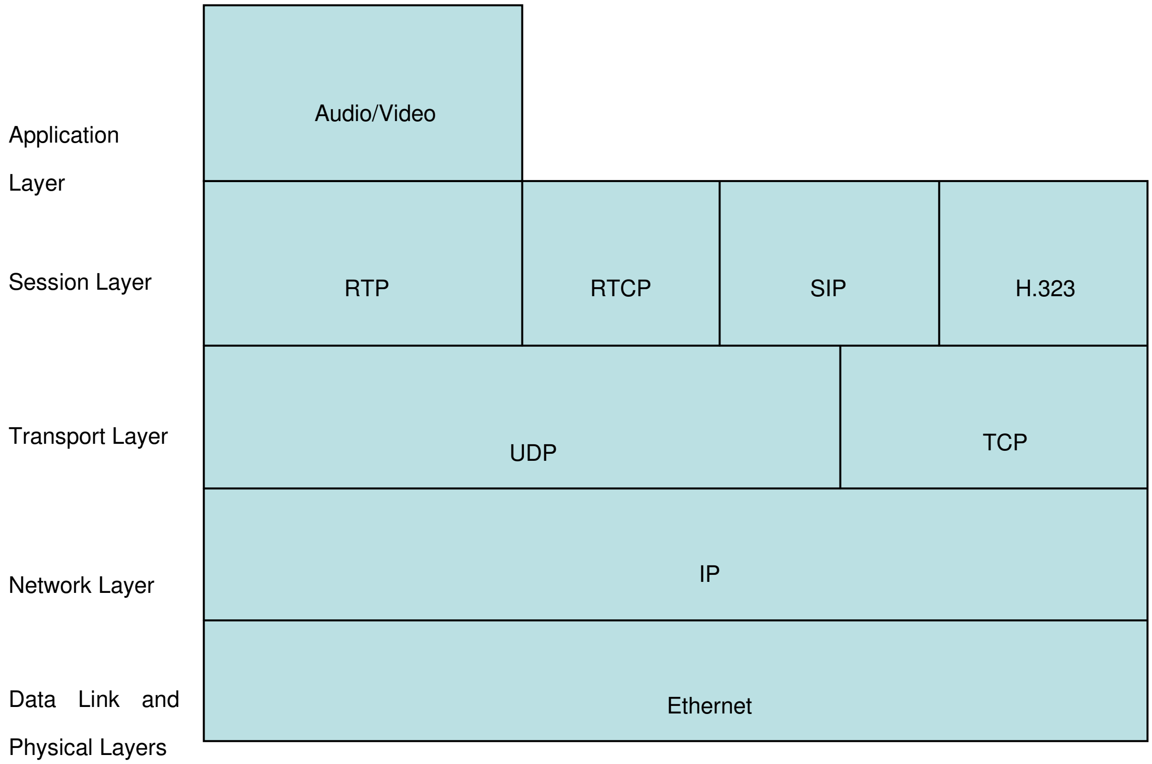

Figure1 VoIP Protocol Architecture

2.2.1 Real-time-Protocol (RTP)

Most of voice and video applications use RTP for data transmission on IP networks. RTP runs on the upper layer of the transport protocol UDP to make use of its checksum and multiplexing services, and give real-time applications such as voice end-to-end delivery services such as payload type identification, sequence number, time stamping and delivery monitoring.

The RTP header has timing information and sequence number, which allows receiver to rebuild the timing information of the sender packets. In IP networks, there is rarely loss and reorder of packets. The RTP header helps the receiver to rebuild the timing produced by source by using its timing information and sequence number.

V=2	P	X	CC	M	PT	Sequence No.
Timestamp						
Synchronisation source (SSRC) Identifier						
Contribution source (CSRC) Identifier						

Figure 2 RTP Header [25]

The RTP payload contains sample of voice and follows the RTP Header, and the sequence number is made up of seven bits. It is incremented by one for each RTP packet was sent and is used by the receiver to detect the packet, which is lost, and recover the sequence.

The timestamp reflects the sampling instant of the first octet of the sample contained in the payload of the RTP packets and is incremented by one for each data sample, whether the data is transmitted onto the network or dropped as silent. The timestamp allows the receiver to calculate the arrival jitter of RTP packets and synchronise them with the sender.

2.4 VoIP System Structure

VoIP system structure comprises of three parts- the sender, the IP network and receiver. At the sender, an analogue voice is sent which is digitised and compressed by encoder. Many encoded speech frames are packetised to form payload inside the RTP datagram. This is encapsulated with UDP and IP to form IP packet, which is forwarded to the IP network.

The incoming packet at receiver end is extracted by using de-packetiser. The jitter is patched up by using play-out buffer, results in a constant stream of speech frames, which are then decoded to produce the voice stream to the user. The packet loss is recovered by codec used at receiver end. This can be done using Packet loss concealment techniques. The previous frame received is inserted to patch up the lost packets in the place of silent periods.

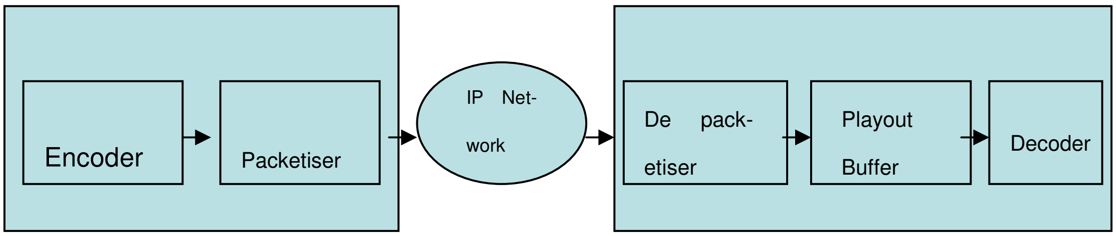

Figure 3 VoIP system structure

2.5 Voice Coding Techniques

The voice coders are referred as codec. It converts the analogue signal usually human speech to digital data. There are three different kind of speech coding technique:

- **Waveform codec:** It preserves the general shape of the signal waveform and tries to explore the relation in time-domain and frequency-domain. E.g. G.711 PCM at 64 Kbps [28] and G.726 ADPCM at 40/32/24/16 Kbps [29].
- **Voice Codec:** It is a simple speech production model and does not try to preserve the original waveform. E.g. 2.4/1.2 Kbps LPC.
- **Hybrid Codec:** It is made up of waveform and voice codecs. They are made up only advantages of both the codecs to achieve good speech quality at 4.8 and 16 Kbps, such as G.729 CSACSELP (8Kbps), G.723.1 MP-MLQ/ACELP (6.3/5.3 Kbps), AMR (Adaptive Multi-Rate, ACELP), and iLBC (Internet Low Bit Rate Codec).

The following three ITU based audio codecs are used frequently in VoIP application.

- **G.711** uses semi-logarithmic scale called Pulse Code Modulation (PCM) to digitize the analogue data. Objective of PCM is to increase the resolution of the small signals when large signal are treated proportionally. The encoding stream is 64 Kbps, consists of 8 KHz sampling of 8 bit signal. The length of the frame is 1ms.

- **G.723.1** codec has been selected as baseline codec for the narrowband H.323 communications by the International Multimedia Telecommunications Consortium (MTC) VoIP forum. G.723.1 is used for compressing the speech component of multimedia services at a low bit rate (Compared to G.711's 64 KBPS). The hybrid has two bit rates associated with it, 5.3 and 6.3 Kbps, whose mode of operation can change dynamically at each frame. The frame length is 30 ms; however, another 7.5 ms delay is necessary for it look-ahead buffer, resulting in a total algorithmic delay of 37.5ms. The G.723.1 encodes speech in frames using linear predictive analysis-by-synthesis coding. The excitation for the high rate coder is multi-pulse-maximum likelihood quantization (MP-MLQ), whereas the low rate coder is algebraic-code-excited linear prediction (ACELP). The codec is capable of providing silence compression: Voice Activity Detection (VAD), Discontinuous transmission (DTX) and Comfort Noise Generation (CNG).
- **G.729A** codec make use of Conjugate-Structure, Algebraic-Code-Excited Linear Prediction (CS-ACELP) coding technique. The speech rate is 8Kbps and algorithmic delay is 15ms (10ms frame length and 5 ms of look-ahead time). G729A is a reduced-complexity version of G729.

2.6 Conclusions

The intention of this chapter was to present a background of VoIP that are been used for voice transmission in this study. The essential VoIP connection types, VoIP are been explained with their deployment in a network infrastructure.

The architecture is described to exhibit the method of converting and carrying an analogue signal (voice) across an IP network. The RTP is considered with its ability to enable the synchronisation and sequencing of voice while dispatched across protocols such as UDP and IP in order to reform the speech at destination is coherent to the end-user.

The VoIP structure is discussed with the procedure of digitising and encoding an analogue voice flow, and packetising using protocols such as RTP, UDP and IP. This section ends with description of voice coding technology that is used in VoIP networks; G711, G.732.1 and G.729A are been explored.

3 Literature Review

3.1 Introduction

The chapter starts with brief discussion on QoS parameters (packet loss, jitter and delay) and different kinds of markings of voice packets (IPP and DSCP). In the next section recent papers (journals) are discussed which are based on different queuing techniques and methods of analysing and determining voice quality. This chapter is followed by Methodology.

3.2 QoS in VoIP

This section deals with the QoS in Voice network. QoS parameters like packet loss, jitters, delay, marking of the voice packets and queuing techniques are discussed.

3.3.1 Packet Loss

Packet loss is a main cause of speech destruction in VoIP networks. It is the measure of the number of packets that were not received compared to the total number of packets transmitted [3]. This happens due to peak loading and periods of congestion. There is major issue with packet loss, voice packets are using UDP for transport and as a result do not guarantee delivery of the packets. In VoIP networks, packet loss result in short periods of silence and voice distortion. The codec determines the effect of lost packets to the listener on VoIP. There are certain codec, which reduces the effects of packet loss. But as loss increases the voice signal is distorted at end user [8]. The codec are discussed later in this document. VoIP packets are small containing payload of 10-15 bytes [7]. The loss of this small packet is negligible, but it is not lost in isolation. Loss of one packet also affects QoS by losing several connected packets.

3.3.2 Jitter

The variation in the packet arrival time is called as Jitter. Most of the time jitter is caused due to low bandwidth and may cause severe dent to overall QoS [5]. Figure 4 shows the difference between normal and jittered stream. Jitter cause the packet to arrive and processed out in a random manner.

RTP is based on UDP, so the reassembling and processing of the packet will not happen at protocol level. However, time stamp and sequence number fields of the packet are used to reorder the packets [5].

When jitter is high, packets reach destination rapidly. This is somewhat similar to road traffic coming to a stop to a traffic light but as soon as the light goes green, the traffic goes in a rush. One approach to avoid jitter is to use buffer at end points, but these buffer has to release packets at every 150 ms or even sooner because of transport delay. According to Khun, Walsh and Fries [6] *the buffer packet is simply delayed an anomalously long amount of time, or it is actually lost. If jitter is particularly erratic, then the system cannot use past delay timer as an indicator for the status of missing packet. This leave the system open to implementation specific behaviour regarding such as packet.*

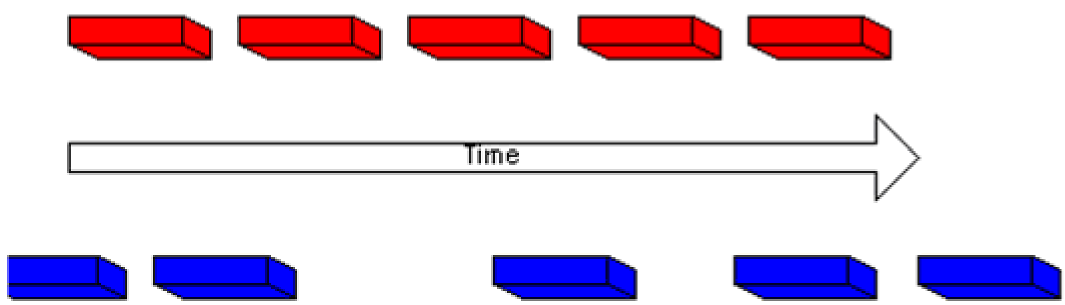

Figure 4: The difference can be seen in blue coloured packets (Variation in arrival) - Jitter Example [5]

3.3.3 Delay

Time elapsed between sending and receiving a packet between two devices is called end-to-end delay. Delay consists of following components:

- **Propagation delay**: depends on the physical distance of the communications path and the communication medium.
- **Transmission delay:** the total time taken from the network interface to send out pack-

ets on to the medium.

- **Queuing delay**: the time spent by a packet in the queues at the input and output ports before it is processed. It is mainly due to congestion in the network.
- **Codec processing delay**: consists of codec's algorithmic delay and look-ahead delay.
- **Packetization/depacketisation delay:** the time taken to assemble packet at the sender end and time taken to strip the headers at the receiver end.
- **Play-out buffer delay**: the time taken at play-out buffer at receiver end.

One-way end-to-end delay should be less than 150 ms for most of the applications. Delays of 150-400 ms are acceptable if administrators are aware of the time impact on the transmission quality to the user [8]. Also delays of over 400ms are unacceptable for general network planning purposes [8].

3.3 Marking of Voice packets

Classification of traffic is a important factor when it comes to mixed network. The reason behind the discrimination of traffic is to mark the packet with a 'flag' to make them relatively more or less important than other packets on the network. This decides which packet to reach destination first or which one to drop [10]. Classification identifies a certain type of traffic where as marking is assigning a value to that class of traffic.

Marking techniques occur at Layer 2 and Layer 3. This section deals with Layer 3 marking that is IP precedence and a major discussion on Differentiated service code point (DSCP).

3.4.1 Type of Service (ToS)

ToS is 8-bit field composed of three fields; Figure 5 demonstrates the ToS fields. The first three bits are for IP precedence, next four bits are service provided indicator and the last bit is unused [12]. The second field below shows how network should make transaction between throughput, delay, reliability and cost [12].

IP Precedence			Type of Service				MBZ
1	2	3	4	5	6	7	8

Figure 5 ToS Fields [11]

The objective of ToS is an indication of rough parameters of QoS desired. These parameters guide actual selection service parameter while transmitting the datagram through network [12]. To achieve the objective of ToS defined by RFC791, ToS is composed of two subfields, the service profile and Precedence field.

