PERFORMANCE ANALYSIS OF VOIP NETWORK BETWEEN KK3 & KK5 UNIVERSITI MALAYSIA PAHANG USING OPNET SIMULATOR

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JUDUL **PERFORMANCE ANALYSIS OF VOIP NETWORK BETWEEN KK3 AND KK5 UNIVERSITI MALAYSIA PAHANG USING OPNET SIMULATOR**

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DEDICATION

This project is dedicated to my beloved parents,
Mr. Choo Yin Yeow and **Ms. Chua Siew Lian**,
who have never failed to give me moral support,
giving all I need during the time I do my research.

To my supervisor,
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To all my siblings and my aunt,
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ABSTRACT

Voice over Internet Protocol (VoIP) is rapidly growing and the market is booming because of its economical benefits and innovative phone services. This research is done by simulating the current network topology of KK3 to KK5 Universiti Malaysia Pahang in OPNET Modeler 14.5 to generate the results to be analyzed. The main objective of this research is to determine how many concurrent VoIP users can the current network between KK3 and KK5 supported at the same time. Besides that, how many VoIP users can be supported when HTTP traffic and video streaming is added in the network. The simulation results will be analyzed in this research is jitter, packet end-to-end delay, and MOS. In the simulation, two scenarios will be created which is scenario with only VoIP traffic and scenario with busy traffic which is HTTP and Video streaming traffic. The number of concurrent VoIP users started from 100 users and added 50 users each time in both scenarios until the value reached the threshold. The simulation result had shown the maximum concurrent VoIP user for scenario with only VoIP traffic is 250 users. On the other hand, the maximum number of concurrent VoIP user for scenario with HTTP and Video Streaming traffic is 100 users.

ABSTRAK

Voice over Internet Protocol (VoIP) berkembang pesat dan pasaran juga dapat dibangunkan dengan cepat kerana faedah ekonomi dan perkhidmatan telefon inovatif. Kajian ini dijalankann dengan mensimulasikan topologi rangkaian KK3 ke KK5 Universiti Malaysia Pahang dalam OPNET modeler 14.5 supaya dapat menjana hasil untuk dianalisis. Objektif kajian ini adalah untuk menentukan jumlah pengguna VoIP boleh disokong pada masa yang sama dalam rangkaian semasa antara KK3 dan KK5. Selain itu, jumlah pengguna VoIP juga telah ditentukan apabila trafik HTTP dan video streaming trafik ditambah dalam rangkaian tersebut. Keputusan simulasi akan menjana dan menganalisis dalam kajian ini ialah ketar, paket hujung ke akhir kelewatan dan MOS. Dalam simulasi ini terdapat dua sinario. Sinario pertama adalah sinario dengan VoIP trafik manakala sinario kedua adalah sinario yang mengandungi VoIP trafik dengan HTTP trafik dan Video Streaming trafik. Dalam kedua-dua sinario, pengguna VoIP akan bermula dari 100 pengguna dan akan ditambah 50 pengguna setiap kali. Hasil simulasi tersebut menunjukkan maksimum pengguna VoIP serentak untuk scenario tanpa trafik ialah 250. Manakala bagi bilangan maksimum pengguna VoIP serentak untuk scenario dengan trafik ialah 100.

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LIST OF ABBREVIATIONS

ATA	Analog Telephone Adapter
ATM	Asynchronous Transport Mode
CODEC	COder/ DE Coder
FTP	File Transfer Protocol
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
IPDV	Instantaneous Packet Delay
ISP	Internet Service Provider
ITU-T	International Telecommunications Union
KK	Kolej Kediaman
LAN	Local Area Network
METRO-E	Metropolitan Area Ethernet
MCU	Multipoint Control Units
MGCP	Media Gateway Control Protocol
MOS	Mean Opinion Score
MPLS	Multi Protocol Label Switching
PSTN	Public Switched Telephone Network
PTMK	Pusat Teknologi Maklumat dan Komunikasi
RTP	Real-Time Transport Protocol
SDP	Session Description Protocol

SIP	Session Initiation Protocol
UAC	User Agent Client
UAS	User Agent Server
TIA	Telecommunication Industry Association
VOIP	Voice Over Internet Protocol
VPN	Virtual Private Network
WLAN	Wireless Local Area Network

CHAPTER 1

INTRODUCTION

1.0 INTRODUCTION

University Malaysia Pahang (UMP), one of the technical universities in Malaysia, with the vision to be the world-class technological university. Its mission is to provide high quality education, research and services in engineering and technology in a culture of creativity and innovation. UMP consists of eight faculties, which are Faculty of Chemical & Natural Resources Engineering, Faculty of Civil Engineering & Earth Resources, Faculty of Computer Systems & Software Engineering, Faculty of Electrical & Electronics Engineering, Faculty of Industrial Sciences & Technology, Faculty of Manufacturing Engineering, Faculty of Mechanical Engineering and also Faculty of Technology. Besides that, UMP provides hostel to the students here. It consists of 5 hostels which are KK1, KK2 KK3, KK4 and KK5. Most of the hostels are located in Gambang campus except KK5 is located in Pekan campus.

In this new era of globalization, voice over internet protocol (VoIP) usage grows rapidly due to its cost effective, more functionality over the traditional telephone network and its compatibility with Public Switched Telephone Network (PSTN). PSTN also known as Plain Old Telephone Service which refers to the international telephone system based on copper wires carrying analog voice data. However, PSTN cost us a lot when we make phone calls to others, especially when we call between the countries. Sometimes, it charged double if we used roaming. That is the main reasons why VoIP deployment is gaining popularity there days. According to Infonetics Research, a telecom market research firm, VoIP will experience revenue growth of $76.1 billion in year 2015[1].

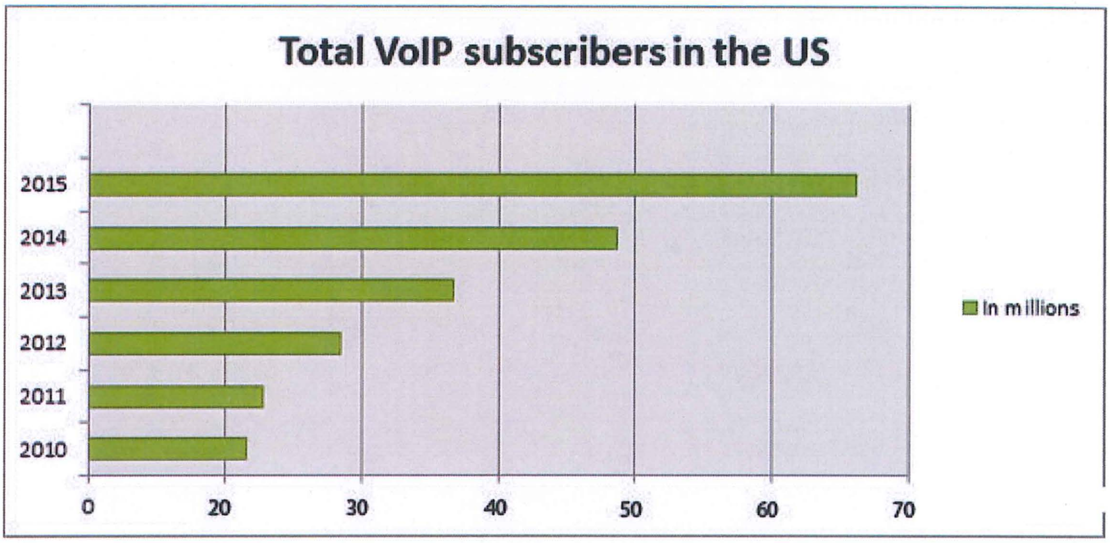


Figure 1.1: Expected total VoIP subscribers in the US for 2015.

"*Voice over Internet Protocol is quietly remarking the telephone system worldwide. It is one of the venerable network's biggest overhauls in decades – but not its last by a long way*" The Economist, March 2001. So, what is VoIP? VoIP refer to IP Telephony or Internet Telephony. It is a high modern technology that allows user to generate phone calls over an internet protocol network using packet-switching instead of the traditional PSTN which uses circuit switching. In an easier way to say is VoIP allows user to call others by using internet. However, it is not necessary needs a phone to call others. We can call others by using a computer with some specific software provided that there is internet connection. In addition, VoIP converts analog voice signals into digital data packets and supports real-time, two-way transmission of conversions using Internet protocol. The voice information is sending to the destination in countless individual network packets across the internet. According to IT Business Edge, VoIP has "moved from being a pure voice replacement to being a value-added proposition by seamlessly integrating voice, video and data." For example, Skype, a VoIP service provider, is the most popular of all softphones, with hundreds of millions of subscribers worldwide. We can use Skype to chat, voice calling and video calling to our families, friends and our beloved.

VoIP services can be classified into four types [2]. First of all is peer-to-peer, which users can call each other through the internet connection without using the traditional telephone network. This is the most common because it is easy and free. The second type is VoIP Out which means the call made from the VoIP network to the PSTN. The third one is VoIP in which means the calls can be made from PSTN to the VoIP service using a telephone number. The last type is the two way call which allows calls to be made both ways between the VoIP service and the PSTN using phone numbers.

Besides that, VoIP brings us more advantages compared to PSTN. First of all, we can save a lot of money. On a PSTN line, time is money. We need to pay for each minute that we spend communicating on the phone. Since VoIP uses internet as the backbone, we only need to pay for the monthly internet bill to the Internet Service Provider (ISP). We can talk as much as we want through VoIP and it is free. By using VoIP, we can also setup a conference with more than 2 people and the number of people is not limited. VoIP compress the data during transmission, and this causes more data to be handled over the carrier. Recently, more and more company likes to use Skype to interview people, so that the interviewees do not need to travel so far just for the purpose of interview. Unlike PSTN, we can only have two persons speak at a time.

Lastly, voice quality of the VoIP will not be smooth always. It mostly depends on the bandwidth. Normally, 90kbps is sufficient for good quality of VoIP. Besides that, there are some factors which will determine the voice quality including the choice of codec, packet loss, delay, delay variation (jitter) and the design of network.

1.1 PROBLEM STATEMENT

There are four Kolej Kediaman in Gambang Campus of University Malaysia Pahang and one Kolej Kediaman in Pekan Campus. Currently, the residents in Kolej Kediaman 3 (KK3) are about 1600 people and the residents in Kolej Kediaman 5 (KK5) are about 1200 people. Some of them will use Skype to communicate with their parents, friends or to have their group discussion online. However, sometimes the line is smooth and sometimes the line is very slow. Mostly, the line is slow during the critical time, which is after 6pm. This is because almost all the classes ended after 6 pm. So, what is the maximum number of VoIP users that can be supported when the network is busy.

In addition, the number of students of UMP is increasing over the years, so with the current network setup from KK3 to KK5, can it support current number of users? If it can support current number of users in KK3 and KK5, then what is the maximum number of concurrent VoIP users be supported before it degrades and cannot be consider as a good VoIP session?

Last but not least, residents in the Kolej Kediaman might surf for video tutorial for their assignment. Therefore, how many VoIP users can be supported when there is video streaming? Based on existing research, there are some researches about VoIP in FTP traffic.

1.2 OBJECTIVES

The objectives of this project are:

i. To study the current network setup between KK3 and KK5.

ii. To simulate the VoIP session with the network setup on OPNET Modeler 14.5.

iii. To analyze the VoIP performance based on the jitter, packet end-to-end delay and Mean Opinion Score (MOS), and recommended the maximum concurrent users.

1.3 SCOPES

Due to time constrains, this research is limited and focused to the following:

I. Network topology from KK3 to KK5 which provided by Pusat Teknologi Maklumat & Komunikasi (PTMK) UMP.

II. A metropolitan area network (MAN) is used to connect Gambang Campus and Pekan Campus.

III. OPNET Modeler 14.5 will be used to simulate the network topology and get the result of jitter, end-to-end delay, and MOS.

IV. Windows 7 Professional will be used to install the OPNET Modeler 14.5 software.

V. Performance measurement will be based on the results simulated by OPNET Modeler 14.5.

1.4 THESIS ORGANIZATION

This research consists of six (6) chapters:

Chapter 1 is about overall of the thesis. Problem statement will be introduced here. Besides that, the objective of the research is defined based on the problem statement. Scopes of the research will be discussed here too.

Chapter 2 is about literature review. In this chapter, all the things that related to the research will be study. More details of VoIP will be introduced here.

Chapter 3 will explains the methodology that will be used in this research. The tools that used in this research will be introduced here too.

Chapter 4 is about design and experimentation. The designs phase of the methodology will be further discussed in this chapter.

Chapters 5 is about results and discussion. The results produced by the simulator will be analyzed and discussed in this chapter.

Chapter 6 will concludes all the chapters depending on the results and also recommendations for future researchers.

CHAPTER 2

LITERATURE REVIEW

The primary objective of this research was defined in Chapter 1. In this chapter, background material is presented to help us understand the following chapters of this thesis.

2.0 INTRODUCTION TO VOIP

Voice over Internet Protocol (VoIP) is a technology that allows us to make and receive cheap telephone calls over internet. Today, VoIP is not only restricted to voice communications, it also handles video, text, radio and multimedia communications nowadays. In February 1995, a company in Israel, VocalTec, Inc had introduced the first VoIP product which was InternetPhone. The InternetPhone allowed one user to call another user through their computers with a set of speakers and a microphone. However, this product only worked when both the caller and the receiver had the same software setup. In year 1998, some entrepreneurs started to market PC-to-Phone and phone to phone VoIP solutions. The phone calls are free for nation-wide long distance calls. During that time, the caller needs to listen for an advertisement before the call was connected. Another development in year 1998 was the hardware's invade into the market. Three IP Switch manufactures which are Cisco, Nortel and Lucent had introduced VoIP switching software as a standard in their routing equipment. However, by the end of

1998, less than 1% of calls in United States made using VoIP. In 2000, the total number of calls made by using VoIP had increased to more than 3%. The percentage increased to 25% in year 2003. By year 2005, the major voice quality issue addressed by prioritizing voice over data traffic. Besides that, the projected revenue from VoIP equipment sale is $3 billion. Lastly, $8.5 billion dollars have revenue from VoIP equipment in year 2008 [3] [4].

Today, there are some popular ways of communicating using VoIP. They are discussed briefly below [5] [6]:

1) Analog Telephone Adapter (ATA)

   - ATA allows us to connect a standard PSTN phone to computer or directly to the internet. It is an analog digital converter, so it converts the analog signal from the traditional phone into the digital format for transmission over the internet.

2) IP Phones

   - IP Phone looks like a traditional phone but with a specialized hardware and software on board. It was connected directly into computer using RJ-45 Ethernet connector. It is different with traditional phones because for traditional phones, it only has the standard RJ-11 connectors.

3) Computer-to-Computer

   - This is the most popular and the easiest way of using VoIP and it is completely free. We just need the computers with the internet connection and adequate bandwidth. For good quality voice, the bandwidth needs to be at least 100kbps for a conversation.

Since the VoIP is carry over the network, so we need a codec to encode and decode the digitalized signals. We can measure the voice quality of codec by using Mean Opinion Score (MOS) and R values. However, the quality of VoIP will be affected by a few factors, which are bandwidth, end-to-end delay, packet loss and etc. All of these will be discussed in the later parts of this chapter. Besides, in this chapter, it will also describe some protocols in VoIP.

2.1 HOW VOIP WORKS?

To transfer voice to another computer, VoIP will convert and compress speech from analog form into digital data packets. At the beginning, the microphone will pick up the voice signal from the sender and convert it to analog signal. Then the sound card will digitalized the voice signals. The digitalized signals needs to be compressed in order to transfer the signals over the internet therefore a software driver “codec” will be used to encodes and compresses the signals. The compression limits the frequency range and also reduces the bandwidth stream. After that, the compressed signal will passed to packetizer which will convert the compressed signal into larger pieces and placed into data packets for transmission over an IP network. Average one packet will contain 20 to 30 milliseconds [7]. These packets also contain the information on how to recover the original form. A gateway used to carry out the signaling and packetization role for the telephone call is placed between the codec and the digital data transport circuit. At the receiving end, the packets will be de-packetized by the gateway and the signal will be decoded using the same “codec” and generates the digital output. The digital output will be converted to an analog signal and will be transmitted to speakers. However, to start all these process, we need to have a unique number or identification for a computer to call another. Just like the phone, every one of us has a unique phone number for others to call us [8].

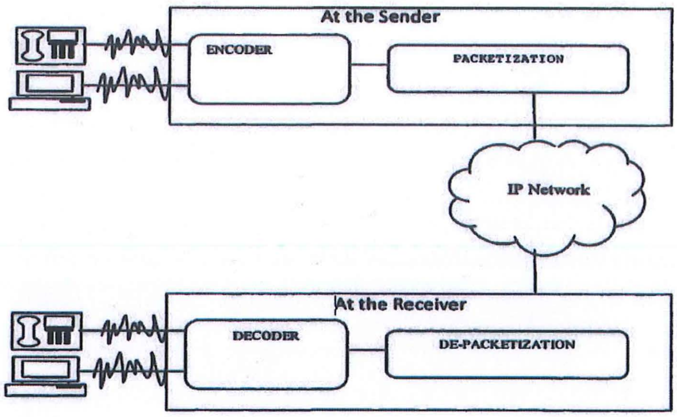


Figure 2.1: High level diagram of the process of digitizing the voice

2.2 PROTOCOLS IN VOIP

VoIP systems need a set of protocols in order to carry voice signal from one computer to another computer. The protocols can be divided into three main parts which are signaling protocols, gateway protocols control and media protocols. There are two types of signaling protocols: H.323 and Session Initiation Protocol (SIP). Next, the gateway protocols control consists of Media Gateway Control Protocol (MGCP). For the media protocols, it consists of Real-Time Transport Protocol (RTP) and Real-Time Transport Control Protocol (RTCP) [9].

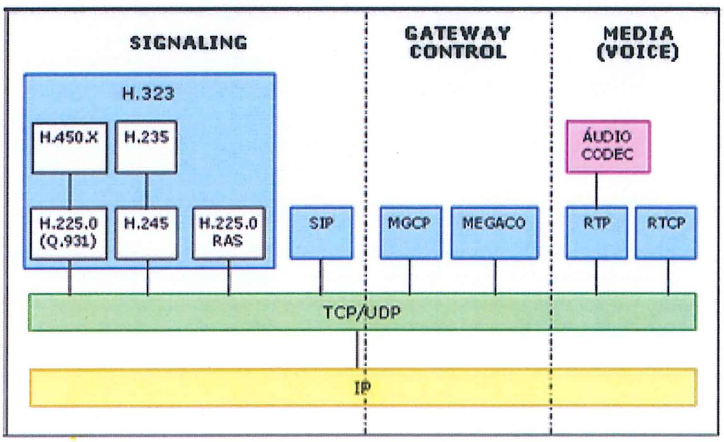


Figure 2.2: Protocols used in VoIP

2.2.1 H.323

Vendors should apply this International Telecommunications Union (ITU-T's) standard when providing VoIP service. There are four logical components defined by H.323 which are Terminals, Gateways, Gatekeepers and Multipoint Control Units (MCUs).

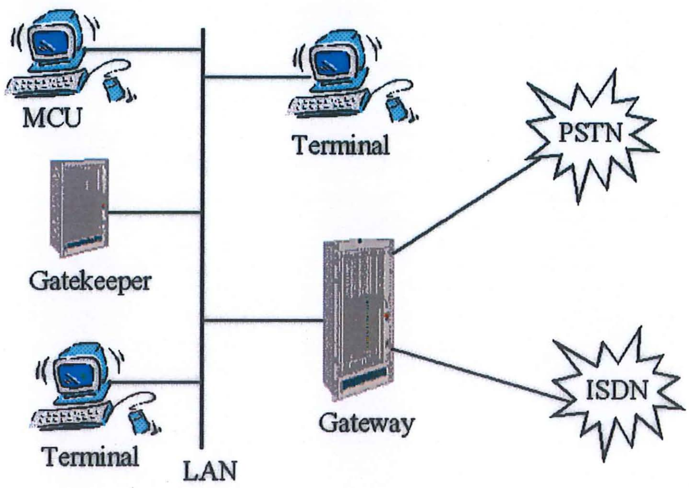


Figure 2.3: Components in H.323

2.2.1.1 Terminals

Terminals are the actual device used on the LAN to provide real-time two way communications. H.323 standard specifies what modes must be supported so that all terminals can work together. It must support H.245 protocol which control the channel usage and capabilities, Q.931 protocol for call setup and signalling, Registration Admission Status (RAS) protocol to communicate with H.323 Gatekeepers and also the RTP/RTCP protocol to sequence the video and audio packets.

2.2.1.2 Gateways

The gateway is an endpoint of the network to provide real-time, two-way communications between H.323 terminals on the IP network and other terminals on a switched based network. The gateway act as a "translator" to perform translation between different transmission formats. They are able to translate between video and audio codecs. Gateway is the interface between internet and PSTN. They take voice from circuit switched PSTN and place it on the public internet and vice versa. The gateways communicate by using the H.245 and Q.931 protocols. It is important when the terminals on a network need to communicate with an endpoint in others network. However, for a single LAN, it is optional to have it [10] [11].

2.2.1.3 Gatekeepers

H.323 gatekeepers provide address translation and had controls access to the LAN for H.323 terminals, Gateways and MCUs. It is optional nodes that manage endpoints in an H.323 network. On startup, endpoints will register with gatekeeper. When they wish to communicate with others endpoint, the endpoint needs to request admission to initiate call using a symbolic alias for the endpoint. If gatekeeper approved the call to proceed, it will return a destination IP address to the originating endpoint. Sometimes, the IP address that the gatekeeper returns not necessary is the actual address of the destination endpoint, it maybe is an intermediate address such as the address that gatekeeper routes call signalling or it may be a proxy address [10].