Figure 6 illustrates the bits of Service profile field. RFC 791 acknowledges use of delay; throughput and reliability will increase the cost of the service and says that not more than 2 bits are to be used except in unusual cases. This failed in defining the feature of data streams in the network [11].

0	1	2	3	4	5	6	7
Precedence			D	T	R	O	O

According to RFC791, service profile field represents bits 3, 4 and 5 of the ToS field.

Bit 3: 0 = Normal Delay 1 = Low Delay

Bit 4: 0 = Normal Throughput 1 = High Throughput

Bit 5: 0 = Normal Reliability 1 = High Reliability

Figure 6 Service profile bit parameters [10]

This service profile was modified and redefined by RFC 1349. Instead of using 3 bits service field, they introduced 4-bit service field [12]. This gave three level of matching the single bit selector and provided the fourth value for minimising the cost [12].

0	1	2	3	4	5	6	7
Precedence			X	X	X	X	O

1000 -- Minimize Delay

0100 -- Maximize Throughput

0010 -- Maximize Reliability

0001 -- Minimize cost

0000 -- Normal Service

Figure 7: Service profile bit parameters and Bit String meaning [12].

3.4.2 IP Precedence (IPP)

The first 3 bits of ToS field are known as precedence subfield that is shown in figure 7. The basic purpose of precedence subfield is to indicate the router the level of packet drop preference for queuing delay avoidance. The precedence subfield was never defined perfectly, always a generalised rule was implied, that a packet with higher priority was routed first then the lower priority packets.

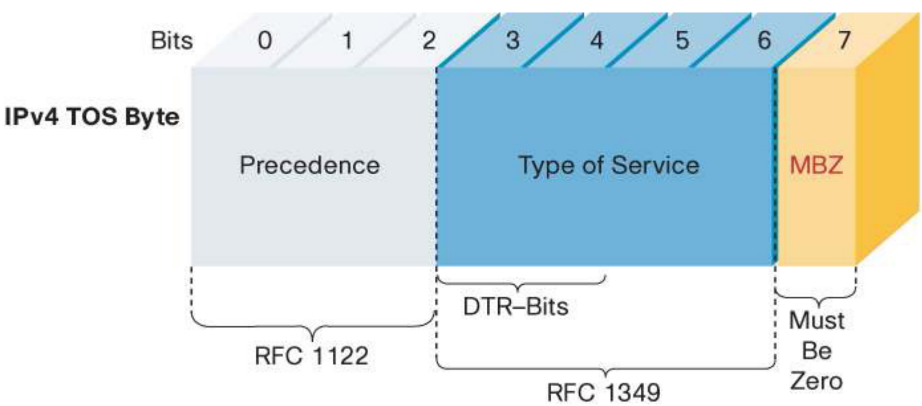

Figure 7 [13]: IPv6 TOS Byte

Precedence Bit Setting Definitions [10]

111 — Network Control (Reserved)

110 — Internetwork Control (Reserved)

101 — CRITIC/ECP

100 — Flash Override

011 — Flash

010 — Immediate

001 — Priority

000 — Routine

IPP value 6 and 7 are reserved for network control such as routing.

IPP value 5 is for voice.

IPP value 4 is for video conferencing and streaming video.

IPP value 3 is for voice control.

IPP values 1 and 2 are for data application.

IPP value 0 is for default marking value.

IPP allows only specification of relative priority of a packet. For example if a network administrator wants to prioritise the two different kinds of traffic at the same priority, during congestion, one of the traffic should be dropped, which is not important at that moment. It will not be possible to do this in IPP. IPP 3-bit limit the possible priority classes [13]. This reduces the successful implementation of QoS end-to end.

3.4.2 **Differentiated Services (*DiffServ) [14]***

DiffServ is a well-defined architecture that guarantees QoS in IP networks under standardisation of IETF. It operates under Layer 3 and uses ToS in the IPv4. ToS is used for marking of the packet to receive a particular forwarding treatment [13]. Figure 8 shows that Differentiated Services Code Point (DSCP) uses 6 bits from the total eight bits of ToS field [14].

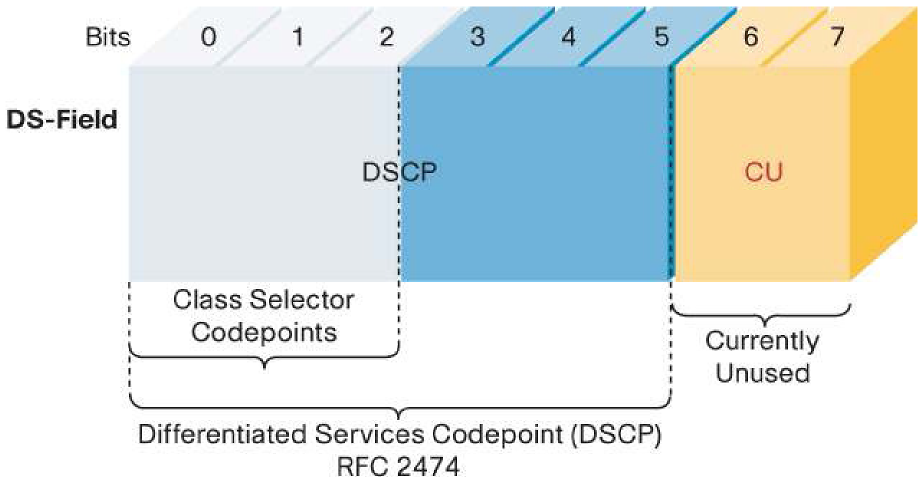

Figure 8 DSCP Field [13]

IPP is completely redefined; here six bits are used to classify the packets. The field is called DS (Differentiated Services) field where two bits are unused. The three bits are replaced by six bits and it is called as DSCP. According to RFC 2474, DSCP can support 64 classes; all classification can be done using DSCP [13, 14].

Packets can be marked using DSCP and meaningful QoS is provided by applying forwarding behaviour at DS complaint node. This forwarding behaviour is called as Per Hop Behaviours (PHB) [13, 15]. PHB [15] refers to packet scheduling, queuing policing and traffic shaping of a particular node belonging to same behaviour aggregate. There are four available standard PHB

- **Default PHB** –specifies a packet marked with a DSCP value of 000000 obtain best effort service from a DS-complaint node. The packet marked with 000000, which turns up at DS-complaint node, will be mapped to default PHB [12, 15].
- **Class Selector PHB-**DiffServ has defined DSCP value with xxx0000 is called Class Selector code points, where x is either 0 or 1. These PHBs retain the same forwarding behaviour as IPP classification and forwarding. For example, if a packet has a DSCP value of 110000, it is equivalent to IPP, with a value of 110. These values feature forwarding treatment DSCP and IPP. This guarantees that DS-nodes can coexist with IPP nodes [15].

- **Assured Forwarding (AF) PHB** [17] - This is method by which behaviour aggregate can be give different forwarding assurances. In this traffic can be classified and allocated with the available bandwidth [17]. The AFxy PHB defines the four AFx classes: AF1, AF2, AF3 and AF4. Each class is assigned with certain buffer space and interface bandwidth, dependent on Service Level Agreement (SLA) with the service provider or policy. In each AFx class, it is likely to specify 3-drop precedence values. For example if there is any congestion in one of the link, we can drop packets of particular AFx class, consider AF1 need to be dropped it will be dropped in this way dp (AF11) <= dp (AF12) <= dp (AF13) the last digit in each AFx class represents the drop precedence. This concept is useful in controlling the flow within the behaviour aggregate that go beyond the allocated bandwidth [17, 15].

Table 1 DiffServ AF Code point (RFC-2597)

Drop Prece-dence	*Class#1*	*Class#2*	*Class#3*	*Class#4*
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF21) 001100	(AF22) 010100	(AF32) 011100	(AF43) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

- **Expedited Forwarding (EF) PHB** [18] - can be used in VoIP networks, and ensures of low loss, low delay, low jitter and assured bandwidth [18]. It is similar to RSVP [18], and give end-to-end services in the network domain. EF PHB provides virtual leased lines for optimal efficiency. This can be implemented by using priority queuing, which is used only on critical application like voice, which requires low latency and low loss, assured bandwidth [18].

DSCP provides QoS for varying network traffic. Policing and classification are done on the boundaries of DS domain. There is no need of negotiation for each flow as in integrated services. Policies are not standardized it is difficult to predict the end to end behaviour. If the packets are dropped in, the core network, which is using lot of resources and are, wasted. This is designed only on core network not on the access network.

3.4 Queuing Techniques

In networks, packets can be handled first come first serve basis, but in certain circumstances such as speed mismatches, when the packet is entering the device congestion occurs. The devices have to buffer for allowing the higher priority packets to exit sooner than lower priority ones, which is called as queuing [14].

Queuing algorithm are activated, when congestion is triggered and deactivated when congestion clears [14]. Available queuing techniques are covered in the following sections.

3.2.1 PQ (Priority Queue)

In PQ [20], higher priority traffic stream is transmitted before lower priority. According to theoretical and practical effort of Zhi Quan says that higher priority queue has strict priority over lower ones. In Zhi Quan's paper, it is assumed that class 1 has highest priority; class 2 has the second highest priority, and so on. The lower priority queues have no effect on the evolution of the workload process of the higher priority queues. On the other hand, the lower priority class traffic will be under the influence of higher and equivalent priority class traffic. Figure 9 provides an insight to how the priority assignment and the class traffic intensity affect the queue length. As the priority decreases, all the higher priority classes will influence the queue length. These priority classes will have an apparent advantage in queuing-time and required system queue-length [22].

The utilization increases on deploying the PQ, which point out the benefits of PQ, are more under intense traffic environments [22].

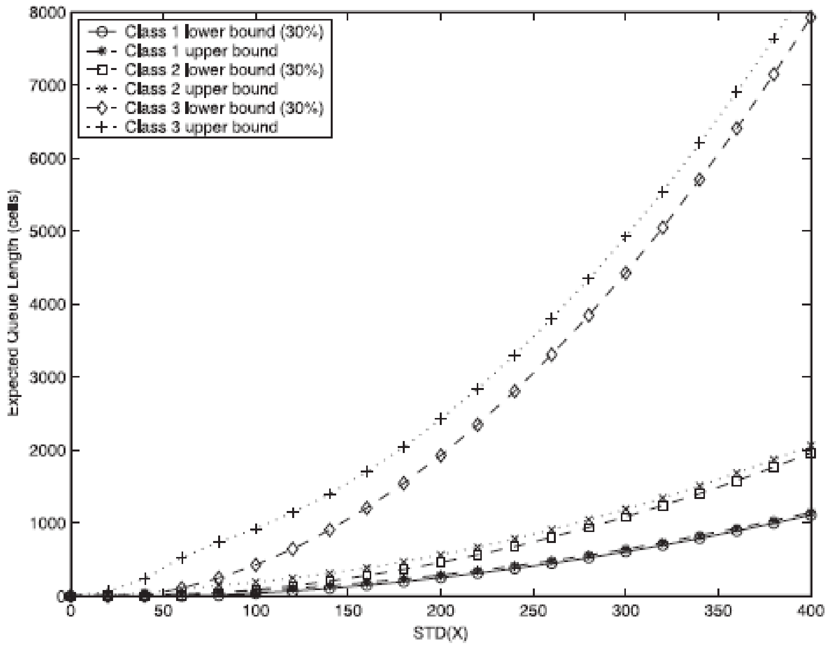

Figure 9: PQ [21]

3.2.2 WFQ (Weighted Fair Queue)

WFQ is defined as:

> "Provides fair queuing which divides the available bandwidth across queues of traffic flow based on weights. Each flow is associated with an independent queue, assigned with a weight to ensure that important traffic gets higher priority over less important traffic". [23]

Jeong-Soo Han, et al conducted experiments to compare the QoS parameters by using WFQ and FIFO on a VoIP network. They have conducted simulation studies by differentiating traffic volume and weight of various application services in order to find out how variable network conditions have an influence on end-to end delay. Experiment is divided into two steps. First, comparing the changes in delay time of voice traffic, which varies, with the volume of various applications. Based on the results of the experiment voice codec algorithm and queue management technique are selected keeping delay to minimum. For the second step the voice codec algorithm and queue management technique is selected to compare delay time of voice traffic is weighted differently. The experimental setup – packet is loss is restricted to 0.1% to maintain a stable network condition. Delay is set to 50 ms. LAN and backbone utilisation rates are assumed to be 20% and 70% respectively. Servers, which generate traffic on lacal LAN, are 10 servers and 1 server generating voice traffic.

The first simulation is conducted to study the changes in delays of voice traffic, which vary with the volume of other applications transmitted simultaneously along with voice traffic. FIFO and WFQ is compared where the results are shown in the table 2

Table 2 End-to end delays (ms) [4]

Queuing algorithm	Data traffic	Other application service traffic (G.723.1/G.729)		
		Web	FTP	Email
FIFO	256 (Kbps)	0.061/0.056	0.081/0.076	0.118/0.087
	512 (Kbps)	0.085/0.081	0.134/0.129	0.202/0.172
	1024(Kbps)	0.189/0.15	0.231/0.222	1.456/0.829
WFQ	256 (Kbps)	0.06/0.053	0.079/0.076	0.109/0.087
	512 (Kbps)	0.079/0.08	0.131/0.13	0.176/0.169
	1024 (Kbps)	0.18/0.11	0.213/0.191	1.352/0.813

According to results, WFQ is proved better than FIFO. The paper says that as end-to end voice traffic user increases the changes in delay patterns will increase. By using efficient voice codec and queue management techniques such as G.729 and WFQ can maintain changes in delay patterns at as small levels as possible.

The results of first simulation lead to second simulation where different weight is allocated to each traffic in order to find out the differences in wait time while queuing and jitter performance of each traffic.

They allocated different weights to each traffic flow in order to find the differences in wait time, while queuing and the jitter performance of each traffic flow. The weight of voice traffic was 5% and 10 % where voice traffic flow was not delivered so efficiently and load was high, when they changed the weights of voice, traffic to 30% and 60% the performance was better, but it was not acceptable. This is clearly illustrated in Figure 10 and 11. They say that WFQ shows better performance in delivering the voice traffic and that among various application services, and has observed that, during high load weighted voice packets show better delay and jitter performance, but not acceptable for a good quality of voice traffic. This shows that WFQ cannot handle voice traffic on a high load and does not assure QoS of voice traffic [4].

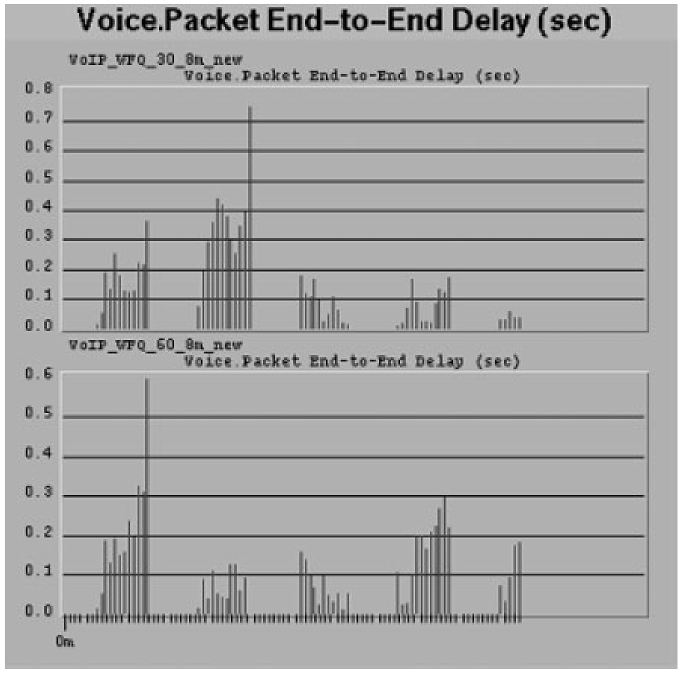

Figure 10 Delay patterns of voice traffic with weights of 5% and 10%[4]

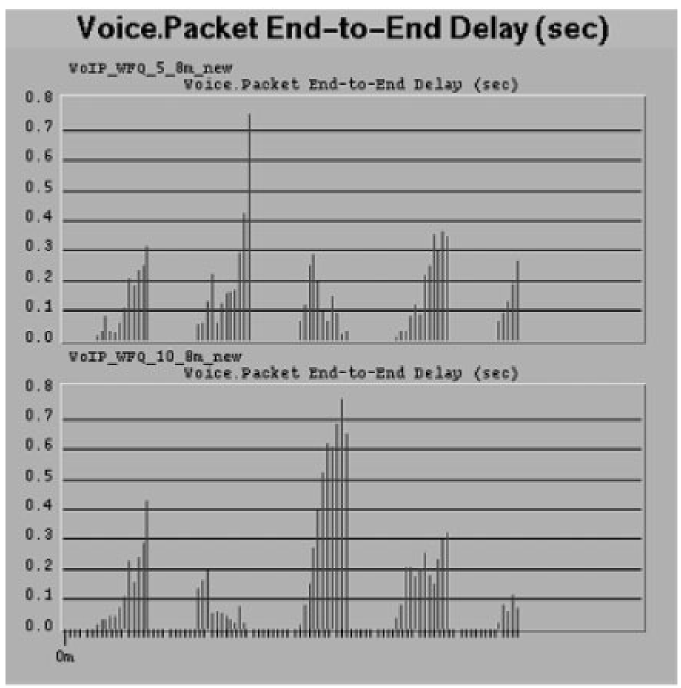

Figure 11 Delay patterns of voice traffic with weights of 30% and 60% [4]

3.2.3 CBWFQ (Class Based Weighted Fair Queue)

An extended WFQ, which guarantees the minimum bandwidth based on user-defined traffic classes, is called CBWFQ. It buffers for each class of traffic and sets the bandwidth for them. When one class of traffic is not using its allocated bandwidth, it allows the other class to utilise its bandwidth and allow for overflowing [19].

Masi et al conducted an experiment to compare the sensitivity of the performance of CBWFQ. In this paper, they have selected three main approaches to CBWFQ scheduling for further investigations and comparisons.

- Random selection of the class for transmission based on the weights.
- Golestani's virtual finish time approach.
- OPNET Modeler's implementation of CBWFQ.

They have shown that the Golestani approach and OPNET Modeler's implementation of CBWFQ have lowered the packet queue, which waits more than their Random Selection based on the weights method of modelling CBWFQ. They have investigated that the Random selection method gave higher estimates than Golestani's approach for classes, with more than enough allocated bandwidth to carry the traffic. Golestani approach to CBWFQ scheduling has been compared and they say under emergency conditions with traffic up to 10 times the normal load, and classes whose weights do not provide sufficient bandwidth to handle the traffic load [20].

3.2.4 LLQ (Low Latency Queue)

LLQ is combination of PQ and CBWFQ. The packets are marked with EF and assigned to single PQ, other packets are marked with AF and default DSCP values, and split the remaining bandwidth using CBWFQ with suitable weights [21]. LLQ reduces jitter in voice conversation and provides strict-priority queuing. This allows delay sensitive data such as voice to be dequeued and sent first. When voice packets enter the LLQ system a fixed bandwidth is allocated, data packets enter CBWFQ system where they are treated according to CBWFQ assigned weights [20].

In the absence of LLQ, CBWFQ provides just weighted queuing based on defined per class bandwidth with no strict-priority queue for real-time traffic. LLQ provide low latency propagation of packets.

Dekeris et al [23] conducted experiment associating both WFQ and LLQ to ensure QoS when the network is loaded with bursty video conferencing traffic." In LLQ, the only class able to provide low latency is a single priority queue. If the Video conference and the Voice traffic are merged together into one class, the bursty, large-packet Video stream would severely punish the small packet Voice traffic, so for Video traffic is used LLQ, and for Voice – weighed fair queue with highest priority" [23]. The paper says that WFQ with |LLQ discipline can be used for reducing the delay of packets with highest priority, when network is highly loaded. The main drawback of WFQ with LLQ is that delay can be reduced on high priority class, but, at same time, highest delay is also observed in AF13 class (voice) traffic. During high network loads, the packet losses of video can be reduced by using WFQ with LLQ, but the losses of packets with lower priority increases. Loss remains the same during low network loads. This has not influenced the QoS of voice because the loss is less than 1%.

According to this experiment the delay of the of packets from highest priority is reduced to two times by using WFQ and LLQ scheduling, but the delay of the lower priority packets has gone to 6ms (milliseconds). Packet loss for high priority video and voice traffic has acceptable losses up to 0.9 % and middle priority is up to 1%. This experiment assures quality parameters at high loads at network. This is not effective for low traffic load networks [23]. Figures 13 and 14 illustrate the results and comparison between the WFQ and WFQ with LLQ.

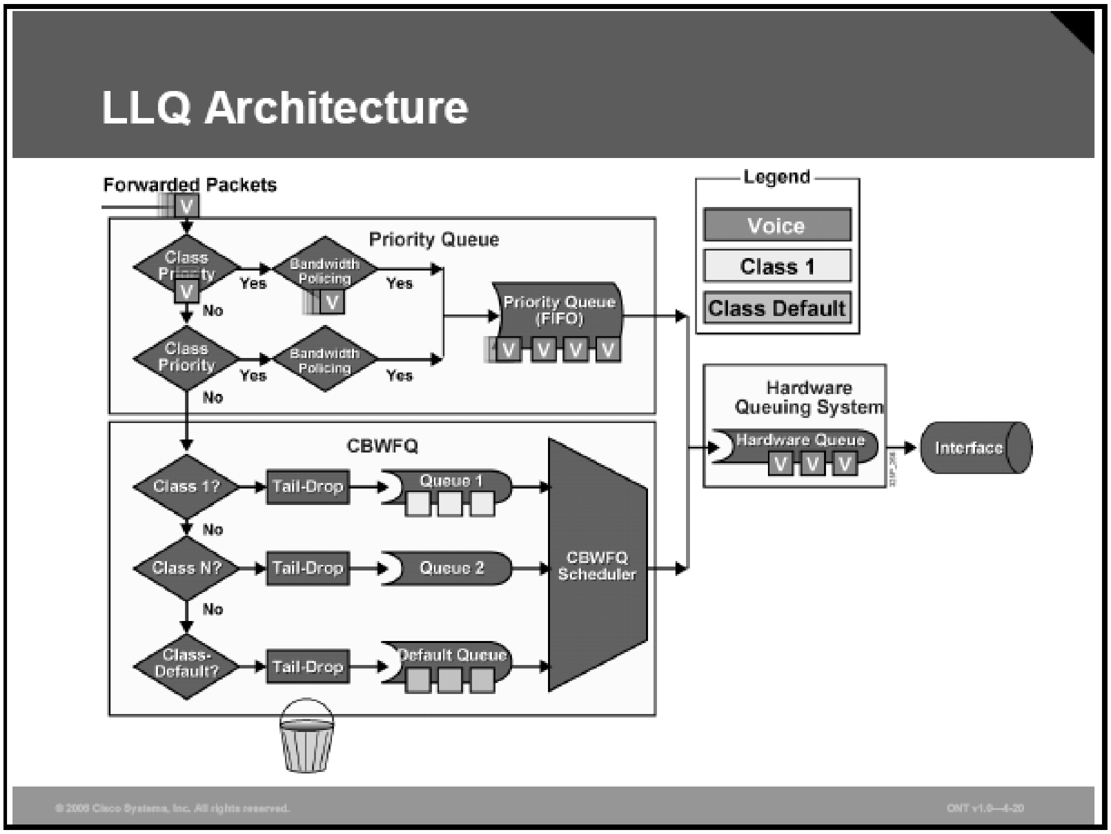

Figure 12 LLQ Architecture [24]

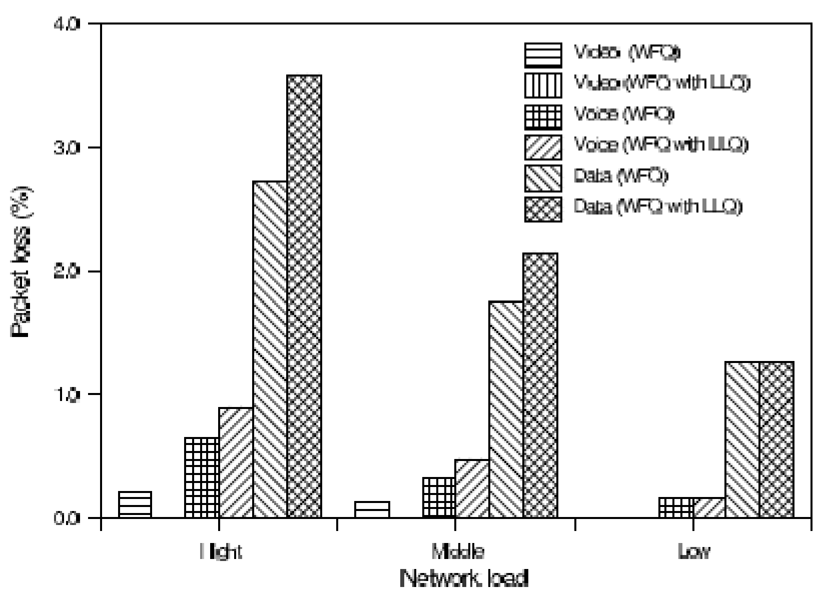

Figure 13 Losses of packets from all flows using WFQ and WFQ with LLQ scheduling queuing [23].

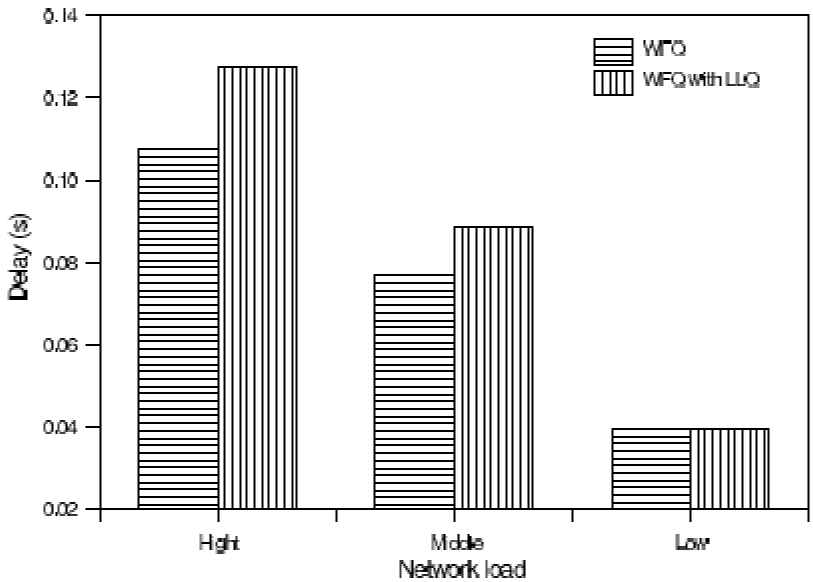

Figure 14 Delay of voice and video [23]

3.5 Conclusions

The aim of this chapter is to be familiar with the recent research on the QoS of VoIP network. The chapter has described various parameters of QoS in VoIP such as packet loss, jitter and delay. To overcome QoS parameters, different classification of traffic is discussed and presented above. Markings of voice packets on Layer 3 is been highlighted. Different methods of marking such as IPP and DSCP have been shown. Four different queuing techniques are discussed using recent research papers. The queuing techniques are PQ, WFQ, CBWFQ and LLQ. These research papers have provided new method of queuing technique on existing queuing methods.

The next chapter introduces the methodology, which includes a design of VoIP network.

4 Methodology

4.1 Introduction

In the beginning of this chapter, experimental design is discussed with providing the technical specification of each device and topology of the network. Experimental design is followed by next section called experimental methodology where experiments are described. In next chapter, implementation is revealed with configuration and experiments conducted using following queuing technique PQ, WFQ, CBWFQ and LLQ. Eventually end with evaluation, where the results are used for analysing the performance of each queuing technique.