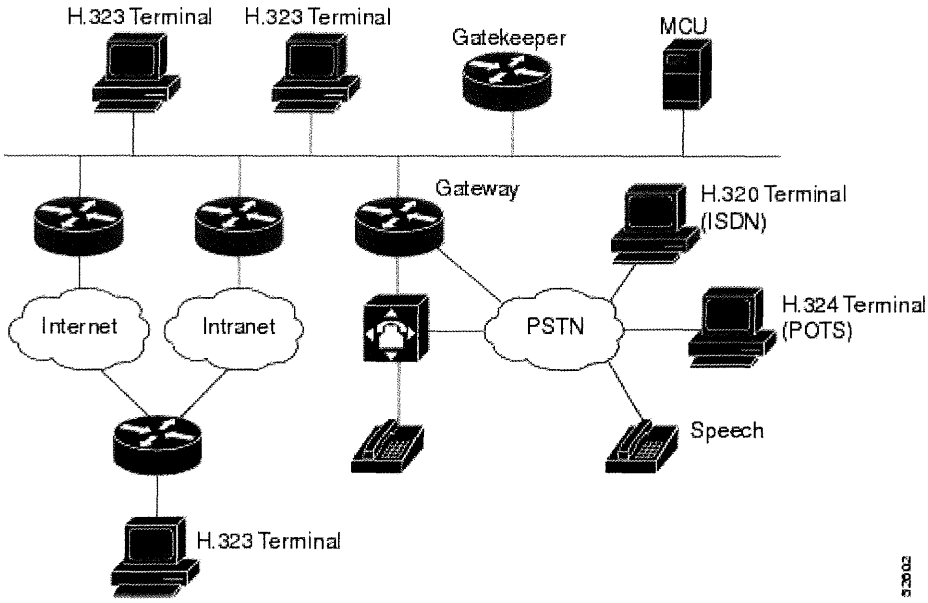


Figure 2.4: Gatekeeper Overview

2.2.1.4 Multipoint Control Units (MCU)

A MCU, an endpoint on the network, allows three or more endpoints to participate in a multipoint conference. It consists of a mandatory Multipoint Controller (MC) and optional Multipoint Processors (MP). MC determines the common capabilities of the terminals by using H.245 while MP controls by MC and mixes audio, data and video from endpoint to create a robust multimedia conference. Besides, it may also connect two endpoints in a point-to-point conference

2.2.2 Session Initiation Protocol (SIP)

SIP, an application layer control protocol for initiating (create), modifying (coordinate), and terminating (tear down) sessions with one or more participants. It is the IETF's Standard for establishing VoIP connections. Its architecture is same like client-server protocol which the requests generated by the client are send to the server. When the server received the requests, it will process the request and send it back to client. Besides, SIP used Session Description Protocol (SDP) for carrying out the negotiation for codec identification. Besides that, SIP invitations used to create sessions that carry session descriptions that allow participants to agree in a set of compatible media types. It also makes use of proxy server to support user mobility and redirecting the request to the user's current location. SIP provides various types of services include: User Location, Call Setup, User Availability, User Capabilities and Call Handling.

SIP consists of two components which are user agents and network servers. User agent can be divided into two parts which are client and server. The client portion is User Agent Client (UAC) which initiates a SIP request. The server portion is User Agent Server (UAS) which is use to receive requests and response it to the user. However, there are 3 types of servers within the network servers which are registration server, proxy server and redirect server. Registration server receives updates based on the current locations of users. Proxy server receiving the requests and forwards them to the next-hop server and the redirect server receive requests and determine the next-hop server and returns the address of the next-hop server to the client [11].

SIP has defined 6 messages to communicating between client and SIP server. The messages are show in Table 2.1.

Table 2.1: SIP Messages and Description

SIP Message	Description
INVITE	Invite user to a call.
BYE	Terminate the connection between two end points.
ACK	Reliable exchange of invitation messages.
OPTIONS	Get information about the capabilities of a call.
REGISTER	Gives information about the location of a user to the SIP registration server.
CANCEL	Terminate the search for a user.

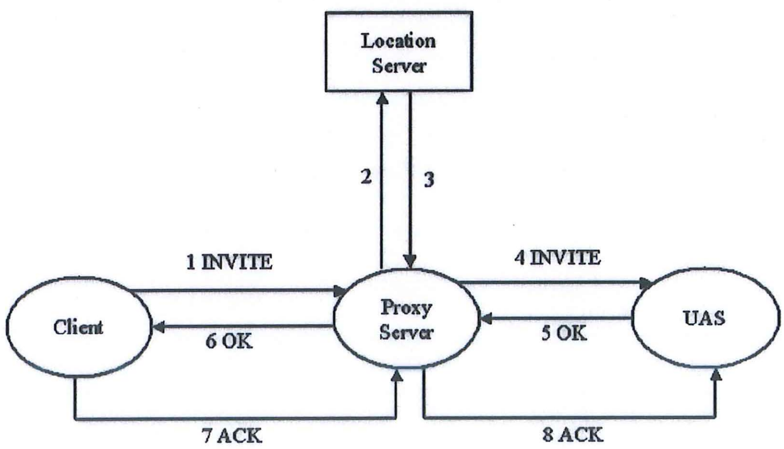


Figure 2.5: Example of SIP Operation

2.2.3 Comparison Between H.323 and SIP

According to proponents of SIP "*Since H.323 was designed with ATM and ISDN signaling, so it is not suit for controlling the VoIP systems. Besides, H.323 is inherently complex and has overheads, so it is inefficient for VoIP. Next, H.323 lacks the extensibility required of the signaling protocol for VoIP too.*" [11] SIP has designed by keeping the internet in mind, so it can avoids both the complexity and extensibility pitfalls. It reuses most of the header fields, encoding rules, error codes and authentication mechanisms of HTTP. Table 2.2 below has summarize some different between H.323 and SIP.

Table 2.2: Comparison between H323 and SIP

H.323	**SIP**
Complex protocol	Comparatively simpler
Binary representation for its messages	Textual representation
Requires full backward compatibility	Does not require full backward compatibility
Not very modular	Very modular
Not very scalable	Highly scalable
Complex signaling	Simple signaling
Large share of market	Backed by IETF
Hundreds of elements	Only 37 headers
Loop detection is difficult	Loop detection is comparatively easy

2.3 CODECS

CODEC is a short form for Coder/DECoder. It performs the process of encoding and decoding the voice on an algorithm specified by G.7xx standards. Besides that, it performs the task of converting a voice signal into a format suitable for transport over the network. After that, it will compress the digital signal and put into packets at the sending end and transmission over the network. This process is known as packetization. At the receiving end, the codec will decompresses the signal and the voice will release from the speaker.

There are different types of CODECs available for implementing in VoIP. Every CODEC has its own properties making it suitable for different networks. According to ITU-T, a G.7xx CODEC is suitable for the audio compression/ de-compression [5]. The most popular standards recommended by G.7xx CODECs are G.711, G.729, and G.723.1.

Besides that, there are two standards available for expressing the quality of voice stream, which are Mean Opinion Score (MOS) and R Values.

2.3.1 Mean Opinion Score (MOS)

MOS is one of the most important factors to measure the quality of performance of the speech CODEC that used in conversion of analog signal to digital signal. MOS has reflects the performance of various CODECs under different circumference. Besides, the quality of the received media after the compression represent by a numeric number. This is the subjective method because the result is based on a group of listener. Every listener will give a rating to the audio that they are listen to. This will be done by large samples so that more accurate results will be generated. MOS value is a number from 1 to 5. Number 5 indicates the best quality where for number 1 indicates the poorest quality. Table 2.3 show the MOS rating table. MOS is a very important measure to check the quality of the voice output that produced by the CODEC. The best CODEC can be chosen by comparing their MOS value. The MOS score for different audio CODEC are shown in Table 2.4.

Table 2.3: MOS Rating Table

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly Annoying
2	Poor	Annoying
1	Bad	Very Annoying

Table 2.4: MOS Score for Different Audio CODECS

CODEC	MOS
G.711	4.1
G.723	3.8
G.729	3.92

2.3.2 R Values

R Value is another way to express the quality of a voice stream that recommended by the Telecommunication Industry Association (TIA) [14]. The value of R Value is between 1 to 100 while 1 is the least quality and 100 is the best quality. The result is based on the user satisfaction with the quality of voice signal after it transmitted to the destination. R Value is significant in the selection of CODEC scheme but it also should not be the only reason for the selection of CODEC. Table 2.5 show the user satisfaction as Function of R Value.

Table 2.5: User Satisfaction as Function of R Value

R Value	User Satisfaction
90 - 100	Very Satisfied
80 - 89	Satisfied
70 - 79	Some users dissatisfied
60 - 69	Many users dissatisfied
50 - 59	All users dissatisfied
00 - 49	Not Recommend

2.3.3 Types and Intrinsic Values of CODEC

As mention in 2.3, ITU-T had recommend G.7xx standards of CODECs for audio compression and de-compression. These CODECs are mainly used in telephony. Some of the popular CODECs are described and compare in the table below.

Table 2.6: Specification of popular CODECS

Intrinsic Values\CODEC	G.711	G.729	G.723.1
Compression Ratio	1:1	8:1	10.1
CODEC Bit rate (Kbps)	64	8	6.3
CODEC Sample interval (ms)	10	10	10
CODEC Sample Size (Bytes)	80	10	24
Packets Per Second	50	50	34
Mean Opinion Score	4.1	3.7	3.65
R Value	83	73	77
Voice payload (Bytes)	160	20	24
Bandwidth (Kbps)	87.2	31.2	21.9

The intrinsic values are briefly described below:

a) Compression ratio
   - The ratio of the original collected analog voice sample to the digitalized voice.
b) CODEC bit rate
   - The rate at which the media stream needs to be transmitted for the delivery of voice call.
c) CODEC Sample Interval
   - Interval at which the CODEC operates.
d) CODEC Sample Size
   - Number of bytes captured by the CODEC at each sample interval.
e) Packets Per Second
   - Number of packets to be transmitted per second to deliver CODEC bit rate.
f) Voice Payload
   - Number of bytes in each packet and is a multiple of codec sample size.

2.4 FACTORS AFFECTING PERFORMANCE OF VOIP

There are a few factors will affect the voice quality. For example end-to-end delay, packet loss, jitter, bandwidth and so on. These factors need to be kept within an acceptable range to make sure the good quality of service. Therefore, the evaluation of these factors becomes very important to help understand the correction to be done to the existing network for the implementation of VoIP.

2.4.1 End-to-End Delay

End-to-end delay is the delay of the voice packet during the journey through the network from one endpoint to another endpoint. It is the most important factors to determine the quality of the call. According to ITU-T recommendation, the end-to-end delay for the voice packet need to within the range of 150 milliseconds. However, based on lab testing, it shown that there is a negligible difference in voice quality MOS using networks built with 200ms delay budgets. Therefore, if constraints exist and this delay target cannot be met, the delay boundary can be extended to 200ms without significant impact on voice quality [22]. The higher the delay, the quality of voice conversation will be lower. Therefore it is better to have low end-to-end delay. End-to-end delay cannot be avoided but we can minimize the delay. Elements of end-to-end delay can be categorized into two, which are endpoint delays and network delays. Endpoint delays is the delay occurs at the sender node and the receiver node. However, network delays occur between the route that packet travels from source to destination.

Delay at the sender consist two different delay which are encoding delay and packetization delay. The encoding delay occurs when the voice CODEC encoding the voice signal. Every CODEC has a sample interval during which it collects speech samples/ frames. Every frame consists of a specified number of speech samples. The frames will not generate until all the speech samples are collected by the CODEC. Therefore, there would be at least some delay of forming the first frame. Besides that, many CODECs look into the succeeding frame to improve the encoding efficiency and causes the look-ahead delay occurs. There is also a processing delay to encode the block of speech samples. In a nutshell, the encoding is the combination of first frame delay, look-ahead delay and the processing delay. On the other hand, packetization delay is the time taken by the CODEC to convert the encoded frames into packets that need to be sends out to the network. Packetization delay can be calculated by multiply the number of frame in each packet with the frame size [5] [13].

Network delay occurs when the voice traffic is travelling through the network before the receiver node. The delay in the network can be categorized into four types. First of all is the processing delay. It occurs when the packet header is examined to determine which path to go. Follow by the transmission delay. It occurs when the time taken for a packet to serialize and push into a link from the source. The third type of delay is propagation delay. It happens from the source until it reach the destination and it is depends on the distance between the source and the destination. The last delay is the queuing delay. Queuing delay occurs when a packet are waiting in a queue of the network node to be transmitted to the destination. Basically, the queuing delay is in the order of milliseconds to microseconds. The queuing delay will be zero if the queue is empty.

Last but not least, the delay at the receiver is cause due to the different processes that the packets need to go through before the voice is delivered. It can be split into three types of delay. First and foremost is the jitter buffer delay. It cause by the packets arrive at different times. It can be reduce by introducing the jitter buffer. Jitter buffer enables the smooth playback of voice even if some packets arrive late. Next is the de-packetization delay. It happen when the CODEC process the encoded frames from the packet. Decoding delay is also one of the delays at the receiver end. It happened when CODEC at the receiver end to decode the encoded frames. The last delay is the playback delay. Playback delay occurs when the voice packet waits to be playback after it arrives at the receiving end [5, 13, 15].

2.4.2 Jitter

Jitter is the time different between the successively arriving packets. There is another name for jitter which is Instantaneous Packet Delay (IPDV). For example, packet A takes 50 ms to traverse through the network however packet B takes 80ms for the same trip. In this case, the jitter is 30ms. Jitter is the most common problem for VoIP. It can be affected by computer usage, the length and quality of Ethernet cables and some others issues. Jitter cannot be avoided but it can be minimize. High levels of jitter lead to large numbers of packets to be discarded by the jitter buffer in the receiving IP phone or gateway and result in severe distortion in call quality. Voice packet can tolerate up to 75 milliseconds. However, the lower jitter will produced the better voice quality.

To correct the effect of jitter, jitter buffer is used. VoIP endpoints collect packets in buffer and reorder them according to their timing and sequence number before the listener hears them. This is work but this also will increase the delay. Processing the buffer adds delay to the call and the bigger the buffer, the longer the delay. If voice packets arrive while the buffer is full, then the packets will be dropped and the receiver will never hear about them.

In the figure below, it shows that the packets sent in a continuous stream evenly at the sender's side. However, due to network congestion and improper queuing, the steady stream became uneven because the delay among each packet is different.

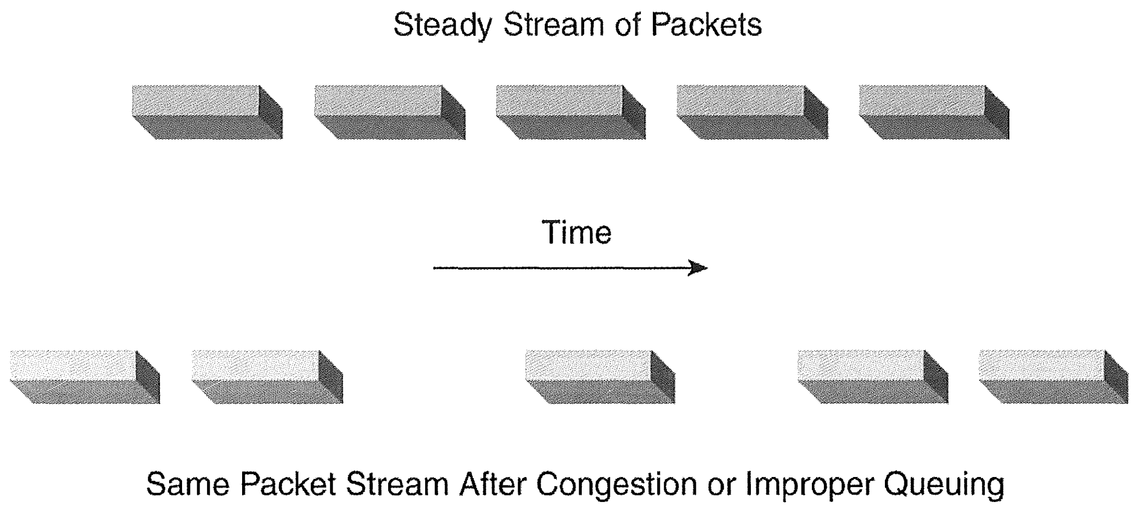


Figure 2.6: Jitter between the packets

2.4.3 Packet Loss

Packet loss occurs when the packets sent but it did not reach the destination. It happens in routers and switches. When network congestion occurs, the buffers in routers and switches will overflow. Number of the packet the buffer can store is limited, such as 100, 500 or 1000 packets. The packets are dropped to manage the network traffic. Corrupted packets rejected in-transit and poor networking hardware will also cause the packet loss. Dropped packets in VoIP are treated as noise. Besides that, packet loss will affect the voice quality in IP telephony. However, packet loss is accepted if it is less than 0.2 percent for a good quality in VoIP network.

Just imagine that when you are having a VoIP session with someone and the packet loss brings to some words are missing when you are calling with that person, it might have a chance for you to misunderstanding that person and not understand the conversation. Ideally, there must be no packet loss in VoIP call.

To recover the loss packets properly, many applications are designed to provide a reliable network using transport protocol such as TCP to request a retransmission of packet loss and insure an acceptable and stable transmission. However in VoIP, the voice is transmitted using UDP protocol so that there is no retransmission of the lost packets. This is because in VoIP, retransmission of lost packets will induce more delay and would lose it relevance in real time. In conclusion, we need to have low packet loss rate for VoIP telephony.

2.4.4 Bandwidth

The most important factor for any network planning is the bandwidth. In a VoIP network, the voice and data are all depends on the bandwidth. Besides that, the quality of voice will be much affected when the bandwidth is insufficient. However, the amount of bandwidth needed to be allocated depends on the number of calls during the peak time. In packet networks, bandwidth refers to quantity of data that a network component or the network path can transfer per unit of time.

2.5 INTRODUCTION TO METROPOLITAN AREA ETHERNET(METRO-E)

There are a few types of network such as Local Area Network (LAN), Wide Area Network (WAN) and also Metropolitan Area Network (MAN). LAN is used to connect networking devices that are in a very near geographical area. For example, LAN is used in a building, different floor of the building and within a campus environment. WAN are used to connect LANs together. It is used when the LAN must be connected but separated by large distance. However, MAN is a hybrid between LAN and WAN. It is a large network that spans a city or large campus. For example, a company has two branches in a city. So, MAN is used to connect the branch together [16]. A MAN typically covers an area of between 5 and 50 km diameter. According to Kenneth C. Laudan and Jade P. Laudon, "*A Metropolitan Area Network (MAN) is a large computer network that spans a* *metropolitan area or campus. Its geographic scope falls between a WAN and LAN. MANs provide internet connectivity for LANs in a metropolitan region, and connect them to wider area networks like Ethernet.*" [17].

Metro-E is a cost-effective and flexible service that allows us to upgrade to higher bandwidth without incurring the costs of setting up new equipments. Metro-E also brings us some benefits. First of all, Metro-E is highly scalability. It enhances the ability to use a network service that is ideal for wide variety of business for data, voice and video. With bandwidth scalability and flexibility, we only need to pay for what we need. Metro-E is very simple to manage. Next, Metro-E is cost effectiveness. Ethernet can reduce subscribers' capital expenses and operation expenses in a few ways: First is due to the broad usage in almost all networking products, the Ethernet interface itself is very cheap. Second, it allows subscribers to add bandwidth more incrementally. The subscribers only need to pay for what they need. Third, Ethernet services often less than competing services due to lower equipment, service and operational costs. Last but not least, Metro-E is flexibility. Many Ethernet services allow subscribers to network their business in ways that are more complex. Subscribers are able to add or change bandwidth within a few minutes instead of a few days or weeks when they are using other access network services. The most important is the changes do not require the subscribers to purchase for new equipment.

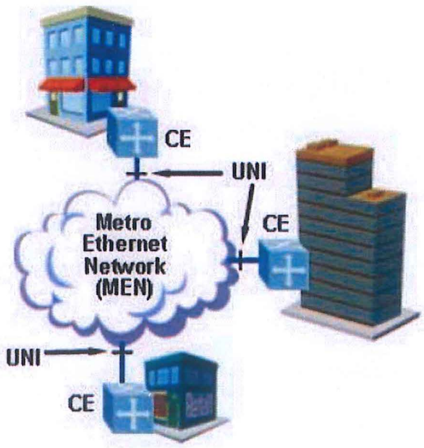


Figure 2.7: Basic Model of Metro-E

2.6 DIFFERENT BETWEEN VPN, MLPS, AND METRO-ETHERNET

Today, Wide Area Network technologies have fall abroad into three categories which are Multi Protocol Label Switching (MPLS), Virtual Private Network (VPN tunneling) and Metro Ethernet. VPN is a network that uses a public telecommunication infrastructure such as internet to provide remote offices or individual users with secure access to their organization network. VPN keeps data safe as it crosses the public internet by using encryption such as PPP and IP-Sec. However, network congestion and other issues can make the internet connection slow since it uses the internet to get data from location to location [18].

MPLS is a technology for speeding up network traffic flow and making it easier to manage. It involves setting up a specific path for a given sequence of packets, identified by a label put in each packet. It works with Internet Protocol (IP), Asynchronous Transport Mode (ATM) and frame relay network protocols. It is highly secure because the traffic of MPLS is segmented from other users on the carrier's network.