4.2 Design

Network design is most important factor in terms of scalability of a network, performance and thereby providing good QoS. The components used in designing the network are:

- Cisco 3600 series Routers. Modular access router with LAN and WAN connections can be configured. These routers provide solutions to data, voice, video and multiprotocol data routing.
- Cisco catalyst 3550 switches. Multilayer switches, which provide high availability security and QoS. It has a range of Fast Ethernet and Gigabit Ethernet configurations. It can play as access layer switch and as a backbone switch.
- Avaya IP phones. Integrated with two full duplex 10/100 Base T switched Ethernet ports and PC pass through.
- PCs- Windows XP systems.
- File Server (FTP) - Windows 2003 server as a File server. FTP application is used for transfer of data from one node to another.

- TRIX Box IP-PBX (Linux Box). This is a complete application platform which has open source PBX (Asterisk)
- V.35 cable - ITU standard for high-speed synchronous data transfer. V.35 is used for most of the routers and DSUs that connect to T1 carriers.
- RJ (Registered Jack) 45 (100-Base-T Ethernet connection) cable. This an eight-wired connector used for connecting computers and IP phones onto a LAN, especially Ethernet.

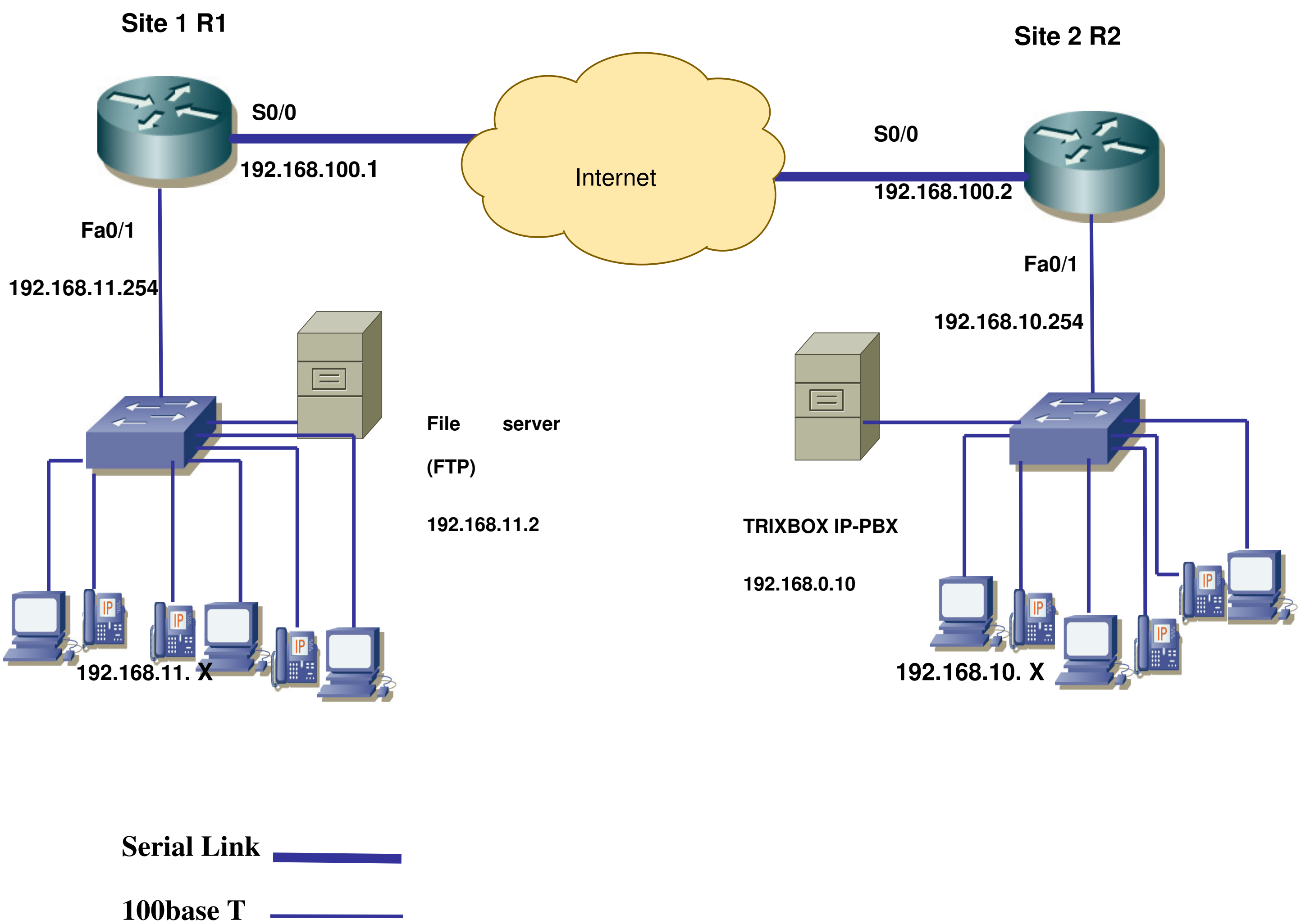

Figure 15 Network Topology

4.3 Experimental methodology

There were two main experiments, which aimed to perform, one without QoS and second are with QoS. Each experiment is performed using the same topology, where voice traffic and data traffic is transmitted simultaneously in the network to analyse the QoS parameters. The experiment description is followed in next sub-section.

4.3.1 Experiment 1

The experiment is performed without QoS implementation. VoIP network is implemented as seen in Figure 15 and observed jitter, delay and packet loss. The applications running on both the sites are as follows:

Table 3 Application details

Location	Application	No of Users	QoS	Link
Site1	Voice and FTP	6	None	500 Kbps
Site2	Voice and FTP	6	None	500 Kbps

4.3.2 Experiment 2

The experiment is performed by marking voice packets and implementing queuing techniques on the routers R1and R2. Here traffic is identified and grouped into a class and QoS is applied to these traffic classes. Queuing techniques like PQ, WFQ, CBWFQ and LLQ are configured and performance is observed. In this experiment, initially PQ is demonstrated and later WFQ, CBWFQ and LLQ respectively. Results have demonstrated the performance of each queuing technique.

4.4 Conclusion

The objective of this chapter is to show the design and hardware used for conducting experiment. This chapter describes the features of the hardware and software that is used in experiment. A short description of experiment is provided. In Chapter 5 of the experiment is described with topology and configuration.

5 Implementation

5.1 Introduction

This chapter deals with the implementation of the network topology and QoSconfiguration. There is concise discussion about the experiments conducted by two baselines but same topology. The first experiment is done without any QoS applied and the second experiment is sub divided into four experiments. In second experiment, there is QoS implementation on routers and voice traffic flow is marked by using DSCP. The four different queuing techniques are used and each queuing technique is used separately and performance is measured. The queuing techniques are PQ, WFQ, CBWFQ and LLQ.

5.2 Topology

Figure 16 is used as network topology for performing experiments and to achieve the objective of the project. Details of the topology as follows: Site1 consists of a router, switch, file server (FTP), PCs and IP phones. Router is connected to a switch where the router is the gateway for the LAN devices. PCs, IP phone and file server are connected to the switch. In Site2, it is very similar to the site1 but in the place of file server, there is an IP-PBX, which is used for linking phone lines.

Site1 and Site2 routers are connected by serial link. In each site there are six end users signed in at the same time. Using FTP application TCP/IP traffic is generated and using IP phones (where call are initiated using IP-PBX), voice traffic (UDP) is generated. Both the traffic voice and data are sent on the same link and QoS is measured.

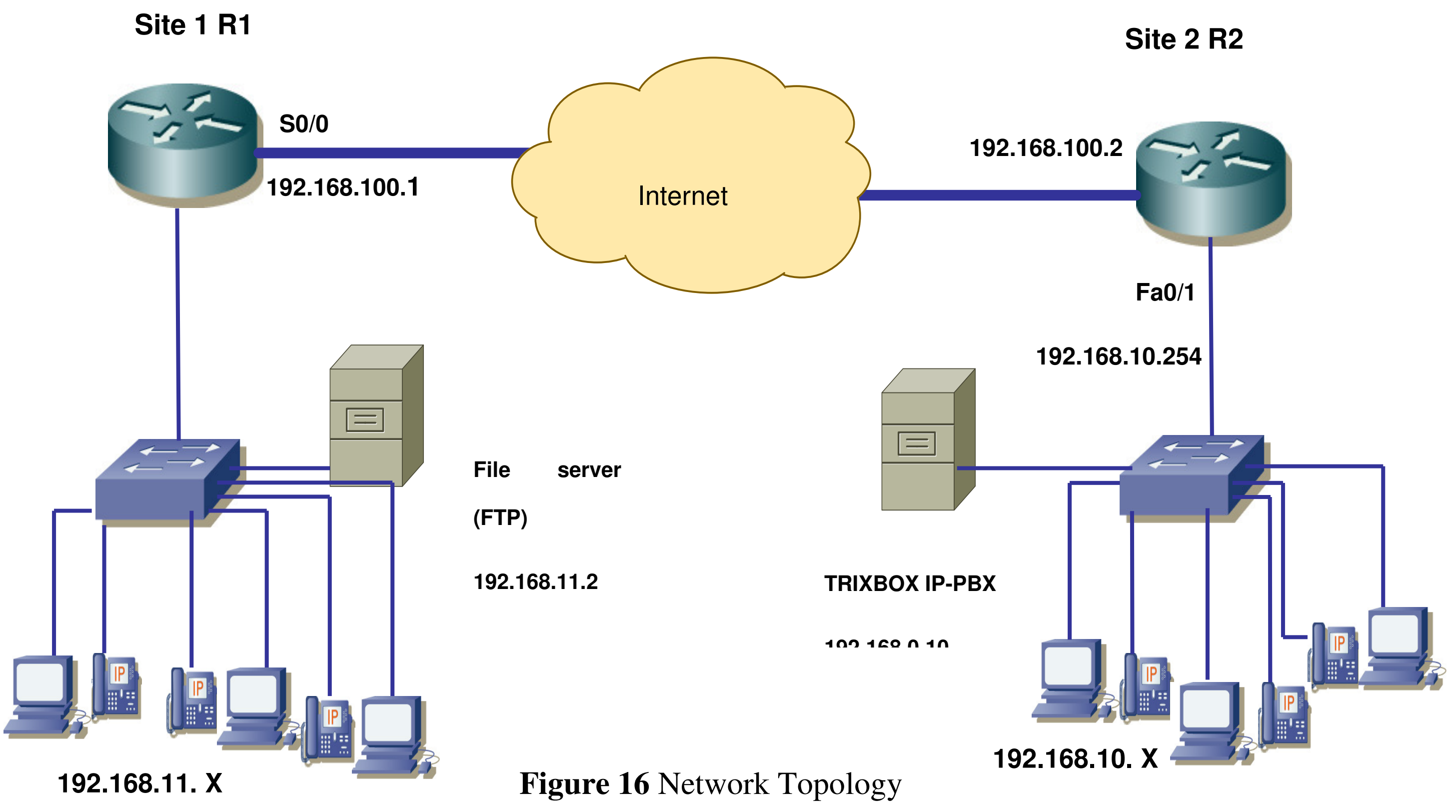

Figure 16 Network Topology

Table 4

Location	Device	Interface	IP Address
Site 1	Router(R1)	Serial 0/0	192.168.100.1
Site 1	Router(R1)	Fast Ethernet 0/1	192.168.11.254
Site 1	Switch		
Site 1	File Server		192.168.11.2
Site 1	PCs and IP phone		192.168.11.X
Site 2	Router(R2)	Serial 0/0	192.168.100.2
Site2	Router(R2)	Fast Ethernet 0/1	192.168.10.254
Site2	Switch		
Site2	TrixBox IP-PBX		192.168.0.10
Site2	PCs and IP phone		192.168.10.X

5.3 QoS Configuration

This section describes the configuration of all queueing techniques used in the experiment to analyse the QoS parameters such as jitter, delay and packet loss. The experiments are performed deploying different queueing techniques such as PQ, WFQ, CBWFQ and LLQ. Each queuing techniques were configured individually on the network and results were observed. All results are shown in the chapter 6.

5.2.1 PQ

To configure PQ, the following commands are used.

- Step1: the priority list is configured to establish the queueing priorities based on the protocol type.
- Step2: The maximum number of packets allowed in each of the queues is specified.
- Step3: Priority list is assigned to an interface and only one priority list can be assigned.
- Step4: When classifying a packet, router searches for the rules specified by priority-list commands for matching the protocol type.

When classifying the packet, the router searches for the rule.

```
R1 (config) #
R1 (config) #priority-list 1 queue-limit 10 20
R1 (config) #int s0/0
R1 (config-if) #priority-group 1
R1 (config) #priority-list 1 protocol ip medium tcp 21
R1 (config) #priority-list 1 protocol ip high
```

5.2.2 WFQ

To configure WFQ the following commands are used.

- Step1: Interface serial 0/0 is configured by assigning an IP address and description.
- Step2: Fair-queue is configured by specifying the congestion threshold value, dynamic conversation queues and reservable conversation queues.
- Step 3: On interface, queue length hold is specified for output queue.

```
R1 (config) # interface S0/0
R1 (config-if) # description 500kbps to R2
R1 (config-if) # ip address 192.168.100.1 255.255.255.252
R1 (config-if) # fair-queue 400 256 9
R1 (config-if) # hold-time 100 out
```

5.2.3 CBWFQ

To configure CBWFQ following commands are used.

- Step1: Access-lists are created for udp, tcp and ftp are created.
- Step2: Class-maps are defined to match the access group and to determine the class of the packets.
- Step3: Policy-map is configured to make up the service policy.
- Step4: Class name is specified to include in the service policy.
- Step5: Bandwidth is allocated in kbps to the assigned class.
- Step6: Queue-limit is configured which specifies the maximum number of packets which can be enqueued for the class. Here policy map uses tail drop.
- Step7: Default class is configured
- Step8: Fair-queue is defined, where number of dynamic queues are reserved for use by flow-based WFQ running on the default class.
- Step9: Service policy is enabled. This enables CBWFQ and attaches the service policy map to the output interface.

```
R1 (config) #access-list 100 permit udp any any range 16384 32767
R1 (config) #access-list 100 tcp any any eq 1720
R1 (config) #access-list 101 permit tcp any any eq 21
R1 (config) #class-map VOIP
R1 (config-cmap) #match access-group 100
R1 (config-cmap) #exit
R1 (config) # class-map DATA
R1 (config-cmap) # match access-group 101
R1 (config-cmap) # exit
R1 (config) # policy-map Aegis
R1 (config-pmap) # class VOIP
R1 (config-pmap-c) # bandwidth percent 60
R1 (config-pmap-c) # queue-limit 60
```

```
R1 (config-pmap-c) # exit
R1 (config-pmap) # class DATA
R1 (config-pmap-c) # bandwidth percent 40
R1 (config-pmap-c) # queue-limit 80
R1 (config-pmap-c) # exit
R1 (config-pmap) # class class-default
R1 (config-pmap-c) # fair-queue 16
R1 (config-pmap-c) # exit
R1 (config-pmap) # exit
R1 (config) # int s0/0
R1 (config-if) # service-policy output Aegis
```

5.2.4 LLQ

To configure LLQ the following commands are used.

- Step1: Step1: Access-lists are created for udp, tcp and ftp are created.
- Step2: Class-maps are defined to match the access group and to determine the class of the packets.
- Step3: Policy-map is configured to make up the service policy.
- Step4: Class name is specified to include in the service policy.
- Step5: Priority is set for the class VOIP
- Step6: Bandwidth is set for class DATA.
- Step7:Defalut class is created
- Step8: Fair-queue is defined, where number of dynamic queues are reserved for use by flow-based WFQ running on the default class.
- Step9: Service policy is enabled. This enables LLQ and attaches the service policy map to the output interface.

```
> en
# config t
R1 (config) # access-list 100 udp any any range 16384 32767
R1 (config) # access-list 100 tcp any any eq 1720
R1 (config) # access-list 101 tcp any any eq 21
R1 (config) # class-map VOIP
R1 (config-cmap) # match access-group 100
R1 (config-cmap) # exit
```

```
R1 (config) # class-map DATA
R1 (config-cmap) # match access-group 101
R1 (config-cmap) # exit
R1 (config) # policy-map Aegis
R1 (config-pmap) # class VOIP
R1 (config-pmap-c) # priority 50
R1 (config-pmap-c) # exit
R1 (config-pmap) # class DATA
R1 (config-pmap-c) # bandwidth 50
R1 (config-pmap-c) # exit
R1 (config-pmap) # class class-default
R1 (config-pmap-c) # fair-queue 16
R1 (config-pmap-c) # exit
R1 (config-pmap) # exit
R1 (config) # int s0/0
R1 (config-if) # service-policy output Aegis
```

5.4 Conclusion

The idea of this chapter is to show the implementation which is done according to the topology. This gives a brief knowledge about the experiment and configuration. Experiments conducted on two different approaches, but it uses same topology. The first experiment is done without any QoS implementation. Later using four different queuing techniques experiments are conducted. The results of experiments are analysed in Chapter 6.

6 Evaluation

6.1 Introduction

In this chapter, possible approaches are investigated to improvise the QoS in VoIP network. This chapter deals with result and analysis of the two experiments, which are discussed, in the chapter 5.1. First experiment is conducted without implementing QoS on the network and measured jitter, delay and packet loss. Later in the same section one more experiment is conducted which is subdivided into four, in each experiment different queuing is used such as PQ, WFQ, CBWFQ and LLQ. Each experiment is illustrated graphs, which measure jitter, delay and packet loss. This chapter is followed by conclusion, critical discussion and future work.

6.2 Result and Analysis

This section defines the experimental work.

6.2.1 Experiment 1

In this experiment, both sites consist of configuration as mentioned in chapter 5.1. The R1 and R2 serial connection is of speed 1.5 Mbps. Firstly the network is set up and calls are initiated and, at the same time FTP application are been used by the end-users.

Figure 17 shows the total bandwidth utilization by all the application. As the span of time increases the utilization increases due to increases in access of FTP server which is located in site1. In addition, call volume was also increased by 10 calls; due to this utilization is 0.75 Megabits per second (Mbps). From 10th minute, there is constant usage of bandwidth.

Figure 18 shows the jitter in voice reception. At eighth minute there 5 ms of jitter which does not affect the voice, this is negligible. This jitter is caused probably due to increase in logging of end-user to FTP server and downloading the files to their PCs. The data packets are larger than voice packets. The available bandwidth has patched up the issue and jitter decreases to zero.

Figure 19 illustrates the delay obtained during the experiment. The delay has remained 60ms throughout the experiment. This has not affected in anyway to the voice calls. The call quality was good.

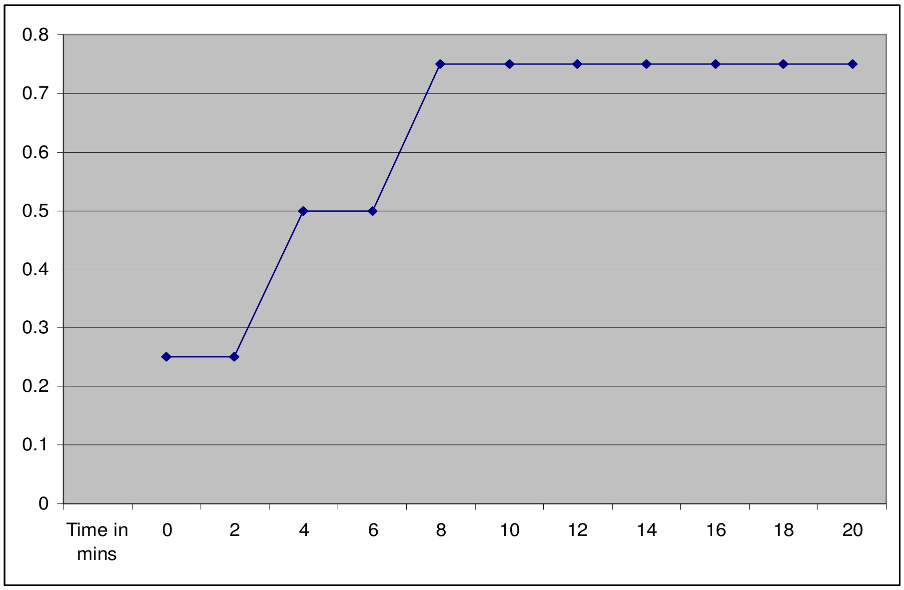

Figure 17 Bandwidth utilization (Mbps)

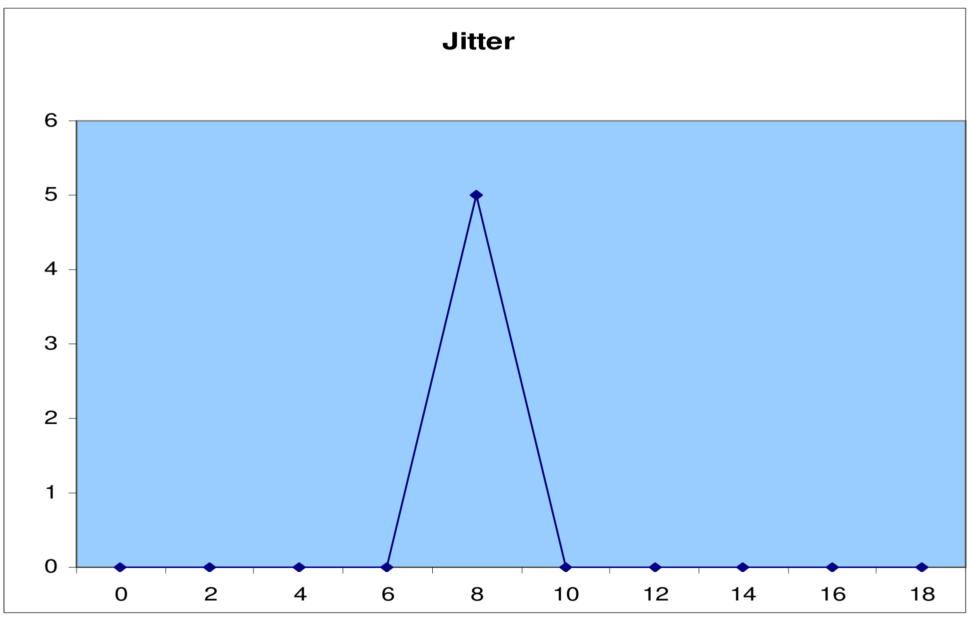

Figure 18 Jitter(ms)

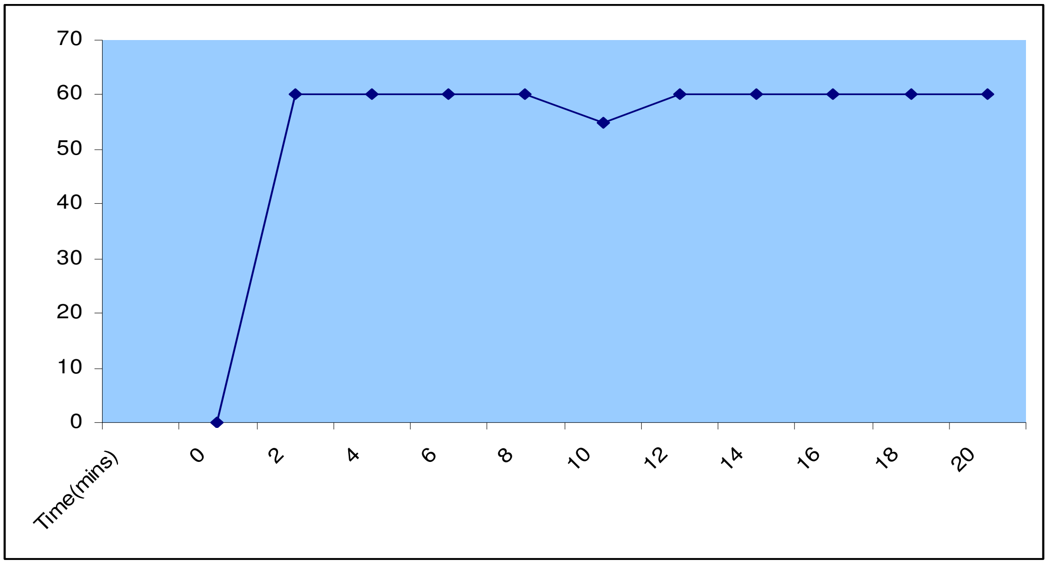

Figure 19 Delay (ms)

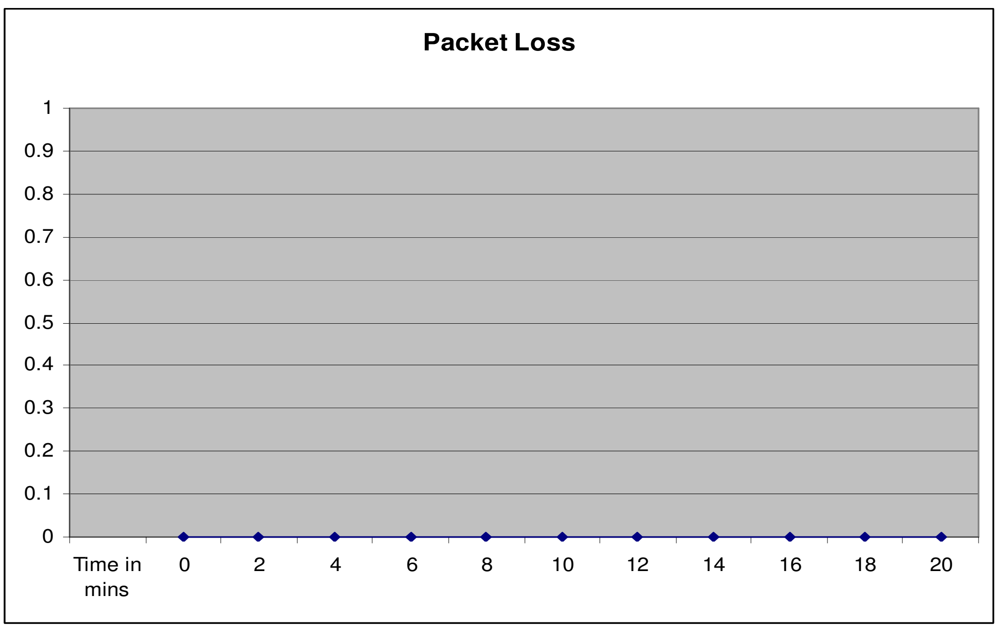

Figure 20 Packet Loss (ms)

Figure 20 shows that, no packet loss was found. End-users were accessing FTP and calling the other extensions, though there was no packet loss due to available bandwidth of 1.5 Mbps.

The results show that if there is sufficient bandwidth, the QoS parameters can be handled.

Later in this experiment to cause congestion, bandwidth between two sites is reduced to

500Kbps. In this environment, the experiment is repeated. The results are given in Figure 21.

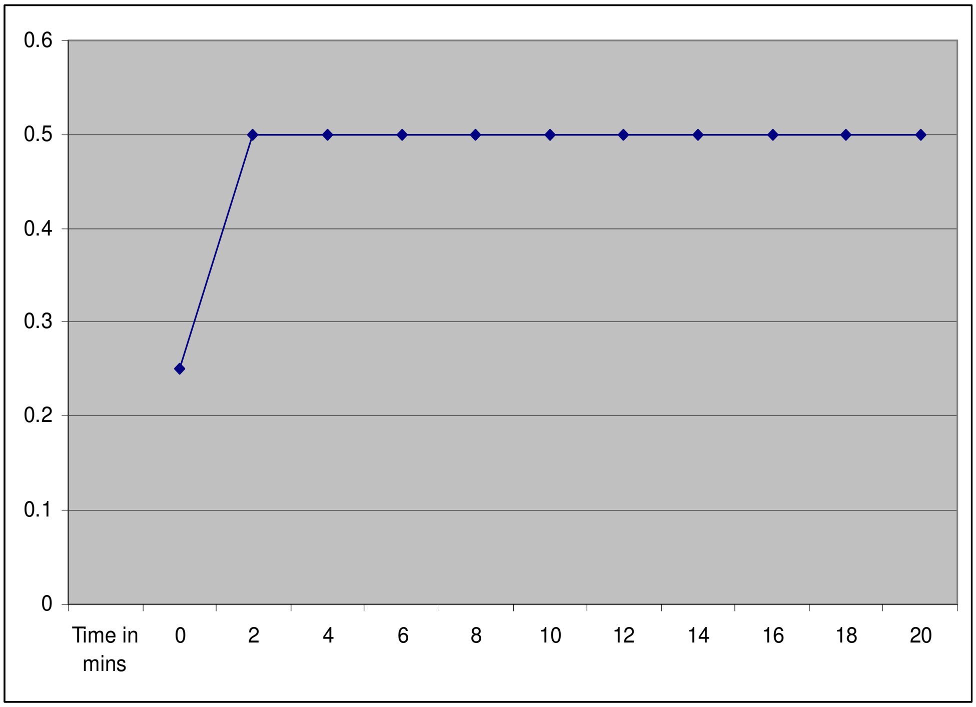

Figure 21 Bandwidth utilization

Figure 21 shows that the utilization has increased and remained 0.5 Mbps throughout the experiment. It has utilized the whole bandwidth, which is available. This affected voice calls; there were voice breakages during calls. This is due to reduction in available bandwidth and lack of QoS in the network. All traffic flow is treated in the equally, voice traffic being UDP packets they cannot be retrieved. This has caused voice distortion. The best-effort service is not recommended for networks, where voice and data are converged into single IP network.