Metro-E is not a router-based technology. It is different with VPN and MPLS. Metro-E operates at more fundamental transport layer, and creates point-to-point or point-to-multipoint paths using switches rather than routers. Since its architecture is point-to-point, therefore it is highly secure and completely segregated from internet [19].

2.7 IMPACT OF EXISTING VOIP SESSION GIVEN BY VARIOUS TYPES OF NETWORK TRAFFIC

As mention before, there are some factors will affects the performance of VoIP. However, the performance of VoIP will also be affected by the network traffic. For example, a network will not only carry the VoIP, it will also carry others external data just like HTTP, FTP and so on. So, as the traffic is busy, the performance of VoIP will get affected. Besides that, different results will be generated by different network. For example, the maximum user for LAN and wireless network is different due to the bandwidth.

2.8 EXISTING RESEARCH

In order to research on the maximum concurrent users that a VoIP network can support, analysis and comparison between researches have been made. The information collected by reading the journals and articles.

2.8.1 Analysis of Performance of VoIP Over various Scenarios

This research evaluate delays and distortion issues that VoIP might have while increasing the traffic load and by adding extra models to the system and evaluate the impact to the overall QoS. The research is simulated in OPNET 14 Software. Five scenarios had been simulated in this research. First and foremost is VoIP call in LAN. The second scenario is the long distance VoIP Calls under LAN. The third scenario is the VoIP calls in LAN with ftp server. Followed by the VoIP calls in WLAN, and the last scenario is VoIP in WLAN with interference.

In scenario 1, it tests the performance of VoIP call in LAN of a small office. The number of client set to 2 nodes at the beginning and gradually increases to 100 nodes. The result shows that the jitter and end-to-end delay increased as the number of nodes increased. Besides that, the delay variation increase exponentially with the number of calls. Last but not least, the packet are lost when the number of calls getting high.

For scenario 2, it tests the long distance VoIP calls under LAN. The calls are made between two offices with LAN placed across the country. Jitter, delay variation and end-to-end delay increased when the distance between two offices increases.

A FTP server is added into the LAN network of VoIP in scenario 3. In this case, the voice jitter, delay variation and end-to-end delay is not affected by the ftp server. It is to be assumed that there is enough bandwidth for the network.

Scenario 4 applies VoIP in WLAN of a small office. The number of nodes is increased from 2 to 20 nodes. The performance of VoIP in WLAN is very poor if compared to wired network. Jitter, delay variation and end-to-end delay in wireless network is multiple times larger than wired network. Besides that, many of the packets are lost.

The last scenario is the VoIP in WLAN with interference. A jammer is added to the network to examine the influence of interference to the network. The result showed that the wireless network with interference has larger jitter, delay variation and packet end-to-end delay.

2.8.2 Analysis on VoIP using OPNET

This research of study is to simulate a VoIP network and study the behavior and quality of VoIP under different scenarios. Besides that, they also study the potential parameters that can deteriorate the quality of VoIP. In the research, they have created different scenarios to understand the behavior of VoIP. The first scenario is the comparison between local and long-distance VoIP communication. The second scenario is the comparison between a busy VoIP network and a non-busy VoIP network. The third scenario is the observation of VoIP quality under different discard ratio (Internet QoS) and the last scenario is the different encoder schemes usage and their effects on VoIP quality.

In scenario 1, it has three types of topology. The first topology is about the long distance VoIP conversation pair which connects two companies in different country. The second topology is about local conversation pair for the same floor in a building and the last topology is the local conversation pair in a same building but different floor. The topology was drawn in OPNET and the result was simulated. Based on the result, the long-distance VoIP conversation pair tends to have more in jitter, longer end-to-end delay and smaller MOS value compared to the local conversation pairs. Therefore the quality of long-distance VoIP communication is not good as the quality of local distance VoIP communication.

For scenario 2, they used 15 workstations in each company and connect to the second company with also 15 workstations. The first 15 VoIP calls start after 10 seconds and each workstation will generate an additional call every 10 second after. This is to generate a busy VoIP network. They test the simulation by using DS1 link and DS3. The result has shown that the quality of VoIP become worst as the network getting busy. The overload happens and gets the larger jitter and longer end-to-end delay happen when using DS1. DS3 is then replacing DS1 in the network and the quality of VoIP improved.

Next, the purpose of scenario 3 is to observe how the internet QoS affects the quality of VoIP. They observe it by changing the packet discard ratio into 0.5%, 4% and 6% for IP could. The result of their research has shown that the network with 6% discard ratio has the highest jitter fluctuation but have the shortest end-to-end delay. In addition, the result also show that the higher the discard ratio, in a network, the more the packet loss and the lower the MOS.

In the last scenario, they had implemented different CODEC to verify would the CODEC affect the quality of VoIP. The CODEC of G.723, G. 711 and G. 729 is used in the scenario. The result shows that CODEC G.711 has higher quality compared to the others.

2.8.3 Study of VoIP In Wired and Wireless Networks

This research studies about the VoIP in Ethernet and Wireless networks by evaluating the End-to-End delay, jitter, packet loss and bandwidth that affect the quality of a voice in a campus network. They implement a campus scenario in the OPNET Modeler 14.5 and quality of voice is measuring by these parameters. After that, they also suggest the maximum concurrent users can be supported. This research using the 100 Mbps and G.711 CODEC in Ethernet and 802.11g wireless network with 54 Mbps. The wired model consists of a main campus router and two Ethernet switches which are connected to four different part of the campus. The network has same number of VoIP users for each zones. However, wireless network is almost the same as wired network as the four zones in LAN is replaced by wireless VoIP users.

The wired network is loaded with voice traffic beginning with 10% up to 60%. After that, the wired network is loaded with data traffic along with the voice traffic to see the end-to-end delay crosses the maximum tolerable amount of delay. In the result of this study, the end-to-end delay crosses the maximum tolerable delay when the network is loaded with 35% voice and background traffic.

For the wireless network, it is loaded with voice traffic beginning at 1% up to 2%. After that, the end-to-end and jitter are measured. After that, it will continue loaded with data traffic along with voice traffic beginning at 3% and up to 10 %. The result of this research is that the network is only able to handle up to 1% or a total of 8 users in the wireless network. It is not manage to support more VoIP calls from more than 2% when they are loaded into the network.

In a conclusion, in wired network, it can support up to 25%, about 546 VoIP callers without any traffic in a wired network. On the other hand, the wireless network can only support up to 8 users, which is only 1%.

2.9 COMPARISON BETWEEN PROPOSED RESEARCH AND EXISTING RESEARCH

2.9.1 Proposed Research In This Thesis

In this research, it will analyze the performance of VoIP between KK3 and KK5 in University Malaysia Pahang. Since both KK is in different campus, therefore, there is a Metro-E connection between two campuses. The diagram below has illustrated the network topology between KK3 and KK5.

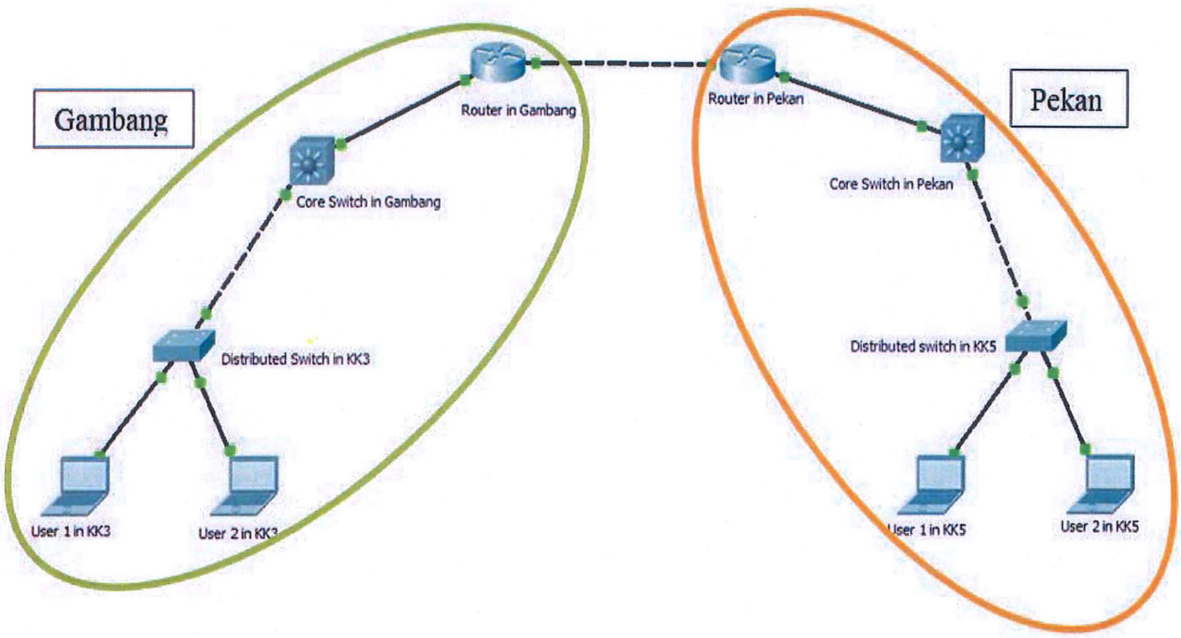


Figure 2.8: Network Topology between KK3 and KK5

Based on the network above, the connection between Gambang's router and Pekan's router is a MAN. In KK3, all the users connected to one layer 3 distributed switch with 1Gbps network. Then, the distributed switch will connect to the Core switch in Gambang. The speed between core switch and distributed switch is 10Gbps. Core switch in Gambang will connect to the router in Gambang by a Fast Ethernet. The router connects to another router in Pekan through the Metro Ethernet provided by TM which is only 100Mbps. After that, the Fast Ethernet connection will connect router in Pekan with the core switch in Pekan. The core switch also uses 10Gbps connection connect to distributed switch in Pekan and 1Gbps network from the distributed switch to the users in KK5.

In this research, CODEC will be used is G.711 CODEC. It is been chosen because according to previous research, it is the best CODEC among the others. After that, jitter, MOS and end-to-end delay will be observed to determine the maximum concurrent users can be supported by this network.

2.9.2 Comparison between Proposed Research and Existing Research

Table 2.7: Comparison between Proposed Research and Existing Research

Proposed Research		**Existing Research**
	Similarity	
	1) Uses OPNET Modeler 14.5 to simulate the results 2) Jitter, MOS and end-to-end delay is observed. 3) Simulate results of busy and non-busy network	
1) Implement VoIP in a MAN network. 2) CODEC is fixed during the research.	**Differences**	1) Implement VoIP in LAN, and WLAN network. 2) CODEC is a manipulating variable which used to get different results.

2.9.3 Comparison between Existing Research

Table 2.8: Comparison between Existing Research

Research Title	Author	Method	Result
Analysis of Performance of VoIP Over Various Scenarios	Yue Pan, Jeffery Chung, ZiYue Zhang	1) VoIP call in LAN	1) Jitter, delay variation and end-to-end delay increased as the number of nodes increased. Packets are lost when number of calls getting high.
		2) Long Distance VoIP Calls under LAN	2) Jitter, delay variation and end-to-end delay increased when the distance between two offices increases
		3) VoIP calls in LAN with ftp server	3) Voice jitter, delay variation and end-to-end delay is not affected by the ftp server since the bandwidth is enough for the network.
		4) VoIP Calls in WLAN	4) Jitter, delay variation and end-to-end delay in wireless network is multiple times larger than

		5) VoIP in WLAN with interference.	wired network. Besides that, many of the packets are lost. 5) Jitter, delay variation and end-to-end delay become larger.
Analysis on VoIP using OPNET	Benson Lam, Winfield Zhao, Mincong Luo	1) local and long distance VoIP communication 2) Busy network with non-busy VoIP network 3) VoIP quality under different discard ratio 4) Effect of encoder used on VoIP Quality	1) Long distance VoIP communication is worse than local-distance VoIP. 2) A high capacity link such as DS3need to be used in a busy network. 3) The lower the packet discard ratio, the better the VoIP communication. 4) G.711 is the best CODEC if compare to others.
Study of VoIP in Wired and Wireless Networks	Krishna Senthil Kumar Kuthalingam	1) wired network is loaded from 10% to 60% 2) Wireless network is loaded from 1% to 10%	1) Wired network can only support up to 35% or 546 VoIP callers without any background traffic in the network. 2) Wireless network can support up to 8 users or 1% of the total network without background traffic in the wireless.

CHAPTER 3

RESEARCH METHODOLOGY

3.0 INTRODUCTION

According to Majid Konting (1990), a research method is the technique and specific methods used to obtain the information needed to solve problems, especially to achieve a research objective. Research methodology is defined as the systematic and theoretical analysis of the methods which applied to a field of study, or the theoretical analysis of the body of methods and principles associated with a branch of knowledge. Besides, it also known as science of studying that how research is done scientifically either in techniques, methods and procedures needed in order to carry out the research.

In this chapter, it had defined the important items in research methodology such as the hardware, software, research system architecture and etc. In order to achieve the objectives in this research, a few strategies had been identified as a guideline. The strategies are:

a) **Information gathering**: the data is collected from resources and interview based on the research topic.
b) **Planning and identifying**: Identifying all the software which needed to carry out for the research.
c) **Design and installation of the required software:** The software was installed and configured to ensure that the experiment can be carried out properly.
d) **Testing and implementation:** The research was conducted and simulate in the software.
e) **Data analysis:** All the results obtained will be gathered and analyzed.

INFORMATION GATHERING → **PLANNING AND IDENTIFYING HARDWARE AND SOFTWARE TOOLS** → **HARDWARE SETUP, SOFTWARE CONFIGURATION AND EXPERIMENT DESIGN** → **IMPLEMENTATION AND SIMULATION** → **DATA ANALYSIS**

PRELIMINARY STUDY PHASE	
PRELIMINARY STUDY	Offline: Interview, books, journals, thesis project and research papers
	Online: Articles, journals and proceeding materials
RESEARCH PLANNING PHASE	
HARDWARE	Laptop
SOFTWARE	OPNET Modeler 14.5
	Microsoft Visual Studio 2010
ARCHITECTURE DESIGN PHASE	
INSTALLATION	Software tools into the laptop
SETTING AND CONFIGURATION	OPNET Modeler 14.5
ARCHITECTURE (SIMULATION MODE)	• Create the topology from KK3 to KK5 • 2 Core switch • 4 Access switch • 2 Distributed Switch • 2 Router
TESTING PHASE	
• Test the network from KK3 to KK5 by increasing the number of user and recommended the maximum concurrent user in topology without traffic and with traffic.	
DATA ANALYSIS PHASE	
• Things to consider are the results of jitter, end-to-end delay and the MOS generated by the OPNET Modeler.	

3.1 OVERVIEW OF THE METHODOLOGY

Five important phases had been used as the guidelines to avoid hindrance and ensure the research can be runs smoothly. The five phases are defined as in the figure above which are:

a) Preliminary Study Phase
b) Research Planning Phase
c) Architecture Design Phase
d) Testing Phase
e) Data Analysis Phase

3.2 PRELIMINARY STUDY PHASE

The preliminary study phase is the first phase out of the five phases. In this phase, it is to study about the research title and to search and gather the right information which is related to the problem statements. The research title and the problem statements had been used as a guideline to extract the related information. Besides that, it ensured the research is on the right path and the objective of this research can be achieved successfully.

In this phase, the study includes finding journals, articles online, books, research papers and interview to collect and gather all the information about this research. In addition, methods, tools and some recommendations were taking into concern and as a reference which can be used in this research.

Through this phase, it shows that many research about VoIP had been done by others in different scenarios. Jitter, end-to-end delay and MOS had been used to determine the recommended maximum concurrent users in the network.

Besides that, since the real network topology will be used during the research, therefore, an interview with the Pusat Teknologi Maklumat dan Komunikasi (PTMK) University Malaysia Pahang is done to get the real network topology. In addition, an interview session with person-in-charge of KK3 and KK5 is needed to get the current number of residents who live in these two hostels. After the interview, I get the number of residents in KK3 is about 1600 and KK5 is about 1200.

The diagram below show the network topology provided by PTMK.

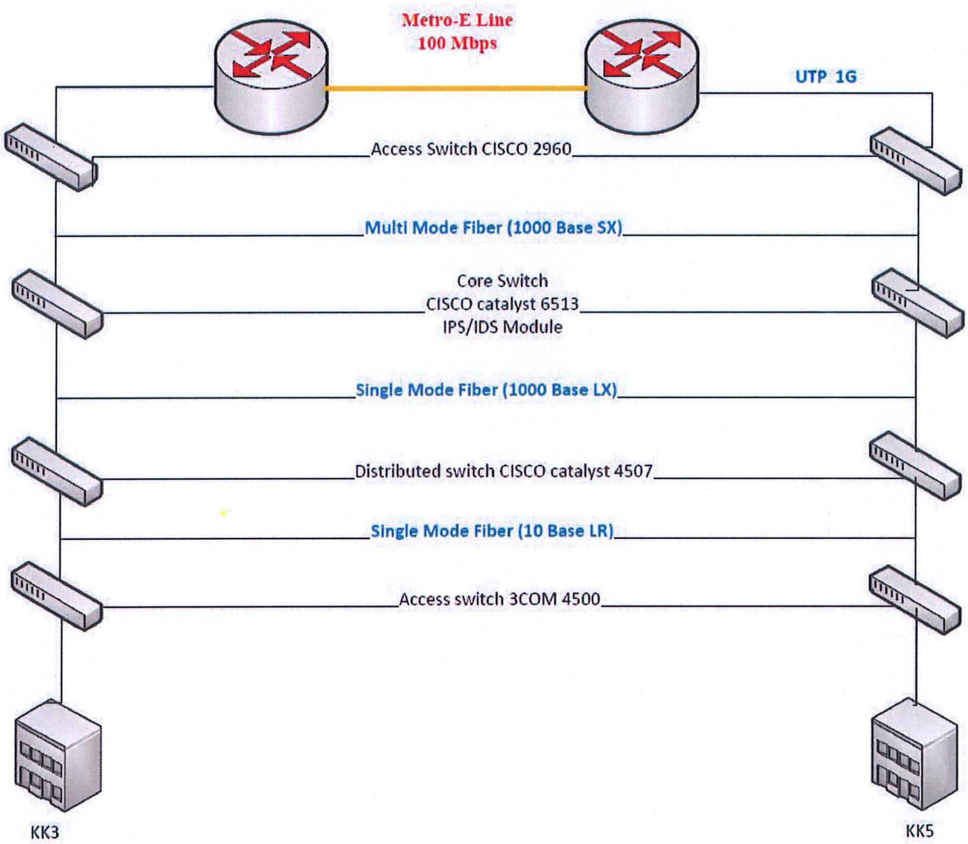


Figure 3.1: Network Topology Provided by PTMK

3.3 RESEARCH PLANNING PHASE

Follow by the preliminary study phase is the research planning phase. This phase plays an important role because the research needed to be plan carefully before it is done to prevent insufficient time and prevent any other problems occurs at the later phases. In this phase, it had determined the hardware and the software which going to be used during the research.

The hardware required in this research is only a laptop. It is needed to install the software to be used in this research. However, there are two software will be used during the research which are Microsoft Visual Studio 2010 and OPNET Modeler 14.5. In this case, Microsoft Visual Studio 2010 was used as a compiler so that OPNET Modeler 14.5 can be run smoothly. OPNET Modeler 14.5 is the main and the most important software to be used during the research. It is used to implement the network topology and simulate the results to be analyzed in later part. By using this tool it will be able to generate the graph for end-to-end delay, jitter, MOS which will be used in the last phase, data analysis phase.

However, the ways to configure the devices that will be used in the OPNET Modeler need to be consider too. This step is to make sure that the VoIP can be implemented in the software.

3.4 ARCHITECTURE DESIGN PHASE

In this phase, the work of installing and configuring the software will be conducted. This phase have to be perfect because the configuration have been made in the software will directly affect the accuracy of the results.

The network topology which provided by the PTMK will be setup and simulate in the OPNET Modeler 14.5. The diagram is shown in Figure 3.2.