Figure 22 demonstrates the jitter of voice reception. All agents were logged in at the same time and accessing the FTP Server. The calls which were in progress started to suffer from voice distortion. The cause same as above mentioned for bandwidth utilization. The data packets are larger than voice packets. By default on Cisco devices, the queuing is FIFO (First In, First Out).

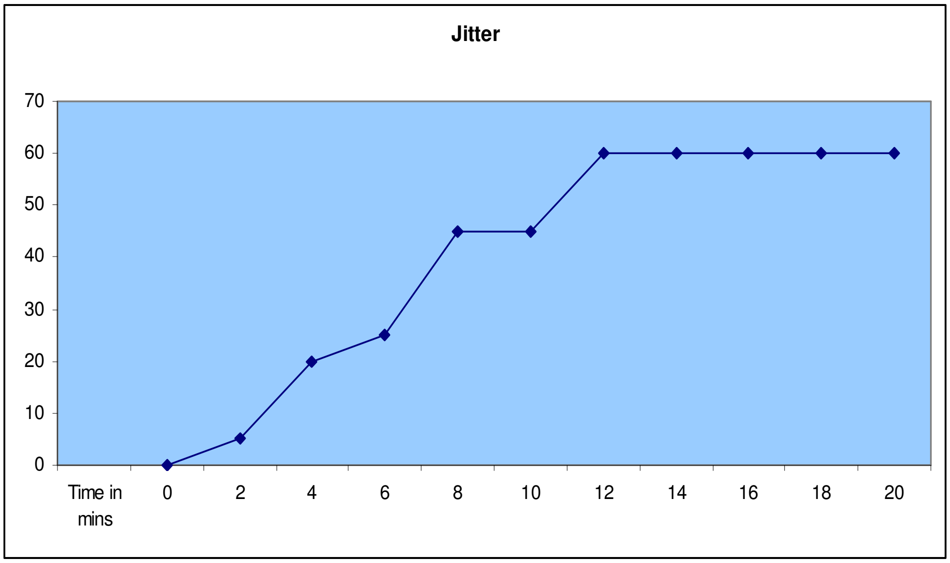

Figure 22 Jitter (ms)

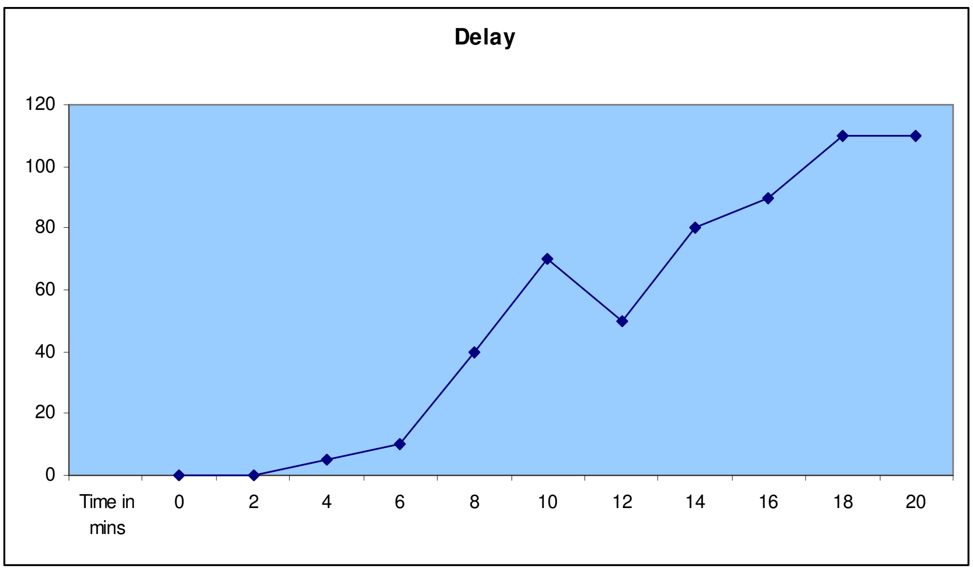

Figure 23 Delay (ms)

Figure 23 shows delay of voice traffic flow. This is due the same reason as above-mentioned jitter. Here the delay is not constant. This was affecting all voice calls in the network on both the sites.

Figure 24, the packet loss has increased. It starts with 3% and remains to 40% at 20th minute, which is huge loss. The voice distortions were high across the network. This experiment illustrates that lack of QoS and classification of voice traffic flow causes lot of jitter, delay and packet loss. This results in voice distortion and slows down the application which is been used by the agents or end-users.

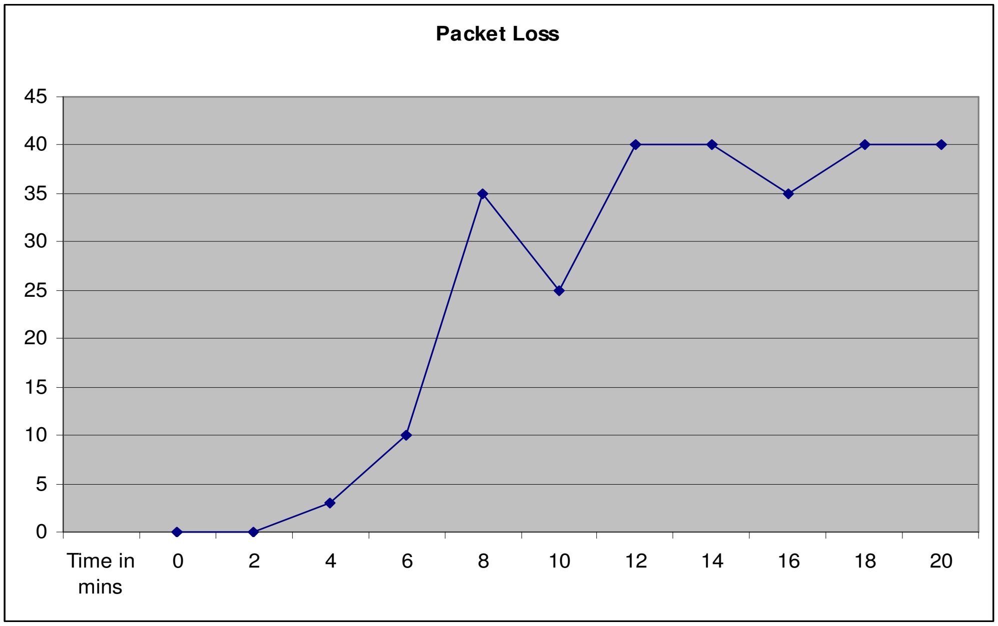

Figure 24 Packet Loss

6.2.2 Experiment 2

Figure 25 illustrates the jitter obtained during voice reception. The voice packets are tagged with DSCP code EF before they exit the WAN interface. The PQ is configured on each router. Voice packets are on high priority. This configured by creating a priority list and specifying the protocol (UDP) and mapping it to the access-list, which specify the udp traffic. According to Figure 25, jitter obtained is acceptable which did not affect voice traffic flow. However, there was lot of influence on FTP application, where FTP becomes slow. This is the result of configuring PQ, as voice is placed on high priority queue and the voice traffic stream is transmitted before any other traffic. Since data traffic is not defined to any priorities, the traffic is placed into normal queue. This can affect the performance of the application considerably, when voice traffic flow is high. PQ is not reliable under pressurised environment.

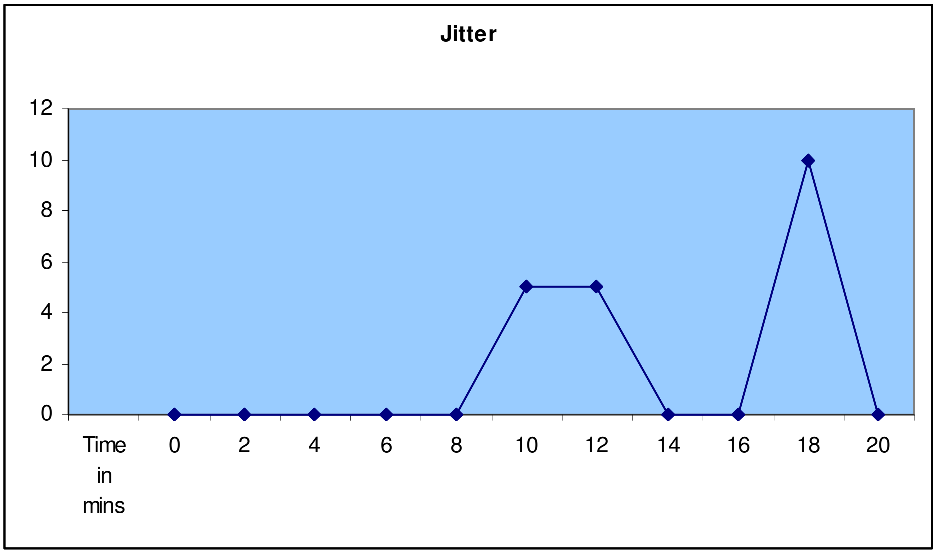

Figure 25 Jitter (ms) for PQ

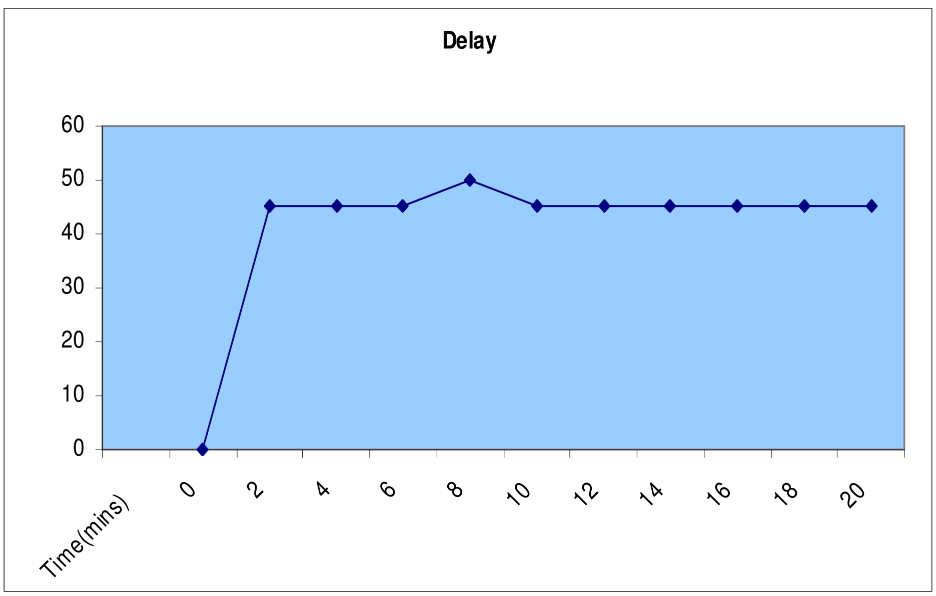

Figure 26 Delay (ms) for PQ

Figure 26 shows obtained delay at voice reception. The delay obtained is acceptable for voice traffic. As the voice traffic flow is placed on high priority, this has affected on FTP application.

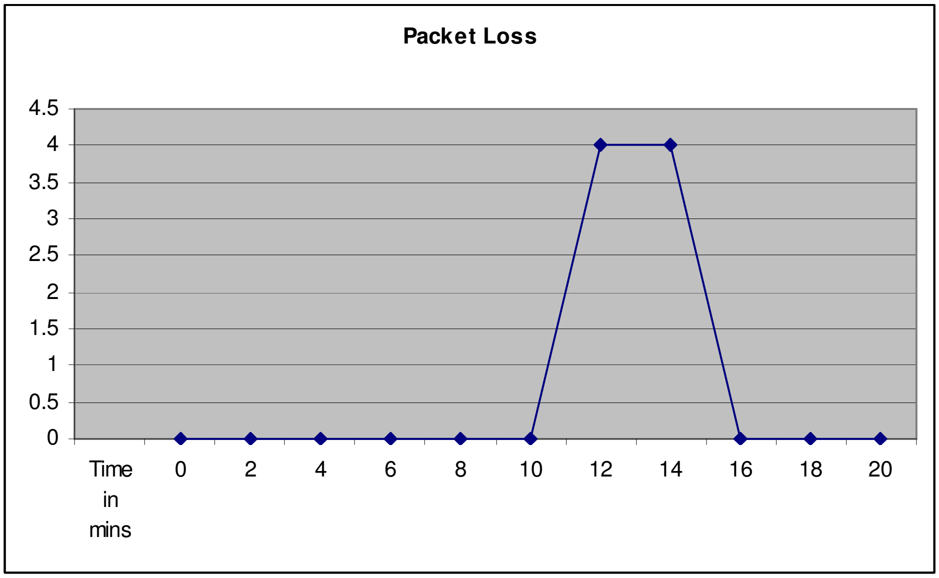

Figure 27 Packet Loss for PQ (%)

Figure 27 illustrates the packet loss obtained on voice reception. This can be considered as negligible. This has not affected voice traffic flow.

Figure 28 has shown the jitter received during the experiment performed. Here WFQ is used as queuing technique. WFQ is considered fair because it assigns the same weight to all traffic flows over high volume flows. WFQ is configured on serial interface of R1 and R2. In this experiment, the thresholds are configured to default, where high bandwidth conversations are dropped. As we know that data packets (FTP) are larger than voice packets. As all agents login to the systems and access FTP server, it utilises more bandwidth than voice packets. High bandwidth conversation is been dropped by WFQ to prioritise lower ones. The jitter obtained has not affected voice packets. Figure 29 gives the information about delay reception throughout experiment. This has not affected voice traffic stream.

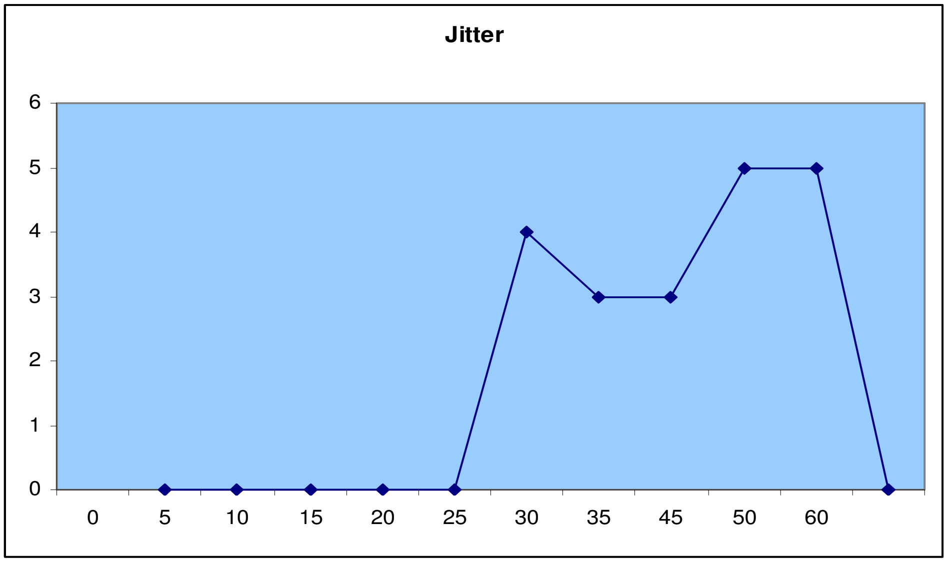

Figure 28 Jitter (ms) for WFQ

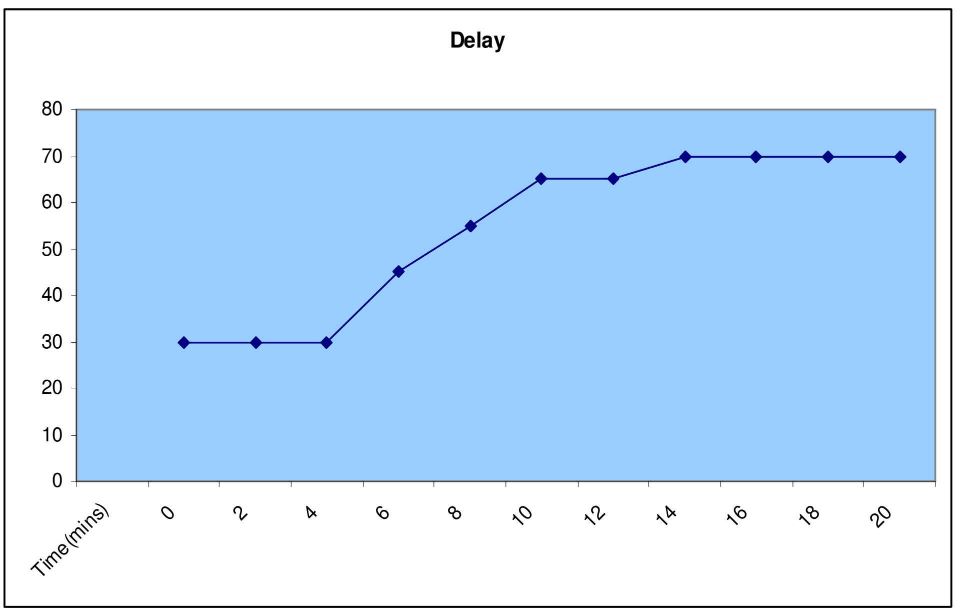

Figure 29 Delay (ms) for WFQ

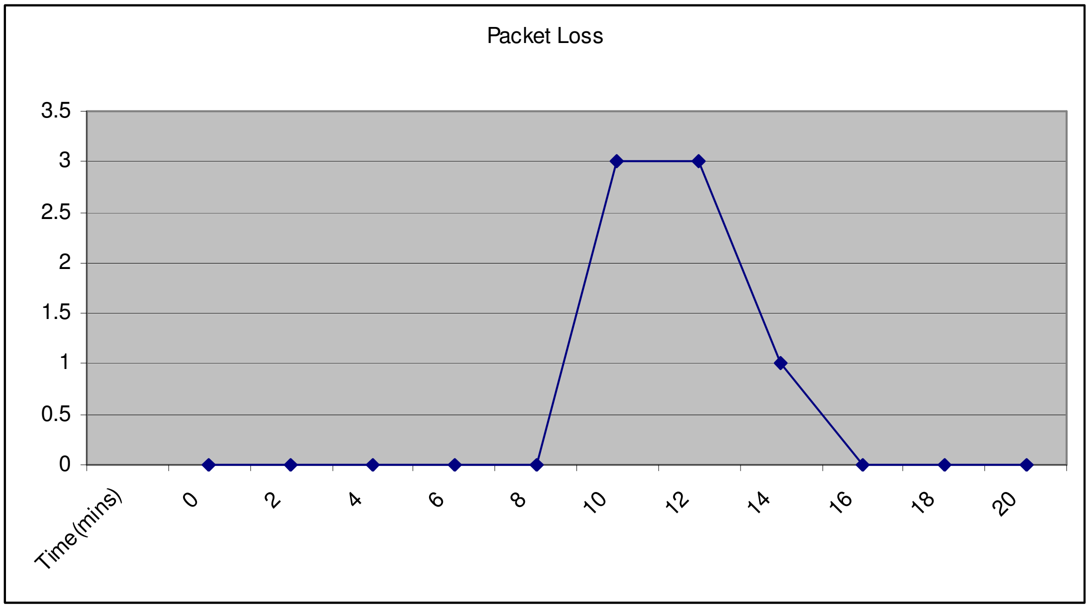

Figure 30 Packet Loss for WFQ (percentage)

Figure 30 represents the packet loss in voice traffic stream. At 10th minute, there is a hike in packet loss to 3% and 5 minutes later, it is dropped to 1 %. This is minor loss, which does not affect voice traffic flow across the network. There is no packet loss from 17th minute. This is acceptable in VoIP network. There was one more thing observed during experiment, FTP application was frozen for few minutes. Later it was working fine.

Figure 31 shows the jitter received on voice reception. Here CBWFQ is been configure on both the routers R1and R2 and voice traffic flow is marked with EF. At 20th minute jitter has risen to 7.5 ms, after 5 minutes it has decreased to 4 ms and at 45th minute it hiked to 10 ms. In Figure 32, delay can be observed that it has risen from 50 ms to 70 ms. This has not affected voice traffic stream. Figure 33 demonstrates the packet loss occurred throughout the experiment. The packet loss is tiny loss, which can be neglected.

Figure 34 represents the jitter on voice reception. Here LLQ is been used as queuing technique and marked with EF. This queuing technique has performed extremely well, which resulted in very low delay. Using this queuing mechanism, it did not affect any of the other application, which was running. When compared to other queuing techniques LLQ can be the efficient queuing mechanism. Figure 35 has shown the incredible performance of LLQ where delay is 60 ms.

This explains the call performance. Finally, Figure 36 is representing the 0% of packet loss by using LLQ.