The network topology contains:

- 2 Layer 3 Core Switches
- 2 Distributed Switches
- 4 Access switches
- 2 1000 Base X LANs which consists 1600 end users in KK3 and 1200 end users in KK5
- 2 Routers
- 1 Application Definition for VoIP
- 1 Profile Definition
- 1 100 Mbps link
- 2 UTP 1 Gbps link
- 6 Single Mode Fiber (1 Gbps)
- 2 10 Gbps Fiber

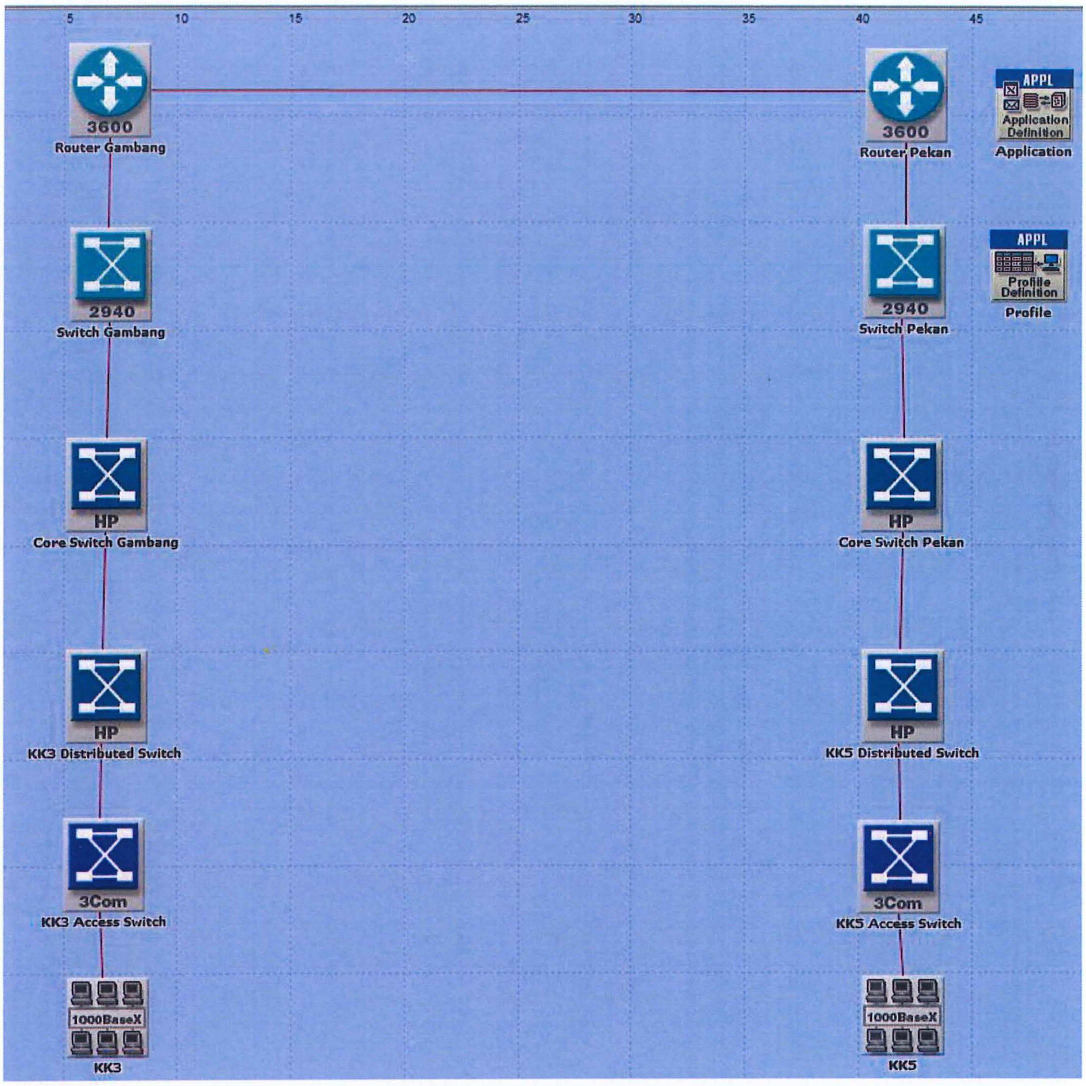


Figure 3.2: Network Topology Drawn in OPNET Modeler

3.5 TESTING PHASE

After the design was completed, it will be proceeding to testing phase. In this phase, the predefined tools will be run to obtain the graphs to be analyzed in the next phase. The simulation needed to be accurate and precise. In this research, the numbers of users will be increased gradually as the bandwidth remains the same. The results will be collected in the table below.

Table 3.1: Table to Summarize the Results for Each Scenario

Scenario With/ Without Traffic			
Number of Users	Jitter	Packet Loss	End-to-end delay

3.6 DATA ANALYSIS PHASE

This phase is the last phase of this research where the data's are obtained and collected. The collected data is very important in this research as it will be used for interpreted and analyzed to achieve the objective in this research. For example, the generated graphs will be analyzed and the maximum concurrent users will be recommended at the end of research.

The maximum concurrent users will be determined by analyze the graphs of jitter, end-to-end delay, and MOS. So, each graphs will be analyze when the number of users increased.

CHAPTER 4

DESIGN AND IMPLEMENTATION

4.0 INTRODUCTION

In this chapter, it will discuss in details of design and implementing phase which how the research is carried out. A step-by-step procedure needs to be defined properly so that the research project is on the right track and the outcome is achieved. In addition, design and implementation phase was defined according to the objective stated in Chapter 1 to ensure the objectives of the particular research are achieved.

In this research, the network topology provided by PTMK will be drawn exactly in OPNET Modeler 14.5. The design will contain a metro network which connected both campuses together. It is very important to properly configure the related components in the OPNET Modeler. This is because any minor changes or wrong configuration will affect the results of even gives us the errors during compilation. In the research, two types of scenario will be created to test for the maximum concurrent VoIP users. The first scenario is the network without traffic (only VoIP traffic), and another scenario is network with traffic (VoIP traffic with HTTP traffic and Video conferencing traffic).

4.1 OPNET

OPNET is an object-oriented simulation tool for planning and modulating and for the performance analysis of the simulation of communication networks in general. This simulation program offers a great number of models for network elements. Besides, it has the ability to simulate a real-life network with all of its complications, simply and accurately. It has other features, such as a complete library of network protocols, a user-friendly Graphical User Interface and easier data collection and results analysis [20][21].

Regard to VoIP, this version of OPNET Modeler can simulate voice traffic. The result obtained can be in either statistical or graphical. In addition, it can generate the graph of jitter, end-to-end delay, and MOS.

The performance of the OPNET simulator differs with different platforms. For this research, the windows version was used. The following will summarize the specifications of the simulation test bed.

- OPNET Modeler 14.5
- Hardware platform: DELL
- Operating System: Windows 7 Professional SP1
- Intel Core i7-3770 CPU @ 3.4 GHz
- 12. 00GB of RAM

4.2 EXPERIMENT NETWORK DESIGN AND ARCHITECTURE

The network topology will be drawn in the software, OPNET Modeler 14.5. The network topology will consist of:

- 4 Access Switches
- 2 Layer 3 Core Switches
- 2 Distributed Switches
- 2 Routers
- 2 1000 Base X LANs which consists 1600 end users in KK3 and 1200 end users in KK5
- 1 Application Definition and Profile Definition for VOIP
- 100 Mbps link which connected 2 routers.
- 1 Gbps UTP link which connected between routers and access switch
- 1 Gbps Single Mode Fiber which connected between access switch and core switch, access switch and distributed switch and also access switch to the LAN.
- 10 Gbps Fiber which connected between core switch and distributed switch.

In Figure 4.1, it showed how the network is being drawn in the OPNET Modeler.

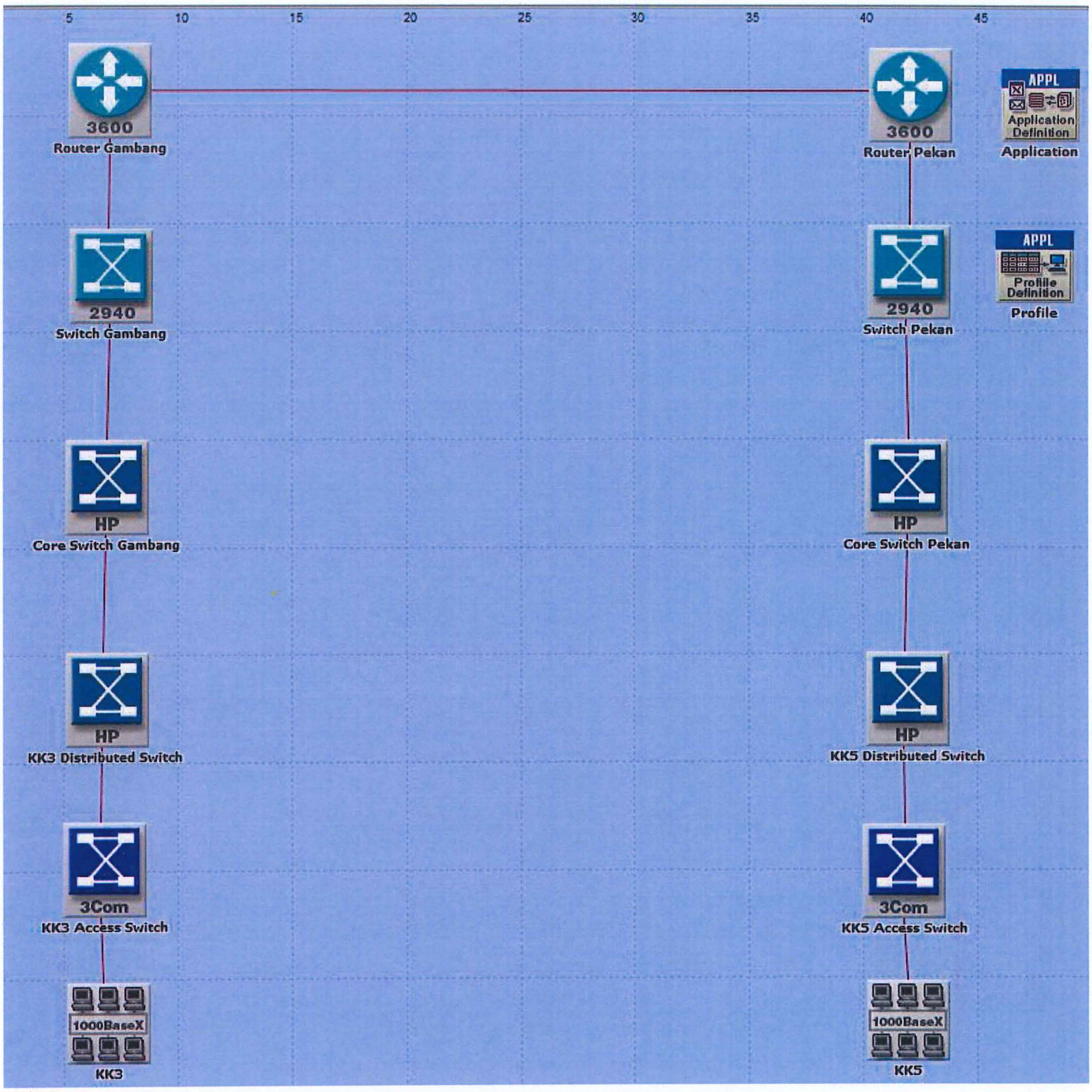
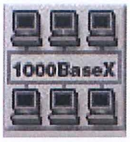
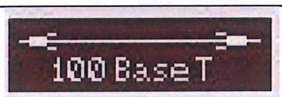
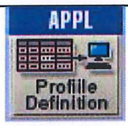


Figure 4.1: Design Drawn in OPNET Modeler

The table has list out the types of switches, routers and the cables used in the network topology.

Table 4.1: Description of Devices Used

Types of devices	Description
 1000 Base X Switched LAN	1000 Base X LAN represented a Gigabyte Ethernet LAN in a switched topology. It can contain any number of clients as well. In the scenario, this symbol represented the end users in KK3 and KK5 University Malaysia Pahang.
 Cisco Catalyst 2948G	This model represents the base configuration of the Catalyst 2948G switch. It has 2 types of interface which are 48-port autosensing 10/100Base T Ethernet and 2-port Gigabit Ethernet. In the scenario, this switch represented the access switch used by PTMK which placed between the routers and the core switch.
 HP ProCurve 4000M Ethernet Switch	This switch had been used as distributed switches in the scenario. It represents the distributed switch in both Gambang and Pekan.
 HP ProCurve 8000M Ethernet Switch	This switch had been used as Core switches in the scenario. It represents the Core switch in both Gambang and Pekan.
 3Com SSII 3900-24 Switch	This switches have 24 switched 10Base T/100Base T ports, an 1000 Base X Gigabit Ethernet port, and two optional one-port 1000BaseX Gigabit Ethernet modules. It is use as the Access switches which connect to the hostel.

 Cisco 3640 Router	In the scenario, this router is placed in the Gambang and Pekan. Both the routers is connected to a metro Ethernet which is 100Mbps provided by Tmnet.
 1000 Base X	The 1000 Base X duplex link represents an Ethernet connection operating at 1Gbps. It is used between the end user and the access switch and between the link of access switch and distributed switch. In addition, it also being used for the access switch connected to the router.
 10Gbps Ethernet	This link represents an Ethernet connection operating at 10Gbps. It is used between the distributed switch and the core switch.
 100 Base T	This link represents an Ethernet connection operating at 100Mbps. In this scenario, it is used between 2 routers.
 Application Definition	It is used to specify applications using available application types. In this topology, it is used to specify VoIP, HTTP and Video conferencing application.
 Profile Definition	It used to create user profiles. This user profiles can then be specify on different nodes in the network to generate application layer traffic.

4.3 SIMULATION SCENARIOS

Two different scenarios will be simulated in this research. The first one is the scenario with the current topology and setup without traffic and the second is the scenario with current topology and setup with the traffic of HTTP and Streaming. In both scenarios, Routing Information Protocol (RIP) is used by the router to route the incoming packets to the appropriate destination. There are 1600 workstations in KK3 LAN while for KK5 LAN, there are 1200 workstations. The number of VoIP users will start from 100, continue with 150, 200, 250, 300, and 350. The detailed configuration of each network element is discussed in detail in Appendix A.

4.4 TESTING PLAN

In the testing plan, the scenario will be run in OPNET Modeler to obtain the graphs to be analyzed in the next phase. The simulation needed to be accurate and precise. In this research, the numbers of users will be increased gradually as the bandwidth remains the same. The results will be collected in the table below.

Table 4.2: Table to Summarize the Results for Each Scenario

Scenario With/ Without Traffic			
Number of Users	Jitter	Packet Loss	End-to-end delay
100			
150			
200			
250			
300			
350			

CHAPTER 5

RESULTS AND DISCUSSION

5.0 INTRODUCTION

This chapter showed and discussed the relevant simulation results for the VoIP performance. The duration of OPNET simulation was set to 50 seconds due to memory limitation. The VoIP traffic started at 5 seconds after the simulation is initially started. There are two scenarios being simulated using OPNET Modeler. First scenario is to determine the maximum concurrent VoIP users without any traffic in the network. However, the second scenario is to determine the maximum concurrent VoIP users in a busy network. The topology of the network is shown in Figure 5.1.

The number of VoIP user was increased from 100 to 350 with an interval of 50 users for both scenarios. The VoIP calls were made from KK3 LAN to KK5 LAN. For the first scenario, the result was shown in red colour while for the second scenario, the result was shown in blue colour. The results of two scenarios were present in the same graph so that it is easy to see the difference. Results of both scenarios will be analyzed based on the performance indicator such as jitter, MOS, and end-to-end delay. Therefore, the scenario with traffic and without traffic will be analyzed under each performance indicator.

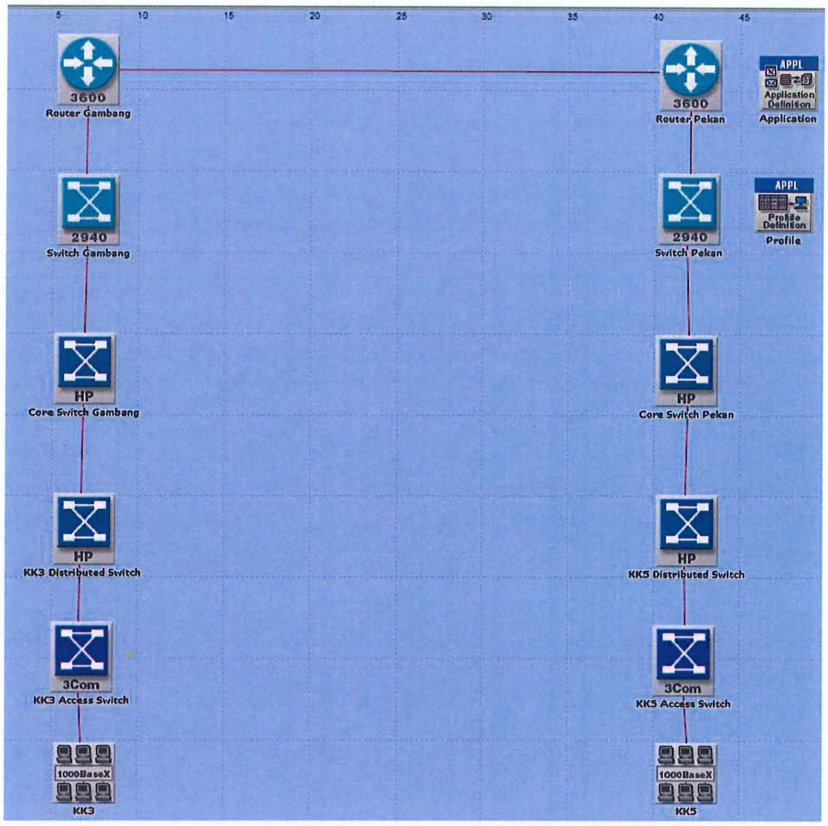


Figure 5.1: Simulation Network Topology

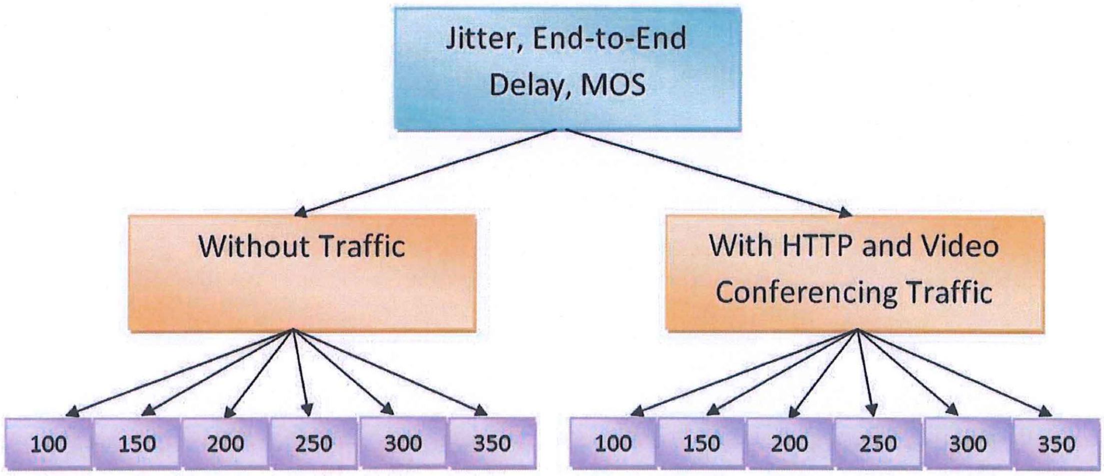
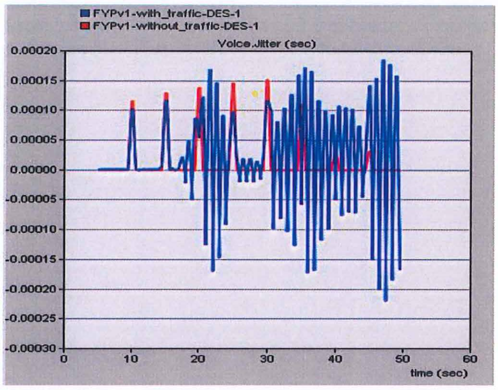


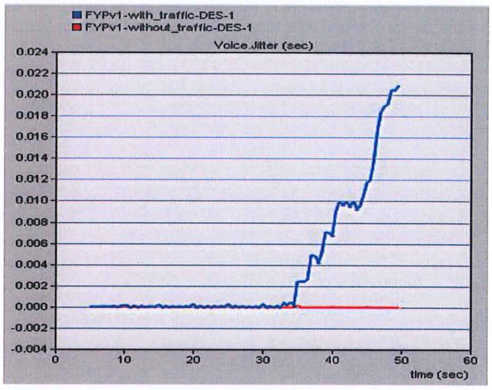
Figure 5.2: Testing Plan

5.1 JITTER

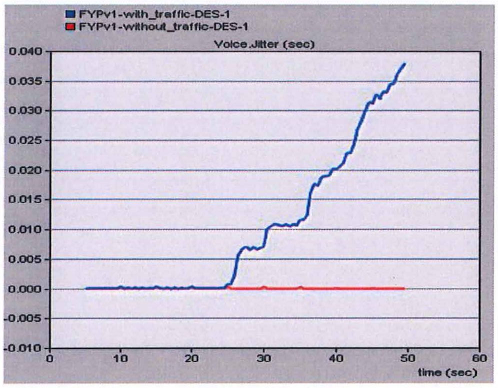
Jitter is the time different between the successively arriving packets. As discussed in Chapter 2, voice jitter can tolerate up to 0.075 seconds.



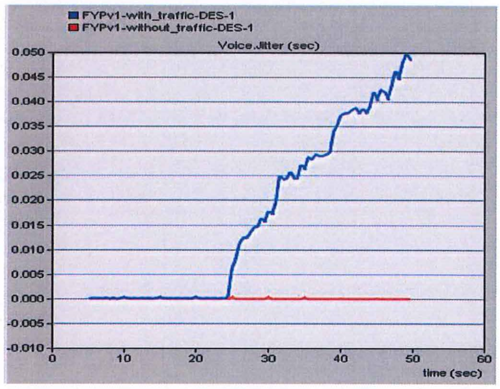
100 Clients



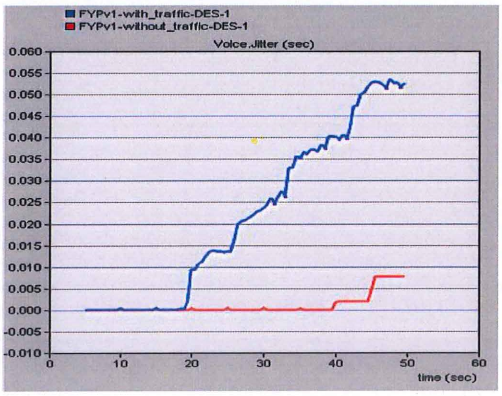
150 Clients



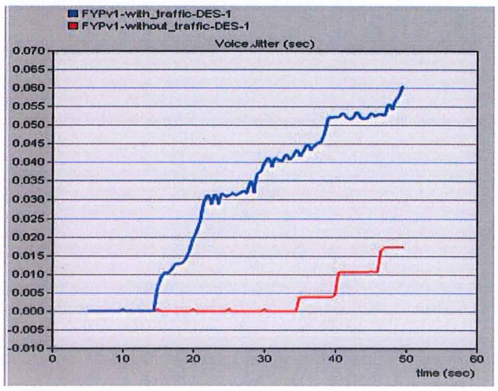
200 Clients



250 Clients



300 Clients



350 Clients

Figure 5.3: Results of Jitter for Both Scenarios

As shown in Figure 5.3, the voice jitter increased as the number of nodes increased for both scenarios.