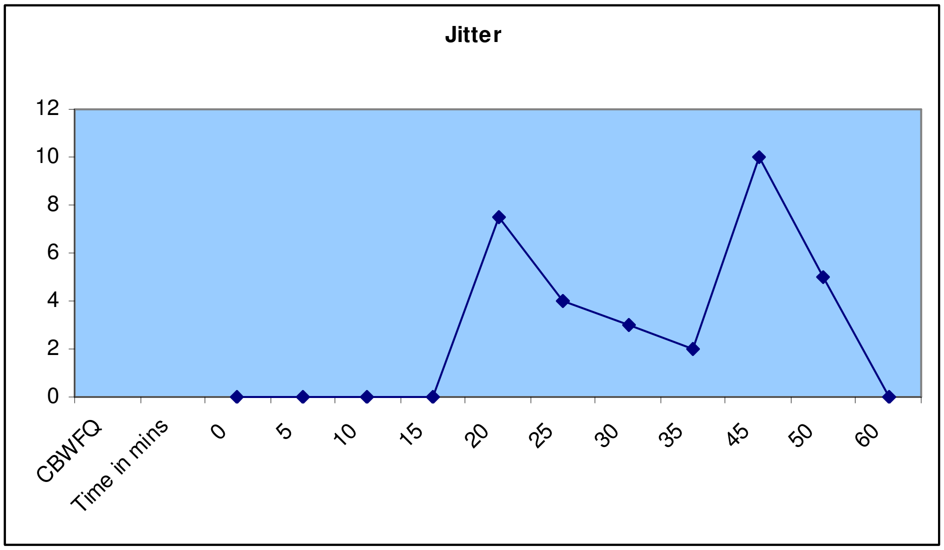

Figure 31 Jitter (ms) for CBWFQ

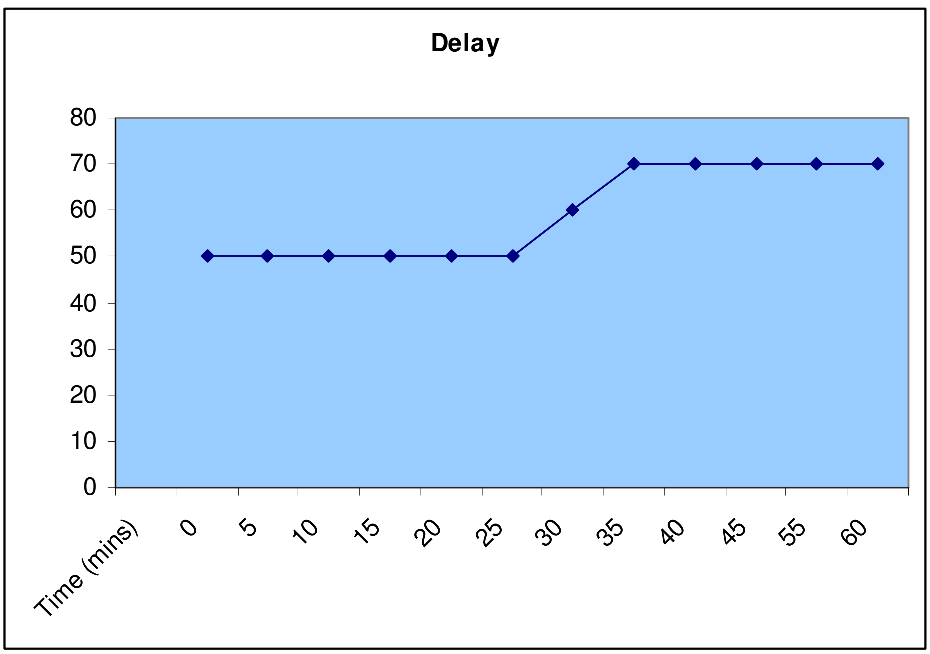

Figure 32 Delay (ms) for CBWFQ

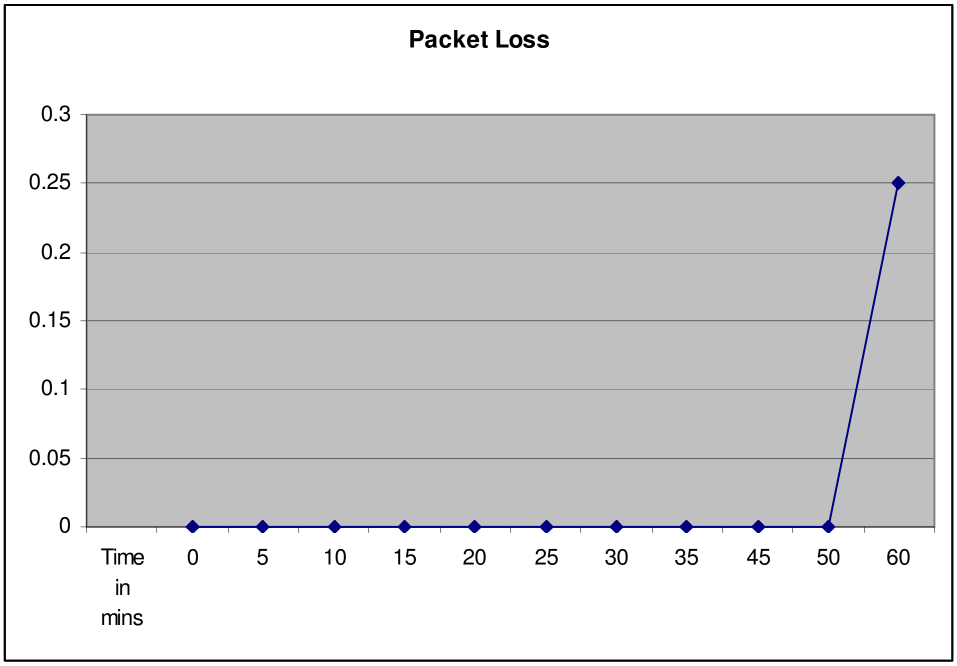

Figure 33 Packet Loss (%) for CBWFQ

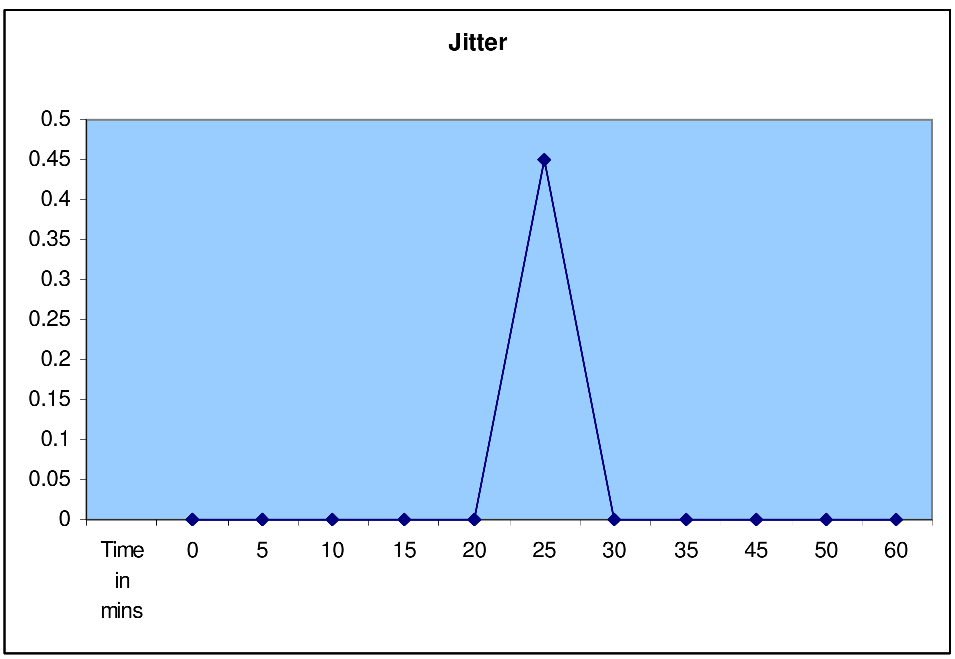

Figure 34 Jitter (ms) for LLQ

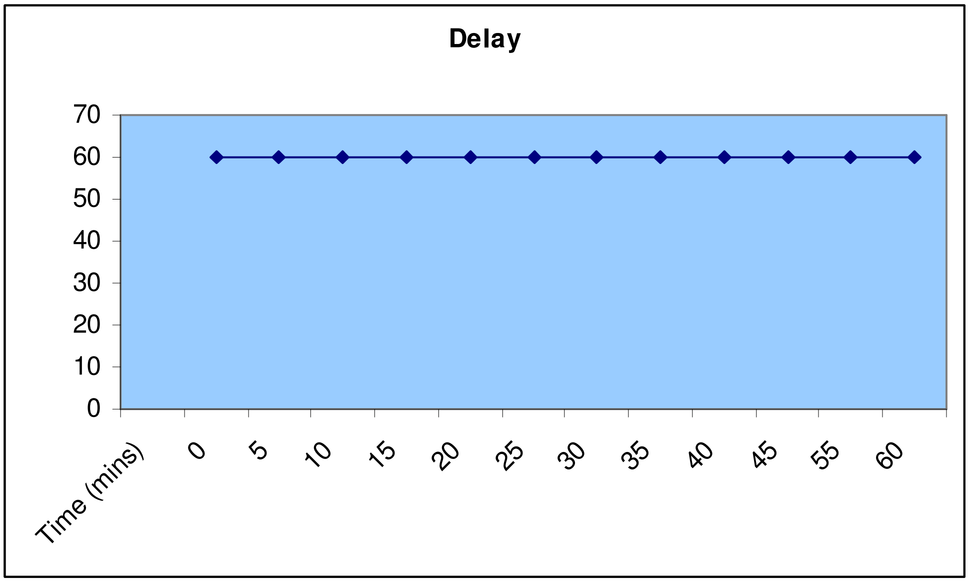

Figure 35 Delay (ms) for LLQ

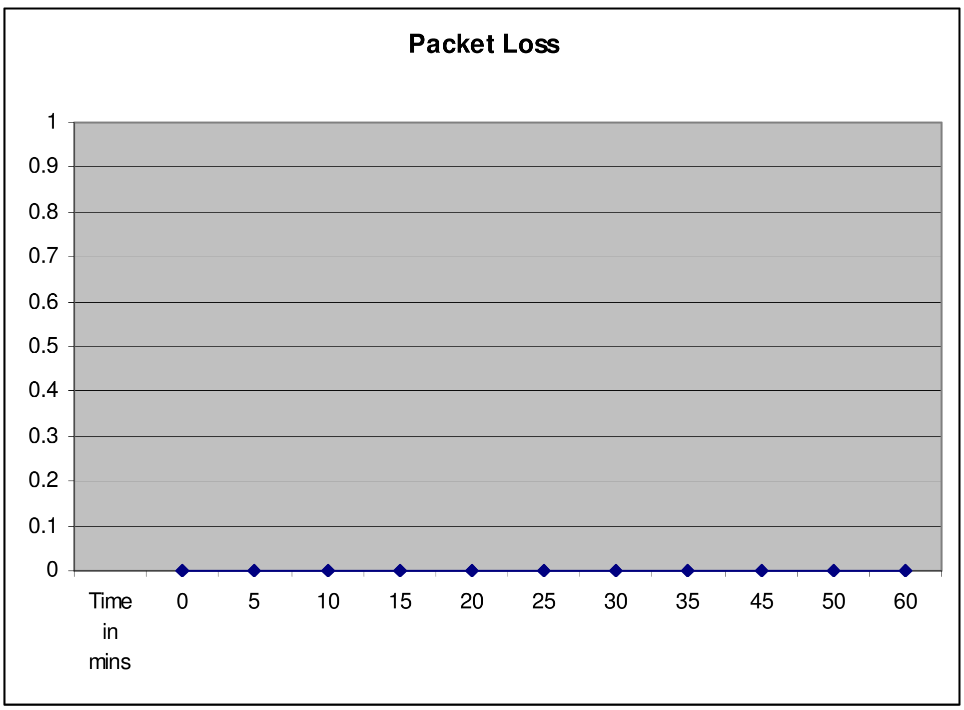

Figure 36(ms) Packet Loss for LLQ

6.3 Conclusions

The evaluations revealed effect of different queuing mechanism on QoS of voice quality. The increase in delay, jitter and packet loss affect the quality of voice. This chapter deals with result and analysis of the two experiments. This section includes two experiment performed to achieve the objective of the project. The first experiment was with out QoS on the network, which gave good results. This was because of available bandwidth of 1.5Mbps. The same experiment was conducted but the available bandwidth was reduced to create congestion, it resulted in voice distortion and lot of packet loss. The delay and jitter was high. This comes under same experiment.

This is followed by second experiment where voice packets are marked and queuing is been enabled on routers. Each queuing is been done separately with same marking called EF.

As the results of each queuing technique indicates their efficiency in queuing and delivering the good QoS for voice. PQ is good when the traffic load is very high there is 4% of packet loss. This has not affected the voice but the data transfers to the FTP server had become slow.WFQ was better than PQ WFQ is considered fair because it assigns the same weight to all traffic flows over high volume flows. In this experiment, the thresholds are been configured to default, where high bandwidth conversations are dropped. This resulted in low delay and jitter, which did not affect voice traffic flow. LLQ is been used as queuing technique and marked with EF. This queuing technique has performed extremely well, which resulted in very low delay. Using this queuing mechanism, it did not affect any of the other application, which was running. When compared to other queuing techniques LLQ can be the efficient queuing mechanism.

7 Conclusion

7.1 Overall conclusions

To achieve reliable, high-quality voice over an IP network, which is designed for data communication is an engineering challenge. Factors involved in designing good quality VoIP system include the choice of marking and a perfect queuing technique. There are other factors are also involved such as codec and call signalling protocol.

VoIP is fast developing technology, most of the ISPs or broad band service providers are planning to start VoIP service. VoIP is time sensitive application and requires real-time support for its QoS requirements. Traditional internet that uses best-effort mechanism ahs failed to support the QoS requirement of VoIP. Differentiated service (DiffServ) is a scheme designed to support QoS requirements in a scalable manner. PHB, EF and AF are designed to provide low loss and low latency, which is major requirement for real-time applications.

In this thesis, experiments are been performed in real-time network in a company called Aegis BPO. Aegis is a contact centre where they are using VoIP in their business. Experiments were conducted in real network based on the design proposed, to evaluate the QoS of real-time network.

At this stage, it can be concluded as QoS may be used for a smooth running of VoIP by using packet classification and queuing mechanism. Generating traffic of both voice and data is difficult in real-time network. Experiment 1 was performed without QoS on network , FTP and voice was working fine with 1.5 Mbps link speed, later as FTP traffic was increased only on one call there was distortion. In these experiments, there were six agents on each site. To create congestion, bandwidth was reduced to 500Kbps. Here results were not promising the quality calls. This generated lot of jitters and packet loss. In the next experiment, QoS was implemented on the routers. Queueing techniques were deployed, such as PQ, WFQ. CBWFQ and LLQ. Each queuing technique has given different results. The delay and jitter were acceptable, packet loss was not an issue but in PQ experiment packet loss reaches to 4 % which affected the voice at that particular time. This will affect only on one call and on the same extension. As QoS is deployed on the routers, the FTP application started losing its pace; the time taken for authenticating was more. Agents were not able to upload or download files easily. File uploading and downloading was time consuming.

For good voice quality, many factors involve such as codec, network design and perfect cabling. We cannot implement all kinds of queuing on all kinds of network. It again depends on the network design and the applications running on the network. It cannot be always voice prioritised, because there are few applications, which will be affected. In real-time QoS works in a different way, it does not work as Cisco recommends.

According to above experiments performed, LLQ has better performed than any other queuing mechanism. Here voice packets are marked using EF for voice which given a very good result with LLQ. When LLQ was used, it kept an acceptable packet loss less than 1% the delay of the voice packets was nearly nil. LLQ can be used for the QoS parameters assurance in VoIP networks.

7.2 Critical discussion

Various solutions have been suggested for smooth operation of voice over data network. One among them is by Adonmkus, Budnikas and Dekeris states that the solution was using a combination of two queuing techniques i.e. WFQ with LLQ scheduling disciplines[23].

They conducted an experiment associating both WFQ and LLQ to ensure QoS when the network is loaded with bursty video conferencing traffic. The paper states: When network is highly loaded, WFQ with |LLQ disciplines are used for reducing the delay of packets with highest priority. The main drawback of WFQ with LLQ is that delay can be reduced on high priority class but at same time, highest delay is also observed in AF13 class (voice) traffic. During high network loads, the packet loss of video is reduced by using WFQ with LLQ, but the loss of packets with lower priority increases. During the low network loads, the packet loss remains the same, which does not have any influence on the QoS of voice.

In spite the experiment has been performed on a test bed, the result of the same is very impressive. The outcome would be still more informative if the experiment is carried out in a real time traffic flow of a complex network.

7.3 Future work

There are more possibilities for future research in this area. Research could be done on automated DiffServ so that it can deployed in anywhere in the internet. Since this project was performed in real-time this was just with one TCP traffic generating application. If experiments were performed with two or more TCP application, the results would have been more interesting.

The research could be done with two different kinds of real-time application and TCP applications. The implementation of queuing and traffic classification would be more complex.

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APPENDIX 1

NAPIER UNIVERSITY

SCHOOL OF COMPUTING

RESEARCH PROPOSAL

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BSc or MSc: MSc Advanced Networking

Full-time/Part-time: Full-time

Telephone no: 07961057744

Project outline details

Title of the proposed project:

Analysis of QoS in Real Time VoIP Network

Brief description of the research area – background

Voice over IP (VoIP) converts audio signals into digital data, which can be transmitted over Internet. This is a revolutionary technology, which is replacing the phone system. VoIP has gained a lot of attention from number of organisations and is growing steadily. Traditional PSTN are resource dedicated, where IP network are resource shared .Therefore VoIP is cost efficient

In real-time network, voice and data can be transmitted simultaneously on the same link. In a VoIP network four things has to be considered: low latency; low jitter; low packet loss and available bandwidth. To withstand to such needs certain level of QoS mechanism is required. However, In IP network congestion is inevitable which directly affects voice packets. So the IP network must be enhanced with by guaranteeing mechanism in order to ensure the good voice quality.

QoS is typically defined on different models: 1) Best-Effort: is a default service, which exists on Cisco routers. Here queuing scheme make no allowances for the special needs of voice traffic.2) Differentiated Service is a method for specifying and controlling traffic by class so that certain types of traffic get priority service. (3) Integrated Service provides the packet flow with a quality of service closely approximating the QoS. That the same flow would receive from an unloaded network, but uses the capacity control to ensure that this service is received even when the network is overloaded.

In a mixed network, voice packets experience long queuing delays as they are trapped behind several large data packets in the queue. So Voice and Data packets are marked and identified by different TOS values and placed in different queues. Generally, QoS is a collection of technologies. QoS features provide better and more predicable network service by following methods: 1) dedicated bandwidth 2) Improving loss characteristics 3) Avoiding and managing network congestion 4) shaping network traffic 5) Setting traffic priorities across the network.

In the light of the above observations, this project will consider identification of different traffic patterns, with the aim of investigating the impact of queuing and tagging on the quality of voice. In addition, the project will conduct intensive experimental study and analysis of packet performance after implementing the QoS in the network, considering different queuing methods.

Project outline for the work that you propose to complete

The idea for this research arose from:

The need for VoIP is increasing and many organisations need to understand the effect and applying different queuing techniques and tagging in traffic flows.

The aims of the project are as follows: This project aims to identify and analyse different queuing mechanism and tagging for the traffic flows.

The main research questions that this work will address include:

- Which is the most efficient Queuing method for VoIP in different network?
- Can network be tagged to identify different traffic streams in the network and the flows?
- How to implement QoS in the mixed traffic network?
- How to improve the quality of voice in a network?

The software development/design work/other deliverable of the project will be:

- Identification of different traffic streams in the network**.**
- Analysis of the Quality of voice after implementing the QoS.
- Range of experiments for different queuing methods
- Analysis of packet behaviour after implanting the QoS in the network.

The project will involve the following research/field work/experimentation/evaluation: Analysis of different queuing methods by conducting range of experiments. Voice and data packets are marked and prioritised. By doing this we can monitor the packet loss jitter and bandwidth utilisation.

This work will require the use of specialist software: QoS device Manager, NMS tools TCP replay OPNET, NS-2 and hping.

This work will require the use of specialist hardware: Routers, Switches, Call manager and IP phones and PSTN emulator.

Supervision :

The following person has agreed to supervise this work:

Prof. Bill Buchanan