For the topology without traffic, which represent red colour in the graph, the voice jitter start from 0.00015 with 100 clients and increased to 0.017 for 350 clients. The voice jitter is very small and unnoticeable for the number of clients within 100 to 250. However, when the number of clients increased to 300, the jitter started to increase after it simulates for 40 seconds. When the number of clients increased to 350, the jitter gradually increased after 35 seconds. Even though the value of jitter increased gradually for 350 clients, but it can still be accepted since the jitter can tolerate up to 0.075 seconds.

For the topology with traffic, which is in blue colour, the jitter increased from 0.00015 seconds to 0.060 seconds as the number of clients increased. The jitter is very small for 100 clients. When clients increased to 150, the jitter increased sharply after 35 seconds. As the number of clients increased, the time for jitter to go up sharply and reduce from 35 seconds to 25 seconds, 23 seconds, 20 seconds and 15 seconds. When there are 350 clients, the jitter reached 0.060 seconds which still can be accepted because it is below the maximum tolerate seconds which is 0.075 seconds.

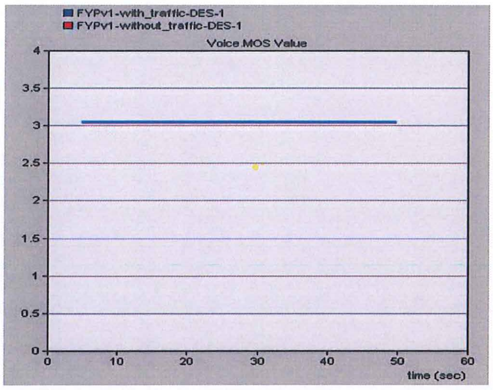
In a conclusion, the results showed that for both scenarios, the maximum number of VoIP users is more than 350 since the jitter is less than 0.075 seconds for 350 clients.

5.2 MOS VALUE

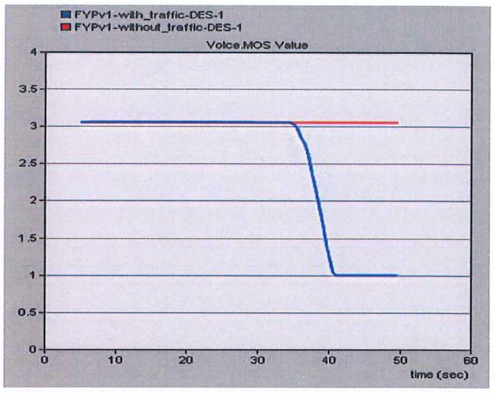
MOS is used to measure the quality of voice output that produced by the CODEC. MOS value is a number from 1 to 5. The MOS score are shown below.

Table 5.1: MOS Rating Table

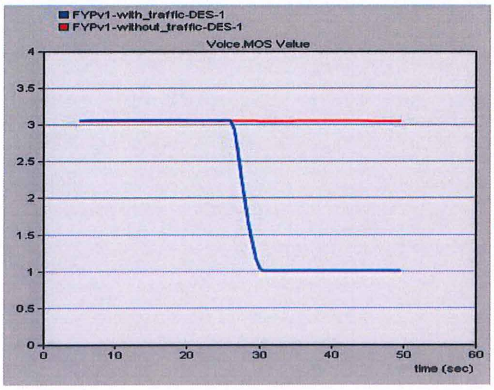
MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly Annoying
2	Poor	Annoying
1	Bad	Very Annoying



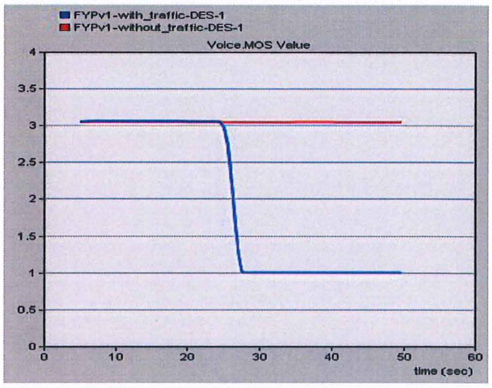
100 Clients



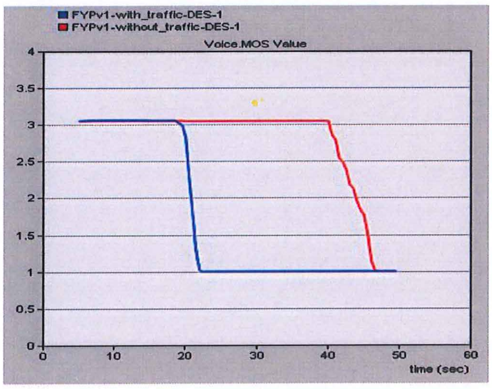
150 Clients



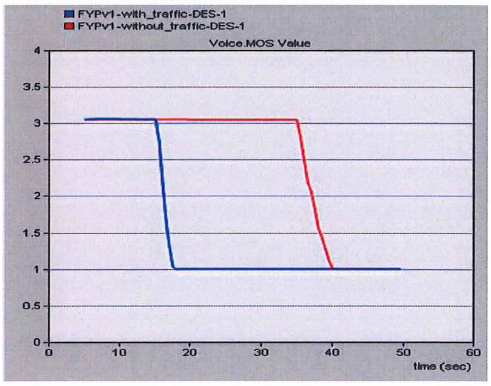
200 Clients



250 Clients



300 Clients



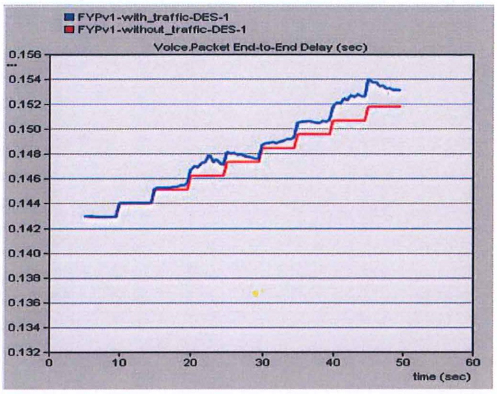
350 Clients

Figure 5.4: Results of MOS for Both Scenarios

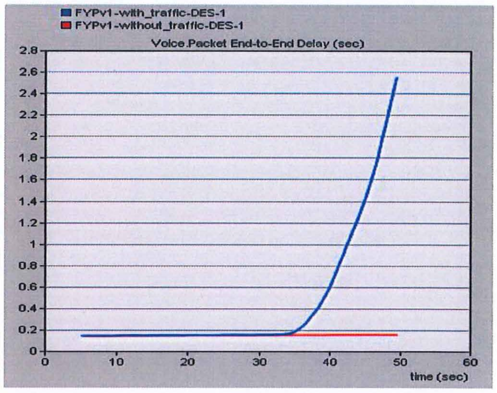
Based on Figure 5.4, we can see that the MOS value decreased for both scenarios as the number of clients increased. When 100 clients, both scenarios showed that the MOS value maintain above 3. When number of clients increased to 150, the MOS value for scenario without traffic maintained the same. However for scenario with traffic, the MOS value dropped dramatically and fell to 1 after 40 seconds. For 200 clients, the MOS value maintained for scenario without traffic but for scenario with traffic, the MOS value dropped dramatically and fell to 1 after 30 seconds. It clearly showed that the time for the MOS to maintain in this scenario is shorter as the clients increased. The graph is almost the same for 200 clients and 250 clients. When clients increased to 300, the scenario without traffic showed the MOS value started to drop dramatically after 40 seconds and it fells to MOS value 1 at about 46 seconds. For scenario with traffic, the MOS can maintain only 20 seconds and drops sharply to 1. The graph is almost the same for 350 clients. The only different is the time for the MOS to drop becomes shorter where the scenario without traffic is about 35 seconds and scenario with traffic is 15 seconds. In a conclusion, the scenario without traffic can support up to 250 clients since the MOS value maintain up to 250 clients while for scenario with traffic, it can only supports up to 100 clients due to the MOS value start to drops at 150 clients.

5.3 END-TO-END DELAY

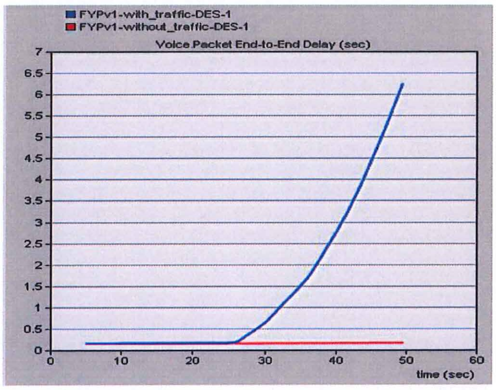
End-to-end delay is the delay of the voice packet during the journey through the network from one endpoint to another endpoint. The end-to-end delay for voice packet can tolerate within the range of 0.15 to 0.20 seconds.



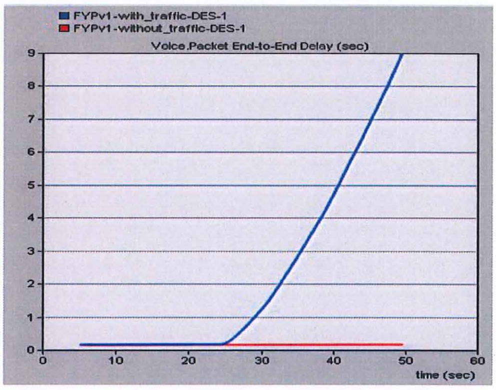
100 Clients



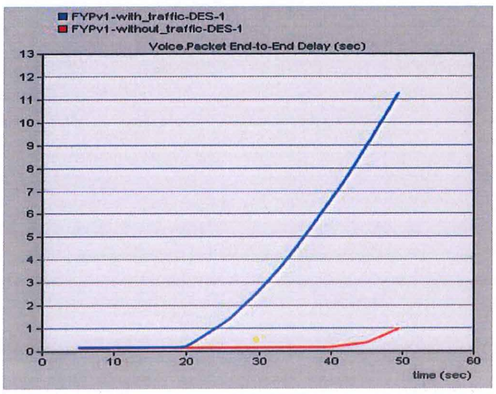
150 Clients



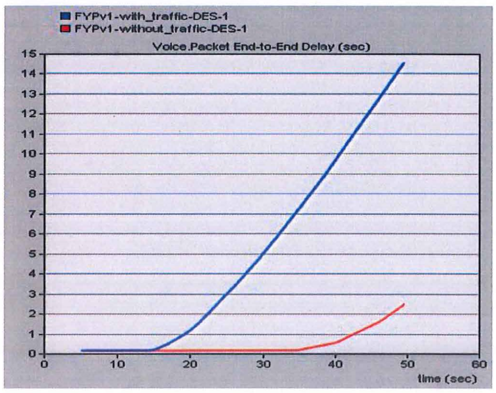
200 Clients



250 Clients



300 Clients



350 Clients

Figure 5.5: Results of End-to-End Delay for Both Scenarios

From the results in Figure 5.5, we can see that the number of end-to-end delay increased as the number of clients increased. For 100 clients, the end-to-end delay can be accepted for both scenarios since the value of end-to-end delay is in the range of 0.20 seconds. When the number of clients increased to 150, the scenario without traffic increased linearly but for the scenario with traffic, it increased dramatically at 34 seconds and the end-to-end delay reached 2.5 seconds. The value cannot be accepted since it already exceed the maximum tolerate range. Both the graph for 200 clients and 250 clients is almost the same with the graph for 150 clients. The only different is the time for scenario with traffic increased gradually become shorter. It means that for 200 and 250 clients, the scenario with traffic takes 25 seconds and it shoot up to 6 seconds and 9 seconds for end-to-end delay. Besides that, for scenario without traffic, the values of end-to-end delay increased linearly and still below the maximum tolerate range. For 300 clients, scenario without traffic started to increase dramatically after 40 seconds and it reached 1 second for end-to-end delay. However for scenario with traffic, it shoots up to 11 seconds for end-to-end delay in 20 seconds. The graph for 350 clients is almost the same with 300 clients. In a conclusion, the number of clients for scenario without traffic can be tolerate up to 250 clients while for scenario with traffic can only tolerate up to 100 clients due to the threshold for end-to-end delay is only 0.20 seconds.

CHAPTER 6

CONCLUSIONS

6.0 CONCLUSION

This thesis evaluated the performance analysis of VoIP network between KK3 and KK5 University Malaysia Pahang using OPNET simulator. Two types of scenario were conducted in the research which is scenario without traffic and scenario with traffic. Besides that, in this thesis, packet end-to-end delay, jitter and MOS are measured in both scenarios to determine the maximum number of VoIP users can be supported in the current topology.

As the number of VoIP clients increased, we can see that it has a significant impact on VoIP performance for both the scenarios. It is very significant in scenario with traffic. As we can see that the impact of increasing of VoIP clients in scenario with traffic is much greater than scenario without traffic.

For scenario without traffic, the maximum number of VoIP users based on jitter is more than 350 users. However, based on end-to-end delay, it shows that the VoIP users can support up to 250 users and based on MOS, it shows that the current topology can support up to 250 VoIP users. The Table 6.1 below summarize the results in this scenario.

According to Table 6.1, to have an excellent VoIP session, only 250 VoIP users can be supported at a time. Based on the topology, there is one bottleneck between 2 routers, therefore, the number of VoIP clients is less although the topology uses fiber in the network.

Table 6.1: Summary of Results in Scenario 1

Scenario Without Traffic			
Number of Clients	**Jitter**	**MOS**	**End-to-End delay**
100	Acceptable	Acceptable	Acceptable
150	Acceptable	Acceptable	Acceptable
200	Acceptable	Acceptable	Acceptable
250	Acceptable	Acceptable	Acceptable
300	Acceptable	Not Acceptable	Not Acceptable
350	Acceptable	Not Acceptable	Not Acceptable

For scenario with traffic, the maximum number of VoIP users based on jitter is more than 350 users. For end-to-end delay, it shows that the VoIP users can only be support up to 100 clients. Based on MOS, it shows that the current topology can support up to 100 VoIP users. The Table below summarize the results in this scenario. According to Table 6.2, we can see that the scenario with traffic can only support up to 100 VoIP clients.

Table 6.2: Summary of Results in Scenario2

Scenario With Traffic			
Number of Clients	**Jitter**	**MOS**	**End-to-End delay**
100	Acceptable	Acceptable	Acceptable
150	Acceptable	Not Acceptable	Not Acceptable
200	Acceptable	Not Acceptable	Not Acceptable
250	Acceptable	Not Acceptable	Not Acceptable
300	Acceptable	Not Acceptable	Not Acceptable
350	Acceptable	Not Acceptable	Not Acceptable

In a nutshell, the maximum number of concurrent VoIP user for scenario without traffic is 250. On another hand, the maximum number of concurrent VoIP user for scenario with traffic is 100.

6.1 LIMITATIONS OF THE RESEARCH

There are some limitations in this research. First and foremost, not all the models of the nodes are supported in OPNET Modeler 14.5. For example, there are some switches that do not include in the object palette of the OPNET. Besides that, although simulation performed quite well under OPNET Modeler, and the simulator is very close to the reality, but the simulation still not in the reality.

6.2 RECOMMENDATION AND FUTURE RESEARCH

Many residents live in KK3 and KK5 hostel. Therefore, with the current topology, it cannot support all the residents in the hostel used VoIP at the same time. This condition can be improved by increasing the bottleneck link between the two routers. Moreover, this research only considers VoIP traffic, HTTP traffic and Video Conferencing only. In future studies, more realistic traffic applications such as background traffic, FTP, and Email can be considered. Besides that, in future research, the VoIP traffic can be tested in the wireless network.

6.3 CONTRIBUTION

In this research, the major factors that affect VoIP quality such as jitter, packet end-to-end delays are measured by the simulator. The simulation results in this research can help the university understand how well the VoIP will perform on current topology. In addition, it helps researchers and network engineer to design a network for VoIP deployment.

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APPENDICES

APPENDIX A

CONFIGURING VOICE IN OPNET MODELER

The experimental network is setup in OPNET modeler. The network nodes can be accessed under the object palette as shown in Figure A.1. The topology is setup by dragging and dropping the network nodes to the project editor in OPNET Modeler.

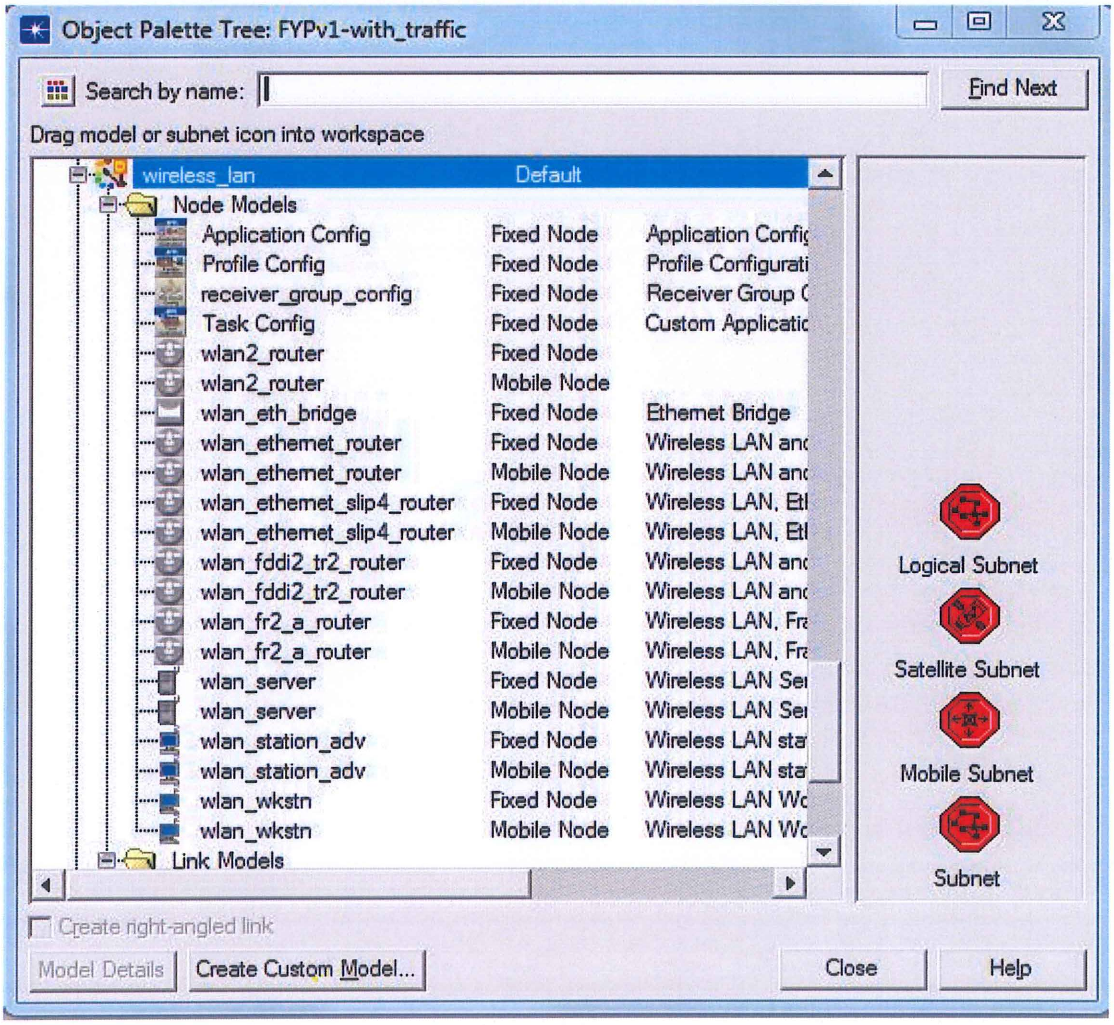


Figure A.1: Object Palette

Configuring Application Definition Object

The application definition object defines a list of applications. OPNET has predefined applications such as Database, Email, FTP, HTTP, Print, Remote login, Video Conferencing and Voice. We can edit the specification of the each application. Figure A.2 to A.4 shows the setting for creating the applications in the application definition.

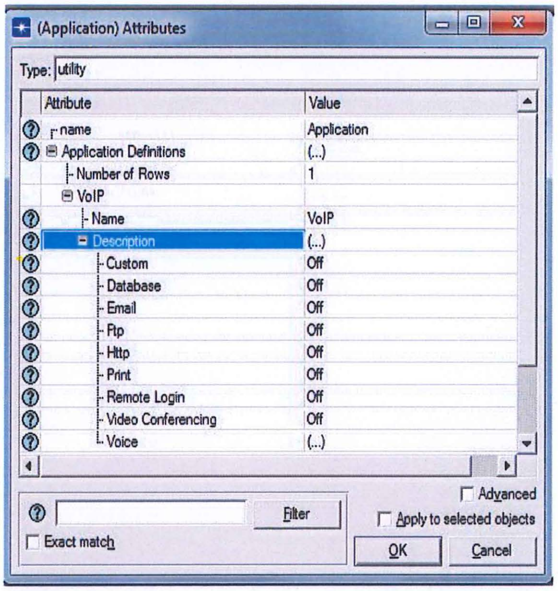


Figure A.2: This is the application configuration for scenario without traffic. Only the VoIP traffic is configured.

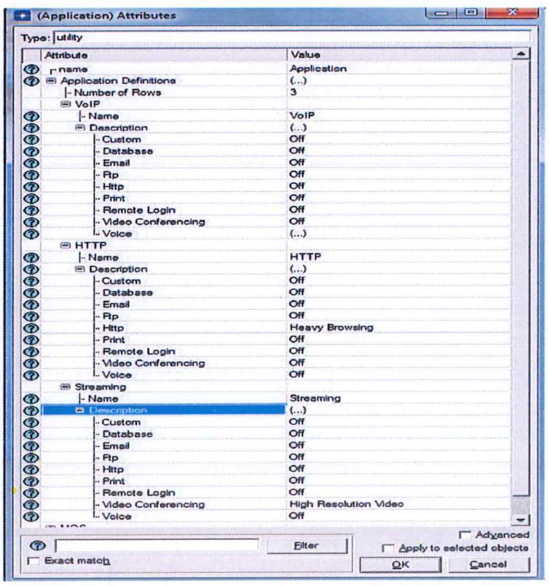
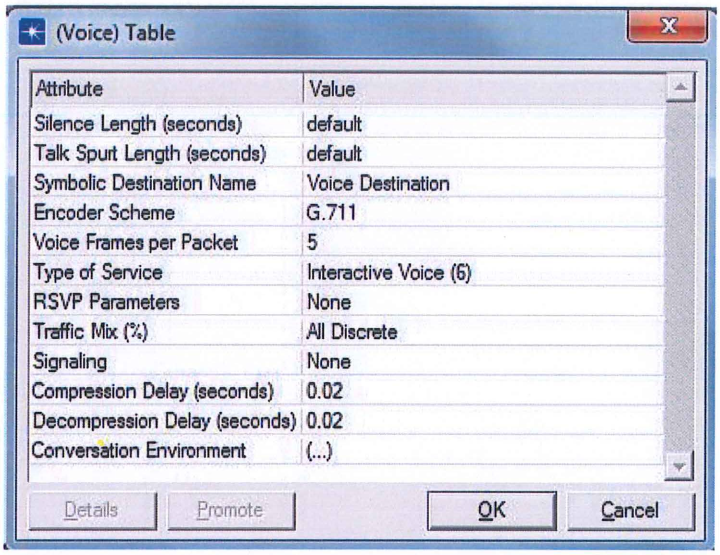


Figure A.3: This is the application configuration for scenario with traffic.

For this scenario, VoIP traffic, HTTP and Video Conferencing is configured. HTTP traffic is configured as heavy browsing and for Video Conferencing, it is configured as High Resolution Video.



Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.711
Voice Frames per Packet	5
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	(...)

Figure A.4: This figure shows the voice table configuration.

We can see that G.711 codec is selected from a list of varying CODECs. The Type of service is selected as Interactive voice. The silence length and talk spurt length is set to default. The symbolic definition name is set as Voice Destination.

Configuring Profile Definition Object

Profile definition used to create user profiles. These user profiles can then be specified on different nodes in the network to generate application layer traffic. A profile consists of a list of applications. Every profile has it configurable parameters. The parameters will be discussed as follow.

1) **Operation mode**: It defines how the application will start. If it set to serial, the application can start after each other. If set to simultaneous, the applications can start all at the same time.
2) **Start Time**: Defines when during the simulation the profile session will start.

3) **Duration:** Defines the maximum amount of time allowed for the profile before it ends. When "End of simulation" is selected, the profile is allowed to continue indefinitely till the simulation ends.
4) **Repeatability**: It specifies the parameters used to repeat the execution of this profile.

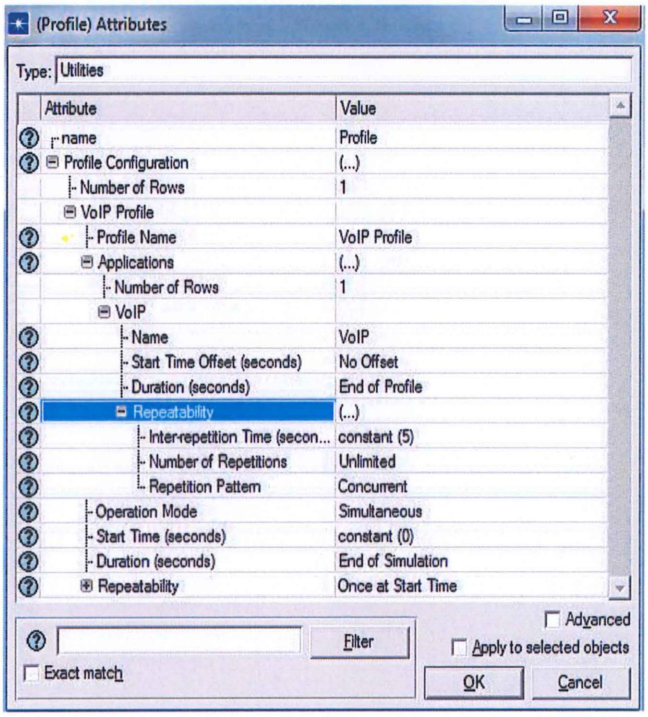


Figure A.5: This is the profile configuration for scenario without traffic. In the VoIP profile, it only has VoIP application.

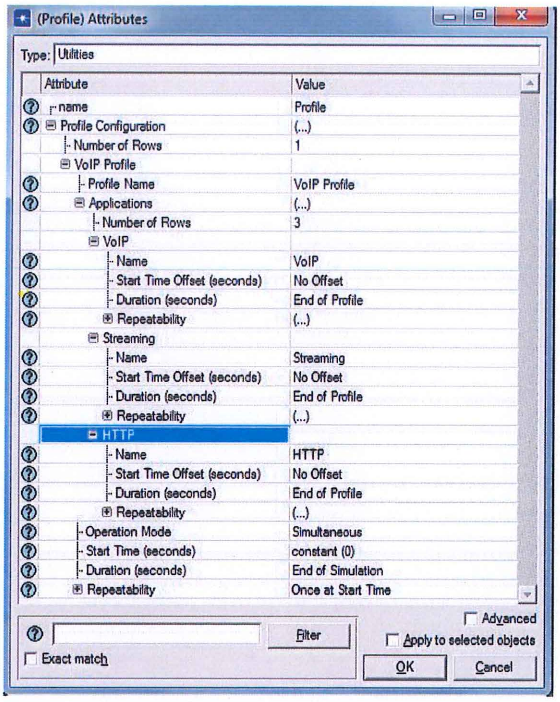


Figure A.6: This is the profile configuration for scenario with traffic. In the VoIP profile, it contains VoIP application, HTTP application and Streaming.

APPENDIX B

STATISTICAL DATA OF THE RESULTS

Without Traffic

Conversion of Packet Delay Variation:

E = x10

E-08 = x10^{-8}

*** 1.65E-07 = 1.65 x 10^{-7} = 0.000000165**

Table B.1: Statistical data of 100 VoIP users without traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	#N/A	#N/A	#N/A	#N/A
5.5	0	3.055426	1.65E-07	0.142906
6	0	3.055426	1.82E-07	0.142893
6.5	0	3.055426	1.81E-07	0.142893
7	0	3.055426	1.81E-07	0.142893
7.5	0	3.055426	1.81E-07	0.142893
8	0	3.055426	1.81E-07	0.142893
8.5	0	3.055426	1.81E-07	0.142893
9	0	3.055426	1.81E-07	0.142893
9.5	0	3.055426	1.81E-07	0.142893
10	0.000117	3.055426	2.86E-07	0.144009
10.5	0	3.055426	4.34E-07	0.144009
11	0	3.055426	5.32E-07	0.144009
11.5	0	3.055426	5.95E-07	0.144009
12	0	3.055426	6.36E-07	0.144009
12.5	0	3.055426	6.62E-07	0.144009
13	0	3.055426	6.77E-07	0.144009
13.5	0	3.055426	6.85E-07	0.144009

14	0	3.055426	6.88E-07	0.144009
14.5	0	3.055426	6.88E-07	0.144009
15	0.00013	3.055426	7.4E-07	0.145123
15.5	0	3.055426	9.06E-07	0.145123
16	0	3.055426	1.04E-06	0.145123
16.5	0	3.055426	1.14E-06	0.145123
17	0	3.055426	1.22E-06	0.145123
17.5	0	3.055426	1.28E-06	0.145123
18	0	3.055426	1.32E-06	0.145123
18.5	0	3.055426	1.36E-06	0.145123
19	0	3.055426	1.39E-06	0.145123
19.5	0	3.055426	1.41E-06	0.145123
20	0.000139	3.055426	1.44E-06	0.146236
20.5	0	3.055426	1.64E-06	0.146236
21	0	3.055426	1.8E-06	0.146236
21.5	0	3.055426	1.93E-06	0.146236
22	0	3.055426	2.04E-06	0.146236
22.5	0	3.055426	2.13E-06	0.146236
23	0	3.055426	2.2E-06	0.146236
23.5	0	3.055426	2.26E-06	0.146236
24	0	3.055426	2.32E-06	0.146236
24.5	0	3.055426	2.36E-06	0.146236
25	0.000147	3.055426	2.37E-06	0.147348
25.5	0	3.055426	2.6E-06	0.147348
26	0	3.055426	2.79E-06	0.147348
26.5	0	3.055426	2.95E-06	0.147348
27	0	3.055426	3.09E-06	0.147348
27.5	0	3.055426	3.21E-06	0.147348
28	0	3.055426	3.31E-06	0.147348
28.5	0	3.055426	3.39E-06	0.147348
29	0	3.055426	3.47E-06	0.147348
29.5	0	3.055426	3.53E-06	0.147348
30	0.000153	3.055426	3.53E-06	0.14846
30.5	0	3.055426	3.79E-06	0.14846
31	0	3.055426	4.01E-06	0.14846
31.5	0	3.055426	4.2E-06	0.14846
32	0	3.055426	4.36E-06	0.14846
32.5	0	3.055426	4.5E-06	0.14846
33	0	3.055426	4.63E-06	0.14846
33.5	0	3.055426	4.74E-06	0.14846
34	0	3.055426	4.83E-06	0.14846
34.5	0	3.055426	4.92E-06	0.14846
35	0.000111	3.055426	4.46E-06	0.149571
35.5	0	3.055426	4.7E-06	0.149572
36	0	3.055426	4.91E-06	0.149571
36.5	0	3.055426	5.1E-06	0.149571
37	0	3.055426	5.27E-06	0.149571
37.5	0	3.055426	5.43E-06	0.149571
38	0	3.055426	5.57E-06	0.149571
38.5	0	3.055426	5.7E-06	0.149571
39	0	3.055426	5.82E-06	0.149571
39.5	0	3.055426	5.93E-06	0.149571
40	3.26E-05	3.055426	5.4E-06	0.150685
40.5	0	3.055426	5.51E-06	0.150683
41	0	3.055426	5.61E-06	0.150683
41.5	0	3.055426	5.7E-06	0.150683
42	0	3.055426	5.78E-06	0.150683
42.5	0	3.055426	5.85E-06	0.150683
43	0	3.055426	5.91E-06	0.150683
43.5	0	3.055426	5.97E-06	0.150683
44	0	3.055426	6.02E-06	0.150683
44.5	0	3.055426	6.07E-06	0.150683
45	3.28E-05	3.055426	5.54E-06	0.151796
45.5	0	3.055426	5.6E-06	0.151794
46	0	3.055426	5.66E-06	0.151794
46.5	0	3.055426	5.71E-06	0.151794

47	0	3.055426	5.75E-06	0.151794
47.5	0	3.055426	5.79E-06	0.151794
48	0	3.055426	5.82E-06	0.151794
48.5	0	3.055426	5.86E-06	0.151794
49	0	3.055426	5.88E-06	0.151794
49.5	0	3.055426	5.91E-06	0.151794
50	#N/A	#N/A	#N/A	#N/A

Table B.2: Statistical data of 150 VoIP users without traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	#N/A	#N/A	#N/A	#N/A
5.5	0	3.055426	4.15E-07	0.144044
6	0	3.055426	4.49E-07	0.14403
6.5	0	3.055426	4.48E-07	0.14403
7	0	3.055426	4.48E-07	0.14403
7.5	0	3.055426	4.48E-07	0.14403
8	0	3.055426	4.48E-07	0.14403
8.5	0	3.055426	4.48E-07	0.14403
9	0	3.055426	4.48E-07	0.14403
9.5	0	3.055426	4.48E-07	0.14403
10	0.000177	3.055426	7.46E-07	0.145711
10.5	0	3.055426	1.08E-06	0.145711
11	0	3.055426	1.29E-06	0.145711
11.5	0	3.055426	1.43E-06	0.145711
12	0	3.055426	1.52E-06	0.145711
12.5	0	3.055426	1.58E-06	0.145711
13	0	3.055426	1.61E-06	0.145711
13.5	0	3.055426	1.63E-06	0.145711
14	0	3.055426	1.63E-06	0.145711
14.5	0	3.055426	1.63E-06	0.145711
15	0.000198	3.055426	1.83E-06	0.14739
15.5	0	3.055426	2.2E-06	0.14739
16	0	3.055426	2.49E-06	0.14739
16.5	0	3.055426	2.72E-06	0.14739
17	0	3.055426	2.89E-06	0.14739
17.5	0	3.055426	3.03E-06	0.14739
18	0	3.055426	3.13E-06	0.14739
18.5	0	3.055426	3.21E-06	0.14739
19	0	3.055426	3.27E-06	0.14739
19.5	0	3.055426	3.32E-06	0.14739
20	0.000212	3.055426	3.47E-06	0.149069
20.5	0	3.055426	3.92E-06	0.149069
21	0	3.055426	4.27E-06	0.149069
21.5	0	3.055426	4.57E-06	0.149069
22	0	3.055426	4.81E-06	0.149069
22.5	0	3.055426	5.01E-06	0.149069
23	0	3.055426	5.18E-06	0.149069
23.5	0	3.055426	5.31E-06	0.149069
24	0	3.055426	5.43E-06	0.149069
24.5	0	3.055426	5.52E-06	0.149069
25	0.000224	3.055426	5.63E-06	0.150748
25.5	0	3.055426	6.16E-06	0.150748

26	0	3.055426	6.58E-06	0.150748
26.5	0	3.055426	6.94E-06	0.150748
27	0	3.055426	7.25E-06	0.150748
27.5	0	3.055426	7.51E-06	0.150748
28	0	3.055426	7.73E-06	0.150748
28.5	0	3.055426	7.92E-06	0.150748
29	0	3.055426	8.08E-06	0.150748
29.5	0	3.055426	8.22E-06	0.150748
30	0.000233	3.055426	8.28E-06	0.152426
30.5	0	3.055426	8.9E-06	0.152426
31	0	3.055426	9.39E-06	0.152426
31.5	0	3.055426	9.81E-06	0.152426
32	0	3.055426	1.02E-05	0.152426
32.5	0	3.055426	1.05E-05	0.152426
33	0	3.055426	1.08E-05	0.152426
33.5	0	3.055426	1.1E-05	0.152426
34	0	3.055426	1.12E-05	0.152426
34.5	0	3.055426	1.14E-05	0.152426
35	0.000168	3.055426	1.03E-05	0.154121
35.5	0	3.055426	1.09E-05	0.154119
36	0	3.055426	1.13E-05	0.154119
36.5	0	3.055426	1.18E-05	0.154119
37	0	3.055426	1.22E-05	0.154119
37.5	0	3.055426	1.25E-05	0.154119
38	0	3.055426	1.28E-05	0.154119
38.5	0	3.055426	1.31E-05	0.154119
39	0	3.055426	1.34E-05	0.154119
39.5	0	3.055426	1.36E-05	0.154119
40	5.24E-05	3.055426	1.24E-05	0.155825
40.5	0	3.055426	1.27E-05	0.155823
41	0	3.055426	1.29E-05	0.155823
41.5	0	3.055426	1.31E-05	0.155823
42	0	3.055426	1.33E-05	0.155823
42.5	0	3.055426	1.34E-05	0.155823
43	0	3.055426	1.36E-05	0.155823
43.5	0	3.055426	1.37E-05	0.155823
44	0	3.055426	1.38E-05	0.155823
44.5	0	3.055426	1.39E-05	0.155823
45	5.25E-05	3.055426	1.27E-05	0.157525
45.5	0	3.055426	1.29E-05	0.157523
46	0	3.055426	1.3E-05	0.157523
46.5	0	3.055426	1.31E-05	0.157523
47	0	3.055426	1.32E-05	0.157523
47.5	0	3.055426	1.33E-05	0.157523
48	0	3.055426	1.34E-05	0.157523
48.5	0	3.055426	1.34E-05	0.157523
49	0	3.055426	1.35E-05	0.157523
49.5	0	3.055426	1.36E-05	0.157523
50	#N/A	#N/A	#N/A	#N/A

Table B.3: Statistical data of 200 VoIP users without traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A

5	#N/A	#N/A	#N/A	#N/A
5.5	0	3.055426	7.87E-07	0.145179
6	0	3.055426	8.45E-07	0.145165
6.5	0	3.055426	8.43E-07	0.145165
7	0	3.055426	8.42E-07	0.145165
7.5	0	3.055426	8.42E-07	0.145165
8	0	3.055426	8.41E-07	0.145165
8.5	0	3.055426	8.41E-07	0.145165
9	0	3.055426	8.41E-07	0.145165
9.5	0	3.055426	8.41E-07	0.145165
10	0.000236	3.055426	1.44E-06	0.147412
10.5	0	3.055426	2.02E-06	0.147412
11	0	3.055426	2.4E-06	0.147412
11.5	0	3.055426	2.65E-06	0.147412
12	0	3.055426	2.81E-06	0.147412
12.5	0	3.055426	2.91E-06	0.147412
13	0	3.055426	2.97E-06	0.147412
13.5	0	3.055426	3E-06	0.147412
14	0	3.055426	3.01E-06	0.147412
14.5	0	3.055426	3E-06	0.147412
15	0.000265	3.055426	3.42E-06	0.149657
15.5	0	3.055426	4.09E-06	0.149657
16	0	3.055426	4.6E-06	0.149657
16.5	0	3.055426	5E-06	0.149657
17	0	3.055426	5.31E-06	0.149657
17.5	0	3.055426	5.55E-06	0.149657
18	0	3.055426	5.73E-06	0.149657
18.5	0	3.055426	5.87E-06	0.149657
19	0	3.055426	5.98E-06	0.149657
19.5	0	3.055426	6.06E-06	0.149657
20	0.000285	3.055426	6.4E-06	0.151902
20.5	0	3.055426	7.2E-06	0.151902
21	0	3.055426	7.83E-06	0.151902
21.5	0	3.055426	8.36E-06	0.151902
22	0	3.055426	8.79E-06	0.151902
22.5	0	3.055426	9.14E-06	0.151902
23	0	3.055426	9.43E-06	0.151902
23.5	0	3.055426	9.67E-06	0.151902
24	0	3.055426	9.87E-06	0.151902
24.5	0	3.055426	1E-05	0.151902
25	0.000302	3.055426	1.03E-05	0.15416
25.5	0	3.055426	1.12E-05	0.15416
26	0	3.055426	1.2E-05	0.15416
26.5	0	3.055426	1.26E-05	0.15416
27	0	3.055426	1.32E-05	0.15416
27.5	0	3.055426	1.36E-05	0.15416
28	0	3.055426	1.4E-05	0.15416
28.5	0	3.055426	1.44E-05	0.15416
29	0	3.055426	1.47E-05	0.15416
29.5	0	3.055426	1.49E-05	0.15416
30	0.000317	3.050682	1.51E-05	0.15644
30.5	0	3.050682	1.62E-05	0.15644
31	0	3.050682	1.71E-05	0.15644
31.5	0	3.050682	1.78E-05	0.15644
32	0	3.050682	1.85E-05	0.15644
32.5	0	3.050682	1.9E-05	0.15644
33	0	3.050682	1.95E-05	0.15644
33.5	0	3.050682	2E-05	0.15644
34	0	3.050682	2.03E-05	0.15644
34.5	0	3.050682	2.07E-05	0.15644
35	0.000227	3.055426	1.86E-05	0.15867
35.5	0	3.055426	1.97E-05	0.158712
36	0	3.055426	2.05E-05	0.158712
36.5	0	3.055426	2.13E-05	0.158712
37	0	3.055426	2.19E-05	0.158712
37.5	0	3.055426	2.26E-05	0.158712

38	0	3.055426	2.31E-05	0.158712
38.5	0	3.055426	2.37E-05	0.158712
39	0	3.055426	2.41E-05	0.158712
39.5	0	3.055426	2.45E-05	0.158712
40	7.28E-05	3.055426	2.26E-05	0.160878
40.5	0	3.055426	2.28E-05	0.161021
41	0	3.055426	2.32E-05	0.161021
41.5	0	3.055426	2.36E-05	0.161021
42	0	3.055426	2.39E-05	0.161021
42.5	0	3.055426	2.42E-05	0.161021
43	0	3.055426	2.45E-05	0.161021
43.5	0	3.055426	2.47E-05	0.161021
44	0	3.055426	2.49E-05	0.161021
44.5	0	3.055426	2.51E-05	0.161021
45	7.15E-05	3.055426	2.31E-05	0.163183
45.5	0	3.050682	2.32E-05	0.163326
46	0	3.050682	2.34E-05	0.163326
46.5	0	3.050682	2.36E-05	0.163326
47	0	3.050682	2.38E-05	0.163326
47.5	0	3.050682	2.4E-05	0.163326
48	0	3.050682	2.42E-05	0.163326
48.5	0	3.050682	2.43E-05	0.163326
49	0	3.050682	2.44E-05	0.163326
49.5	0	3.050682	2.45E-05	0.163326
50	#N/A	#N/A	#N/A	#N/A

Table B.4: Statistical data of 250 VoIP users without traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	#N/A	#N/A	#N/A	#N/A
5.5	0	3.055426	1.28E-06	0.146314
6	0	3.055426	1.37E-06	0.1463
6.5	0	3.055426	1.37E-06	0.1463
7	0	3.055426	1.37E-06	0.1463
7.5	0	3.055426	1.36E-06	0.1463
8	0	3.055426	1.36E-06	0.1463
8.5	0	3.055426	1.36E-06	0.1463
9	0	3.055426	1.36E-06	0.1463
9.5	0	3.055426	1.36E-06	0.1463
10	0.000296	3.055426	2.37E-06	0.149113
10.5	0	3.055426	3.28E-06	0.149113
11	0	3.055426	3.87E-06	0.149113
11.5	0	3.055426	4.26E-06	0.149113
12	0	3.055426	4.5E-06	0.149113
12.5	0	3.055426	4.66E-06	0.149113
13	0	3.055426	4.74E-06	0.149113
13.5	0	3.055426	4.79E-06	0.149113
14	0	3.055426	4.8E-06	0.149113
14.5	0	3.055426	4.79E-06	0.149113
15	0.000333	3.055426	5.52E-06	0.151924
15.5	0	3.055426	6.56E-06	0.151924
16	0	3.055426	7.37E-06	0.151924
16.5	0	3.055426	7.99E-06	0.151924

17	0	3.055426	8.47E-06	0.151924
17.5	0	3.055426	8.84E-06	0.151924
18	0	3.055426	9.13E-06	0.151924
18.5	0	3.055426	9.34E-06	0.151924
19	0	3.055426	9.51E-06	0.151924
19.5	0	3.055426	9.62E-06	0.151924
20	0.000359	3.055426	1.02E-05	0.154751
20.5	0	3.055426	1.15E-05	0.154751
21	0	3.055426	1.25E-05	0.154751
21.5	0	3.055426	1.33E-05	0.154751
22	0	3.055426	1.4E-05	0.154751
22.5	0	3.055426	1.45E-05	0.154751
23	0	3.055426	1.5E-05	0.154751
23.5	0	3.055426	1.53E-05	0.154751
24	0	3.055426	1.57E-05	0.154751
24.5	0	3.055426	1.59E-05	0.154751
25	0.000381	3.050682	1.64E-05	0.157599
25.5	0	3.050682	1.79E-05	0.157599
26	0	3.050682	1.91E-05	0.157599
26.5	0	3.050682	2.01E-05	0.157599
27	0	3.050682	2.09E-05	0.157599
27.5	0	3.050682	2.16E-05	0.157599
28	0	3.050682	2.23E-05	0.157599
28.5	0	3.050682	2.28E-05	0.157599
29	0	3.050682	2.32E-05	0.157599
29.5	0	3.050682	2.36E-05	0.157599
30	0.000403	3.050682	2.36E-05	0.160308
30.5	0	3.050682	2.56E-05	0.160449
31	0	3.050682	2.7E-05	0.160449
31.5	0	3.050682	2.82E-05	0.160449
32	0	3.050682	2.92E-05	0.160449
32.5	0	3.050682	3.01E-05	0.160449
33	0	3.050682	3.09E-05	0.160449
33.5	0	3.050682	3.15E-05	0.160449
34	0	3.050682	3.21E-05	0.160449
34.5	0	3.050682	3.26E-05	0.160449
35	0.000285	3.055426	2.91E-05	0.163116
35.5	0	3.055426	3.1E-05	0.163285
36	0	3.055426	3.23E-05	0.163284
36.5	0	3.055426	3.35E-05	0.163284
37	0	3.055426	3.46E-05	0.163284
37.5	0	3.055426	3.56E-05	0.163284
38	0	3.055426	3.65E-05	0.163284
38.5	0	3.055426	3.73E-05	0.163284
39	0	3.055426	3.8E-05	0.163284
39.5	0	3.055426	3.86E-05	0.163284
40	8.88E-05	3.055426	3.56E-05	0.165981
40.5	0	3.050682	3.59E-05	0.166156
41	0	3.050682	3.65E-05	0.166156
41.5	0	3.050682	3.71E-05	0.166156
42	0	3.050682	3.76E-05	0.166156
42.5	0	3.050682	3.81E-05	0.166156
43	0	3.050682	3.85E-05	0.166156
43.5	0	3.050682	3.88E-05	0.166156
44	0	3.050682	3.92E-05	0.166156
44.5	0	3.050682	3.95E-05	0.166156
45	8.93E-05	3.050682	3.64E-05	0.168845
45.5	0	3.050682	3.65E-05	0.169022
46	0	3.050682	3.68E-05	0.169022
46.5	0	3.050682	3.71E-05	0.169022
47	0	3.050682	3.74E-05	0.169022
47.5	0	3.050682	3.77E-05	0.169022
48	0	3.050682	3.79E-05	0.169022
48.5	0	3.050682	3.81E-05	0.169022
49	0	3.050682	3.83E-05	0.169022
49.5	0	3.050682	3.85E-05	0.169022

50	#N/A	#N/A	#N/A	#N/A

Table B.5: Statistical data of 300 VoIP users without traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	#N/A	#N/A	#N/A	#N/A
5.5	0	3.055426	1.9E-06	0.147448
6	0	3.055426	2.03E-06	0.147434
6.5	0	3.055426	2.02E-06	0.147434
7	0	3.055426	2.02E-06	0.147434
7.5	0	3.055426	2.02E-06	0.147434
8	0	3.055426	2.02E-06	0.147434
8.5	0	3.055426	2.02E-06	0.147434
9	0	3.055426	2.01E-06	0.147434
9.5	0	3.055426	2.01E-06	0.147434
10	0.000356	3.055426	3.53E-06	0.150813
10.5	0	3.055426	4.84E-06	0.150813
11	0	3.055426	5.69E-06	0.150813
11.5	0	3.055426	6.24E-06	0.150813
12	0	3.055426	6.6E-06	0.150813
12.5	0	3.055426	6.82E-06	0.150813
13	0	3.055426	6.94E-06	0.150813
13.5	0	3.055426	7.01E-06	0.150813
14	0	3.055426	7.02E-06	0.150813
14.5	0	3.055426	7.01E-06	0.150813
15	0.000401	3.055426	8.13E-06	0.154199
15.5	0	3.055426	9.64E-06	0.154199
16	0	3.055426	1.08E-05	0.154199
16.5	0	3.055426	1.17E-05	0.154199
17	0	3.055426	1.24E-05	0.154199
17.5	0	3.055426	1.29E-05	0.154199
18	0	3.055426	1.33E-05	0.154199
18.5	0	3.055426	1.36E-05	0.154199
19	0	3.055426	1.39E-05	0.154199
19.5	0	3.055426	1.4E-05	0.154199
20	0.000434	3.050682	1.5E-05	0.15761
20.5	0	3.050682	1.68E-05	0.15761
21	0	3.050682	1.82E-05	0.15761
21.5	0	3.050682	1.94E-05	0.15761
22	0	3.050682	2.04E-05	0.15761
22.5	0	3.050682	2.12E-05	0.15761
23	0	3.050682	2.18E-05	0.15761
23.5	0	3.050682	2.24E-05	0.15761
24	0	3.050682	2.28E-05	0.15761
24.5	0	3.050682	2.32E-05	0.15761
25	0.000465	3.050682	2.36E-05	0.160858
25.5	0	3.050682	2.6E-05	0.161027
26	0	3.050682	2.77E-05	0.161027
26.5	0	3.050682	2.92E-05	0.161027
27	0	3.050682	3.04E-05	0.161027
27.5	0	3.050682	3.15E-05	0.161027
28	0	3.050682	3.23E-05	0.161027
28.5	0	3.050682	3.31E-05	0.161027

29	0	3.050682	3.37E-05	0.161027
29.5	0	3.050682	3.43E-05	0.161027
30	0.000486	3.050682	3.44E-05	0.16425
30.5	0	3.050682	3.72E-05	0.164448
31	0	3.050682	3.92E-05	0.164448
31.5	0	3.050682	4.1E-05	0.164448
32	0	3.050682	4.24E-05	0.164448
32.5	0	3.050682	4.37E-05	0.164448
33	0	3.050682	4.48E-05	0.164448
33.5	0	3.050682	4.58E-05	0.164448
34	0	3.050682	4.66E-05	0.164448
34.5	0	3.050682	4.74E-05	0.164448
35	0.000342	3.050682	4.27E-05	0.167651
35.5	0	3.050682	4.49E-05	0.167852
36	0	3.050682	4.68E-05	0.167852
36.5	0	3.050682	4.85E-05	0.167852
37	0	3.050682	5.01E-05	0.167852
37.5	0	3.050682	5.15E-05	0.167852
38	0	3.050682	5.28E-05	0.167852
38.5	0	3.050682	5.39E-05	0.167852
39	0	3.050682	5.5E-05	0.167852
39.5	0	3.050682	5.59E-05	0.167852
40	0.001826	3.050682	5.94E-05	0.179323
40.5	0.001968	2.869623	7.41E-05	0.197997
41	0.001968	2.792821	0.000121	0.216748
41.5	0.001968	2.550187	0.000229	0.23643
42	0.001968	2.497787	0.000373	0.254363
42.5	0.001968	2.391751	0.000583	0.273825
43	0.001968	2.186764	0.000862	0.293019
43.5	0.001968	2.136931	0.001209	0.31122
44	0.001968	1.972252	0.001618	0.330902
44.5	0.001968	1.845954	0.002106	0.349388
45	0.004727	1.773889	0.002543	0.376717
45.5	0.007738	1.513242	0.003205	0.438474
46	0.007743	1.148286	0.004523	0.506185
46.5	0.007742	1.015279	0.006597	0.572378
47	0.007743	1	0.009479	0.639645
47.5	0.007742	1	0.012924	0.707357
48	0.007742	1	0.017396	0.773421
48.5	0.007743	1	0.02293	0.840817
49	0.007742	1	0.028858	0.90853
49.5	0.007742	1	0.036128	0.974461
50	#N/A	#N/A	#N/A	#N/A

Table B.6: Statistical data of 350 VoIP users without traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	#N/A	#N/A	#N/A	#N/A
5.5	0	3.055426	2.65E-06	0.148582
6	0	3.055426	2.81E-06	0.148568
6.5	0	3.055426	2.8E-06	0.148568
7	0	3.055426	2.8E-06	0.148568
7.5	0	3.055426	2.8E-06	0.148568

8	0	3.055426	2.8E-06	0.148568
8.5	0	3.055426	2.8E-06	0.148568
9	0	3.055426	2.8E-06	0.148568
9.5	0	3.055426	2.8E-06	0.148568
10	0.000416	3.055426	4.93E-06	0.152513
10.5	0	3.055426	6.71E-06	0.152513
11	0	3.055426	7.87E-06	0.152513
11.5	0	3.055426	8.62E-06	0.152513
12	0	3.055426	9.1E-06	0.152513
12.5	0	3.055426	9.39E-06	0.152513
13	0	3.055426	9.56E-06	0.152513
13.5	0	3.055426	9.65E-06	0.152513
14	0	3.055426	9.67E-06	0.152513
14.5	0	3.055426	9.65E-06	0.152513
15	0.00047	3.055426	1.12E-05	0.156481
15.5	0	3.055426	1.33E-05	0.156481
16	0	3.055426	1.49E-05	0.156481
16.5	0	3.055426	1.61E-05	0.156481
17	0	3.055426	1.7E-05	0.156481
17.5	0	3.055426	1.78E-05	0.156481
18	0	3.055426	1.83E-05	0.156481
18.5	0	3.055426	1.87E-05	0.156481
19	0	3.055426	1.9E-05	0.156481
19.5	0	3.055426	1.93E-05	0.156481
20	0.000514	3.050682	2.04E-05	0.160321
20.5	0	3.050682	2.31E-05	0.160463
21	0	3.050682	2.5E-05	0.160463
21.5	0	3.050682	2.66E-05	0.160463
22	0	3.050682	2.8E-05	0.160463
22.5	0	3.050682	2.9E-05	0.160463
23	0	3.050682	2.99E-05	0.160463
23.5	0	3.050682	3.07E-05	0.160463
24	0	3.050682	3.13E-05	0.160463
24.5	0	3.050682	3.17E-05	0.160463
25	0.000545	3.050682	3.24E-05	0.164228
25.5	0	3.050682	3.57E-05	0.164451
26	0	3.050682	3.8E-05	0.164451
26.5	0	3.050682	4E-05	0.164451
27	0	3.050682	4.17E-05	0.164451
27.5	0	3.050682	4.31E-05	0.164451
28	0	3.050682	4.43E-05	0.164451
28.5	0	3.050682	4.53E-05	0.164451
29	0	3.050682	4.62E-05	0.164451
29.5	0	3.050682	4.7E-05	0.164451
30	0.000568	3.050682	4.75E-05	0.168211
30.5	0	3.050682	5.1E-05	0.168442
31	0	3.050682	5.37E-05	0.168442
31.5	0	3.050682	5.61E-05	0.168442
32	0	3.050682	5.81E-05	0.168442
32.5	0	3.050682	5.98E-05	0.168442
33	0	3.050682	6.13E-05	0.168442
33.5	0	3.050682	6.27E-05	0.168442
34	0	3.050682	6.38E-05	0.168442
34.5	0	3.050682	6.48E-05	0.168442
35	0.003806	3.050682	6.63E-05	0.187383
35.5	0.003843	2.812454	0.000153	0.222786
36	0.003842	2.502689	0.000344	0.257965
36.5	0.003842	2.198362	0.000711	0.294083
37	0.003842	2.068499	0.001252	0.330204
37.5	0.003842	1.828387	0.002045	0.364941
38	0.003842	1.56513	0.003056	0.400899
38.5	0.003842	1.440179	0.004272	0.437018
39	0.003842	1.283434	0.005824	0.472515
39.5	0.003842	1.126302	0.007633	0.507716
40	0.00458	1.021284	0.009546	0.545545
40.5	0.010534	1	0.010808	0.614776

41	0.010572	1	0.013887	0.702327
41.5	0.010572	1	0.018523	0.788998
42	0.010572	1	0.024657	0.876107
42.5	0.010572	1	0.0319	0.96377
43	0.010572	1	0.040545	1.051429
43.5	0.010572	1	0.051653	1.137993
44	0.010572	1	0.064144	1.225214
44.5	0.010572	1	0.077378	1.312873
45	0.010772	1	0.092698	1.401089
45.5	0.010572	1	0.110037	1.487904
46	0.010571	1	0.130016	1.574565
46.5	0.016537	1	0.135116	1.684593
47	0.017265	1	0.15391	1.813001
47.5	0.017263	1	0.180618	1.940871
48	0.017264	1	0.203406	2.07017
48.5	0.017298	1	0.241407	2.197236
49	0.017302	1	0.271178	2.327019
49.5	0.017302	1	0.312328	2.454568
50	#N/A	#N/A	#N/A	#N/A

With Traffic

Table B.7: Statistical data of 100 VoIP users with traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation.	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	#N/A	#N/A	#N/A	#N/A
5.5	0	3.055426	1.65E-07	0.142906
6	0	3.055426	1.82E-07	0.142893
6.5	0	3.055426	1.81E-07	0.142893
7	0	3.055426	1.81E-07	0.142893
7.5	0	3.055426	1.81E-07	0.142893
8	0	3.055426	1.81E-07	0.142893
8.5	0	3.055426	1.81E-07	0.142893
9	0	3.055426	1.81E-07	0.142893
9.5	0	3.055426	1.81E-07	0.142893
10	0.000117	3.055426	2.86E-07	0.144009
10.5	0	3.055426	4.34E-07	0.144009
11	0	3.055426	5.32E-07	0.144009
11.5	0	3.055426	5.95E-07	0.144009
12	0	3.055426	6.36E-07	0.144009
12.5	0	3.055426	6.62E-07	0.144009
13	0	3.055426	6.77E-07	0.144009
13.5	0	3.055426	6.85E-07	0.144009
14	0	3.055426	6.88E-07	0.144009
14.5	0	3.055426	6.88E-07	0.144009
15	0.00013	3.055426	7.4E-07	0.145123
15.5	0	3.055426	9.06E-07	0.145123
16	0	3.055426	1.04E-06	0.145123
16.5	0	3.055426	1.14E-06	0.145123
17	0	3.055426	1.22E-06	0.145123
17.5	0	3.055426	1.28E-06	0.145123
18	0	3.055426	1.32E-06	0.145123

18.5	0	3.055426	1.36E-06	0.145123
19	0	3.055426	1.39E-06	0.145123
19.5	0	3.055426	1.41E-06	0.145123
20	0.000139	3.055426	1.44E-06	0.146236
20.5	0	3.055426	1.64E-06	0.146236
21	0	3.055426	1.8E-06	0.146236
21.5	0	3.055426	1.93E-06	0.146236
22	0	3.055426	2.04E-06	0.146236
22.5	0	3.055426	2.13E-06	0.146236
23	0	3.055426	2.2E-06	0.146236
23.5	0	3.055426	2.26E-06	0.146236
24	0	3.055426	2.32E-06	0.146236
24.5	0	3.055426	2.36E-06	0.146236
25	0.000147	3.055426	2.37E-06	0.147348
25.5	0	3.055426	2.6E-06	0.147348
26	0	3.055426	2.79E-06	0.147348
26.5	0	3.055426	2.95E-06	0.147348
27	0	3.055426	3.09E-06	0.147348
27.5	0	3.055426	3.21E-06	0.147348
28	0	3.055426	3.31E-06	0.147348
28.5	0	3.055426	3.39E-06	0.147348
29	0	3.055426	3.47E-06	0.147348
29.5	0	3.055426	3.53E-06	0.147348
30	0.000153	3.055426	3.53E-06	0.14846
30.5	0	3.055426	3.79E-06	0.14846
31	0	3.055426	4.01E-06	0.14846
31.5	0	3.055426	4.2E-06	0.14846
32	0	3.055426	4.36E-06	0.14846
32.5	0	3.055426	4.5E-06	0.14846
33	0	3.055426	4.63E-06	0.14846
33.5	0	3.055426	4.74E-06	0.14846
34	0	3.055426	4.83E-06	0.14846
34.5	0	3.055426	4.92E-06	0.14846
35	0.000111	3.055426	4.46E-06	0.149571
35.5	0	3.055426	4.7E-06	0.149572
36	0	3.055426	4.91E-06	0.149571
36.5	0	3.055426	5.1E-06	0.149571
37	0	3.055426	5.27E-06	0.149571
37.5	0	3.055426	5.43E-06	0.149571
38	0	3.055426	5.57E-06	0.149571
38.5	0	3.055426	5.7E-06	0.149571
39	0	3.055426	5.82E-06	0.149571
39.5	0	3.055426	5.93E-06	0.149571
40	3.26E-05	3.055426	5.4E-06	0.150685
40.5	0	3.055426	5.51E-06	0.150683
41	0	3.055426	5.61E-06	0.150683
41.5	0	3.055426	5.7E-06	0.150683
42	0	3.055426	5.78E-06	0.150683
42.5	0	3.055426	5.85E-06	0.150683
43	0	3.055426	5.91E-06	0.150683
43.5	0	3.055426	5.97E-06	0.150683
44	0	3.055426	6.02E-06	0.150683
44.5	0	3.055426	6.07E-06	0.150683
45	3.28E-05	3.055426	5.54E-06	0.151796
45.5	0	3.055426	5.6E-06	0.151794
46	0	3.055426	5.66E-06	0.151794
46.5	0	3.055426	5.71E-06	0.151794
47	0	3.055426	5.75E-06	0.151794
47.5	0	3.055426	5.79E-06	0.151794
48	0	3.055426	5.82E-06	0.151794
48.5	0	3.055426	5.86E-06	0.151794
49	0	3.055426	5.88E-06	0.151794
49.5	0	3.055426	5.91E-06	0.151794
50	#N/A	#N/A	#N/A	#N/A

Table B.8: Statistical data of 150 VoIP users with traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	7.36E-07	3.051098	1.16E-06	0.144075
5.5	-2.9E-07	3.051098	1.39E-06	0.144075
6	2.94E-07	3.051098	1.39E-06	0.144075
6.5	-2.1E-07	3.055426	1.3E-06	0.144054
7	3.33E-06	3.055426	1.38E-06	0.14404
7.5	-3.3E-06	3.055426	1.38E-06	0.14404
8	3.33E-06	3.055426	1.38E-06	0.144047
8.5	6.21E-05	3.055426	1.4E-06	0.144233
9	-6.2E-05	3.055426	1.47E-06	0.14431
9.5	0.0001	3.055426	1.51E-06	0.144338
10	8.6E-05	3.055426	1.65E-06	0.145965
10.5	0.000131	3.055426	2.15E-06	0.146144
11	-0.00011	3.055426	2.51E-06	0.146202
11.5	0.000149	3.055426	2.86E-06	0.146557
12	-0.00014	3.055426	3.09E-06	0.146262
12.5	7.27E-05	3.055426	3.17E-06	0.146297
13	-5.3E-05	3.055426	3.2E-06	0.146219
13.5	2.84E-06	3.055426	3.21E-06	0.146273
14	1.63E-05	3.055426	3.19E-06	0.146213
14.5	-5.1E-05	3.055426	3.16E-06	0.146293
15	0.000253	3.055426	2.91E-06	0.148003
15.5	-7.1E-05	3.055426	3.37E-06	0.148114
16	6.86E-05	3.055426	3.73E-06	0.148083
16.5	-6.8E-05	3.055426	4.08E-06	0.148552
17	2.73E-05	3.055426	4.55E-06	0.148416
17.5	-2.8E-05	3.055426	4.92E-06	0.148714
18	2.67E-05	3.055426	5.17E-06	0.148286
18.5	-2.8E-05	3.055426	5.31E-06	0.148462
19	4.94E-05	3.055426	5.38E-06	0.148183
19.5	-5.3E-05	3.055426	5.41E-06	0.14824
20	0.000303	3.055426	5.04E-06	0.14984
20.5	-0.00011	3.055426	5.53E-06	0.14994
21	0.000141	3.055426	5.86E-06	0.149877
21.5	-0.00013	3.055426	6.12E-06	0.149753
22	0.000113	3.055426	6.3E-06	0.149785
22.5	-9.4E-05	3.055426	6.45E-06	0.149762
23	8.8E-05	3.055426	6.54E-06	0.149791
23.5	-5.6E-05	3.055426	6.62E-06	0.149823
24	8.17E-05	3.055426	6.68E-06	0.149897
24.5	-5.1E-05	3.055426	6.75E-06	0.149964
25	0.000308	3.055426	6.41E-06	0.151774
25.5	-5.5E-05	3.055426	6.97E-06	0.151854
26	9.42E-05	3.055426	7.46E-06	0.151943
26.5	-5.9E-05	3.055426	7.92E-06	0.152008
27	6.3E-05	3.055426	8.31E-06	0.152053
27.5	-6.3E-05	3.055426	8.66E-06	0.152041
28	2.8E-05	3.055426	8.95E-06	0.152032
28.5	-6.8E-05	3.055426	9.2E-06	0.151897
29	4.43E-05	3.055426	9.4E-06	0.151836
29.5	-7.3E-05	3.055426	9.54E-06	0.151745
30	0.000298	3.051174	9.07E-06	0.153671
30.5	-6.2E-05	3.051174	9.7E-06	0.153615

31	7.45E-05	3.051174	1.02E-05	0.153799
31.5	-6.3E-05	3.051174	1.06E-05	0.153716
32	9.43E-05	3.051174	1.1E-05	0.153984
32.5	-5.5E-05	3.052344	1.14E-05	0.153979
33	0.000372	3.052468	1.19E-05	0.155971
33.5	0.000194	3.051098	1.38E-05	0.158766
34	0.000491	3.051098	1.81E-05	0.162196
34.5	0.000105	3.051098	2.55E-05	0.165189
35	0.002415	3.036942	3.96E-05	0.178496
35.5	0.002426	2.972278	9.95E-05	0.200885
36	0.002433	2.853732	0.000234	0.223852
36.5	0.002675	2.773842	0.000462	0.248366
37	0.004911	2.62775	0.000847	0.286104
37.5	0.004807	2.381592	0.001502	0.329123
38	0.004177	2.174294	0.0024	0.368338
38.5	0.005154	1.950429	0.003678	0.412753
39	0.007057	1.725295	0.005432	0.467589
39.5	0.007006	1.452355	0.007694	0.525868
40	0.006727	1.246563	0.010345	0.583791
40.5	0.009017	1.04719	0.013383	0.661061
41	0.009962	1	0.017032	0.74009
41.5	0.00964	1	0.022918	0.822561
42	0.009952	1	0.029149	0.896146
42.5	0.009488	1	0.038518	0.986222
43	0.009972	1	0.047615	1.063402
43.5	0.009239	1	0.058702	1.14513
44	0.009588	1	0.070875	1.225721
44.5	0.010588	1	0.08569	1.314665
45	0.011767	1	0.1005	1.40188
45.5	0.012031	1	0.117858	1.497626
46	0.013828	1	0.135989	1.602098
46.5	0.016608	1	0.146988	1.714144
47	0.018409	1	0.162481	1.836523
47.5	0.018982	1	0.190749	1.984774
48	0.019152	1	0.217576	2.113401
48.5	0.02048	1	0.253905	2.265846
49	0.020433	1	0.291079	2.404203
49.5	0.020907	1	0.332596	2.55207
50	#N/A	#N/A	#N/A	#N/A

Table B.9: Statistical data of 200 VoIP users with traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	0	3.050606	2.08E-06	0.145209
5.5	0	3.050606	2.46E-06	0.145209
6	0	3.051098	2.47E-06	0.145219
6.5	1.65E-07	3.055426	2.33E-06	0.145196
7	-2.4E-06	3.055426	2.45E-06	0.145179
7.5	2.46E-06	3.055426	2.46E-06	0.145327
8	1.87E-05	3.055426	2.57E-06	0.145515
8.5	-1.5E-05	3.055426	2.7E-06	0.145699
9	1.15E-05	3.055426	3.4E-06	0.146291
9.5	-1.1E-05	3.055426	3.83E-06	0.146154

10	0.000288	3.055426	3.75E-06	0.147942
10.5	-8.3E-05	3.055426	4.5E-06	0.148094
11	0.000117	3.055426	4.95E-06	0.148122
11.5	-9E-05	3.055426	5.2E-06	0.148048
12	0.000123	3.055426	5.31E-06	0.148114
12.5	-9.9E-05	3.055426	5.35E-06	0.148064
13	9.42E-05	3.055426	5.37E-06	0.148184
13.5	-6.5E-05	3.055426	5.37E-06	0.148075
14	2.41E-05	3.055426	5.33E-06	0.148098
14.5	6.94E-06	3.055426	5.29E-06	0.148093
15	0.000233	3.055426	5.09E-06	0.150395
15.5	6.86E-05	3.055426	5.77E-06	0.150436
16	-0.00011	3.055426	6.3E-06	0.150329
16.5	0.000121	3.055175	6.69E-06	0.150464
17	-0.00014	3.055426	6.98E-06	0.150251
17.5	0.000122	3.055426	7.17E-06	0.150373
18	-0.00012	3.055426	7.31E-06	0.150145
18.5	5.39E-05	3.055426	7.38E-06	0.150148
19	-5.7E-05	3.055426	7.4E-06	0.150016
19.5	1.38E-05	3.055426	7.37E-06	0.150003
20	0.000252	3.051174	7.18E-06	0.152294
20.5	6.33E-06	3.051174	7.84E-06	0.152277
21	-8.7E-06	3.051174	8.32E-06	0.152222
21.5	3.74E-06	3.051174	8.68E-06	0.152236
22	-5.8E-06	3.051174	8.96E-06	0.152199
22.5	2.93E-06	3.051174	9.18E-06	0.152208
23	-4.8E-06	3.051174	9.34E-06	0.152198
23.5	1.97E-06	3.051174	9.47E-06	0.152215
24	-2.5E-06	3.051174	9.58E-06	0.152444
24.5	5.64E-08	3.051174	9.67E-06	0.152405
25	0.000685	3.051098	9.37E-06	0.156421
25.5	0.000673	3.051098	1.5E-05	0.162156
26	0.002118	3.051098	3.34E-05	0.172919
26.5	0.005585	2.932787	0.000135	0.210589
27	0.006561	2.635948	0.000495	0.266944
27.5	0.007092	2.268291	0.001371	0.328283
28	0.006438	1.948045	0.002797	0.385678
28.5	0.006872	1.631046	0.0048	0.445507
29	0.006565	1.384659	0.007639	0.505905
29.5	0.006799	1.162682	0.011208	0.563891
30	0.007035	1.039467	0.014991	0.622262
30.5	0.010067	1	0.017688	0.693669
31	0.01053	1	0.022861	0.783324
31.5	0.010813	1	0.03025	0.867496
32	0.010657	1	0.039802	0.960879
32.5	0.01048	1	0.04986	1.044716
33	0.010757	1	0.062238	1.133083
33.5	0.010447	1	0.076337	1.22046
34	0.010745	1	0.091576	1.30213
34.5	0.010677	1	0.109658	1.394049
35	0.01156	1	0.129089	1.487174
35.5	0.011586	1	0.149818	1.582792
36	0.012378	1	0.172063	1.673407
36.5	0.016194	1	0.182419	1.786303
37	0.017564	1	0.197767	1.914058
37.5	0.017268	1	0.22678	2.043722
38	0.018795	1	0.256737	2.172195
38.5	0.018892	1	0.296516	2.318636
39	0.01902	1	0.336429	2.459008
39.5	0.020157	1	0.382903	2.597935
40	0.020084	1	0.43067	2.738087
40.5	0.0206	1	0.485763	2.88176
41	0.021195	1	0.545779	3.033943
41.5	0.022838	1	0.606381	3.177828
42	0.022824	1	0.674135	3.337224
42.5	0.024328	1	0.755155	3.508273

43	0.026921	1	0.814356	3.667478
43.5	0.028281	1	0.87089	3.847925
44	0.03	1	0.898682	4.026645
44.5	0.031347	1	0.992152	4.229602
45	0.031194	1	1.087361	4.429587
45.5	0.032687	1	1.189384	4.622448
46	0.032033	1	1.297611	4.812258
46.5	0.033115	1	1.419936	5.004576
47	0.033041	1	1.555514	5.205159
47.5	0.034434	1	1.696707	5.406163
48	0.034444	1	1.856927	5.616076
48.5	0.036133	1	1.997354	5.81965
49	0.037069	1	2.163874	6.034892
49.5	0.037939	1	2.344489	6.249133
50	#N/A	#N/A	#N/A	#N/A

Table B.10: Statistical data of 250 VoIP users with traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	1.59E-09	3.050606	3.29E-06	0.146344
5.5	-6.4E-10	3.050606	3.86E-06	0.146344
6	0	3.051098	3.87E-06	0.146345
6.5	1.38E-07	3.055426	3.65E-06	0.14633
7	-3.6E-06	3.055426	4.05E-06	0.146611
7.5	9.82E-05	3.055426	4.38E-06	0.146641
8	-9.8E-05	3.055426	4.48E-06	0.146532
8.5	9.36E-05	3.055426	4.53E-06	0.146641
9	-9.4E-05	3.055426	5.27E-06	0.147134
9.5	3.07E-05	3.055426	5.29E-06	0.146751
10	0.00024	3.055426	5.05E-06	0.149265
10.5	3.34E-05	3.055426	5.91E-06	0.149273
11	-3.3E-05	3.055426	6.36E-06	0.149225
11.5	3E-05	3.055426	6.61E-06	0.149252
12	-3.8E-05	3.055426	6.73E-06	0.14922
12.5	3.64E-05	3.055426	6.78E-06	0.149263
13	-2.9E-05	3.055426	6.77E-06	0.149215
13.5	2.51E-05	3.055426	6.74E-06	0.149251
14	-2.5E-05	3.055426	6.68E-06	0.14922
14.5	2.5E-05	3.055426	6.62E-06	0.149393
15	0.00025	3.055426	6.42E-06	0.152471
15.5	3.63E-05	3.055426	7.73E-06	0.152489
16	-3.4E-06	3.055426	8.7E-06	0.152822
16.5	1.08E-05	3.055426	9.56E-06	0.152958
17	-1.8E-05	3.055426	1.03E-05	0.152976
17.5	1.5E-05	3.055426	1.08E-05	0.153033
18	-1.6E-05	3.055426	1.13E-05	0.153001
18.5	1.14E-05	3.055426	1.17E-05	0.153042
19	-9.6E-06	3.055426	1.2E-05	0.152881
19.5	6.07E-06	3.055426	1.22E-05	0.153035
20	0.000318	3.051174	1.2E-05	0.155792
20.5	1.41E-05	3.051174	1.34E-05	0.155949
21	-1.6E-05	3.051174	1.45E-05	0.155682
21.5	9.69E-06	3.051174	1.54E-05	0.155874

22	-1.5E-05	3.051174	1.61E-05	0.155603
22.5	3.21E-05	3.051174	1.66E-05	0.155856
23	-3.4E-05	3.051174	1.7E-05	0.155593
23.5	4.89E-05	3.051174	1.74E-05	0.155891
24	-3.3E-05	3.051174	1.76E-05	0.15575
24.5	0.000502	3.051174	1.77E-05	0.157137
25	0.005971	2.977021	4.45E-05	0.186718
25.5	0.007999	2.741221	0.000303	0.250224
26	0.011167	2.232395	0.001119	0.33067
26.5	0.012387	1.677456	0.003007	0.429146
27	0.012331	1.242377	0.006384	0.527854
27.5	0.013431	1.006709	0.011051	0.62603
28	0.014497	1	0.018157	0.737348
28.5	0.014568	1	0.0274	0.848179
29	0.016324	1	0.039173	0.967762
29.5	0.015605	1	0.054852	1.090008
30	0.017803	1	0.07104	1.211459
30.5	0.01725	1	0.092855	1.342814
31	0.019523	1	0.111451	1.473265
31.5	0.024936	1	0.125967	1.636977
32	0.024232	1	0.152061	1.796089
32.5	0.02437	1	0.190254	1.958469
33	0.025807	1	0.231586	2.132171
33.5	0.02433	1	0.274008	2.295027
34	0.024409	1	0.332444	2.456386
34.5	0.027299	1	0.38923	2.628947
35	0.026351	1	0.455715	2.805923
35.5	0.028683	1	0.523501	2.983338
36	0.028257	1	0.61087	3.165475
36.5	0.02927	1	0.69089	3.348662
37	0.028953	1	0.787769	3.529001
37.5	0.02901	1	0.886425	3.711382
38	0.029331	1	0.998273	3.892934
38.5	0.029755	1	1.078787	4.078868
39	0.034036	1	1.138134	4.266487
39.5	0.035322	1	1.204683	4.48344
40	0.037371	1	1.309165	4.684528
40.5	0.037618	1	1.456091	4.909069
41	0.03775	1	1.596624	5.117422
41.5	0.038374	1	1.752157	5.338591
42	0.038721	1	1.920946	5.55662
42.5	0.037713	1	2.088347	5.777639
43	0.038625	1	2.281298	5.989547
43.5	0.037664	1	2.47396	6.201575
44	0.039051	1	2.695582	6.418183
44.5	0.041947	1	2.895345	6.658252
45	0.040424	1	3.131932	6.869642
45.5	0.042405	1	3.383616	7.115765
46	0.042071	1	3.617072	7.315836
46.5	0.040585	1	3.882687	7.565231
47	0.043105	1	4.177285	7.781677
47.5	0.046247	1	4.191628	8.009983
48	0.044686	1	4.490882	8.253577
48.5	0.047933	1	4.475986	8.49699
49	0.049685	1	4.781405	8.736435
49.5	0.04858	1	5.068695	8.997699
50	#N/A	#N/A	#N/A	#N/A

Table B.11: Statistical data of 300 VoIP users with traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	0	3.050606	4.79E-06	0.147478
5.5	0	3.050606	5.58E-06	0.147478
6	6.71E-07	3.050606	5.6E-06	0.147479
6.5	-4.8E-07	3.055426	5.31E-06	0.147449
7	0	3.055426	5.56E-06	0.147434
7.5	0	3.055426	5.57E-06	0.147435
8	0	3.055426	5.57E-06	0.147434
8.5	0	3.055426	5.57E-06	0.147434
9	0	3.055426	6.19E-06	0.148019
9.5	0	3.055426	6.5E-06	0.147882
10	0.000315	3.055426	6.72E-06	0.150903
10.5	-1.3E-06	3.055426	8.05E-06	0.150932
11	7.91E-07	3.055426	8.81E-06	0.150925
11.5	-6.3E-07	3.055426	9.25E-06	0.150932
12	6.73E-07	3.055426	9.49E-06	0.150924
12.5	-3.8E-07	3.055426	9.61E-06	0.15093
13	-5.4E-07	3.055426	9.64E-06	0.150928
13.5	8.73E-07	3.055426	9.61E-06	0.150977
14	6.34E-06	3.055159	9.55E-06	0.150978
14.5	-5.7E-06	3.055426	9.47E-06	0.150964
15	0.000375	3.051174	9.84E-06	0.154463
15.5	-2.4E-05	3.051174	1.13E-05	0.154449
16	2.23E-05	3.051174	1.23E-05	0.154483
16.5	-2.3E-05	3.051174	1.32E-05	0.154475
17	2.71E-05	3.051174	1.38E-05	0.154514
17.5	-2.8E-05	3.051174	1.42E-05	0.154576
18	6.77E-05	3.051174	1.45E-05	0.154695
18.5	0.000253	3.051174	1.49E-05	0.155473
19	0.000141	3.029135	1.55E-05	0.156923
19.5	0.002197	2.988876	2.06E-05	0.168355
20	0.009391	2.822661	0.000119	0.220316
20.5	0.009318	2.27216	0.000696	0.297229
21	0.01074	1.824817	0.002109	0.380602
21.5	0.011226	1.304646	0.004888	0.474384
22	0.012539	1.028437	0.009135	0.571475
22.5	0.013605	1	0.015388	0.674899
23	0.013726	1	0.023634	0.781236
23.5	0.013545	1	0.034566	0.887653
24	0.013583	1	0.048041	0.994344
24.5	0.013353	1	0.0642	1.10039
25	0.013575	1	0.082307	1.207349
25.5	0.013475	1	0.103997	1.313756
26	0.016321	1	0.117978	1.426095
26.5	0.020059	1	0.128188	1.562462
27	0.020583	1	0.152965	1.706933
27.5	0.020836	1	0.187364	1.852908
28	0.021408	1	0.220811	2.000811
28.5	0.021918	1	0.262485	2.152626
29	0.022787	1	0.311475	2.307295
29.5	0.023106	1	0.361795	2.463289
30	0.023607	1	0.420803	2.625946
30.5	0.024519	1	0.48355	2.792398

31	0.025976	1	0.551284	2.956019
31.5	0.024527	1	0.629398	3.121871
32	0.026187	1	0.717308	3.295192
32.5	0.027526	1	0.801467	3.46924
33	0.026175	1	0.894926	3.640642
33.5	0.032967	1	0.878919	3.823915
34	0.032943	1	0.946093	4.022908
34.5	0.035768	1	1.070499	4.225926
35	0.035147	1	1.160119	4.433076
35.5	0.036547	1	1.281982	4.648042
36	0.036207	1	1.404661	4.861743
36.5	0.03716	1	1.543122	5.069899
37	0.037165	1	1.70787	5.28065
37.5	0.03666	1	1.87929	5.494065
38	0.038454	1	2.046765	5.706697
38.5	0.037335	1	2.224061	5.926077
39	0.040282	1	2.410093	6.146943
39.5	0.04027	1	2.602109	6.361174
40	0.040062	1	2.823461	6.584031
40.5	0.039474	1	3.094852	6.807866
41	0.040771	1	3.300751	7.026709
41.5	0.039725	1	3.520362	7.252695
42	0.042235	1	3.687687	7.483785
42.5	0.047271	1	3.563765	7.71098
43	0.04742	1	3.852542	7.953827
43.5	0.05012	1	4.057508	8.20429
44	0.050503	1	4.32314	8.450019
44.5	0.051922	1	4.587514	8.711228
45	0.052801	1	4.848286	8.963646
45.5	0.052824	1	5.204819	9.213381
46	0.052633	1	5.516185	9.47033
46.5	0.052149	1	5.84671	9.733396
47	0.051343	1	6.20803	9.993955
47.5	0.053465	1	6.550046	10.23595
48	0.052537	1	6.968336	10.49269
48.5	0.052669	1	7.365156	10.75493
49	0.051586	1	7.790078	11.01722
49.5	0.052448	1	8.180142	11.27048
50	#N/A	#N/A	#N/A	#N/A

Table B.12: Statistical data of 350 VoIP users with traffic

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.5	#N/A	#N/A	#N/A	#N/A
1	#N/A	#N/A	#N/A	#N/A
1.5	#N/A	#N/A	#N/A	#N/A
2	#N/A	#N/A	#N/A	#N/A
2.5	#N/A	#N/A	#N/A	#N/A
3	#N/A	#N/A	#N/A	#N/A
3.5	#N/A	#N/A	#N/A	#N/A
4	#N/A	#N/A	#N/A	#N/A
4.5	#N/A	#N/A	#N/A	#N/A
5	3.82E-07	3.051098	6.58E-06	0.148612
5.5	-1.5E-07	3.051098	7.62E-06	0.148637
6	1.53E-07	3.051098	7.64E-06	0.148628
6.5	-1.1E-07	3.055426	7.25E-06	0.148602
7	7.47E-06	3.055426	7.58E-06	0.148616
7.5	-7.1E-06	3.055426	7.59E-06	0.148619
8	7.07E-06	3.055426	7.6E-06	0.148611
8.5	-7.1E-06	3.055426	7.6E-06	0.148616
9	7.07E-06	3.055426	8.09E-06	0.149176
9.5	-7.1E-06	3.055426	8.66E-06	0.149288

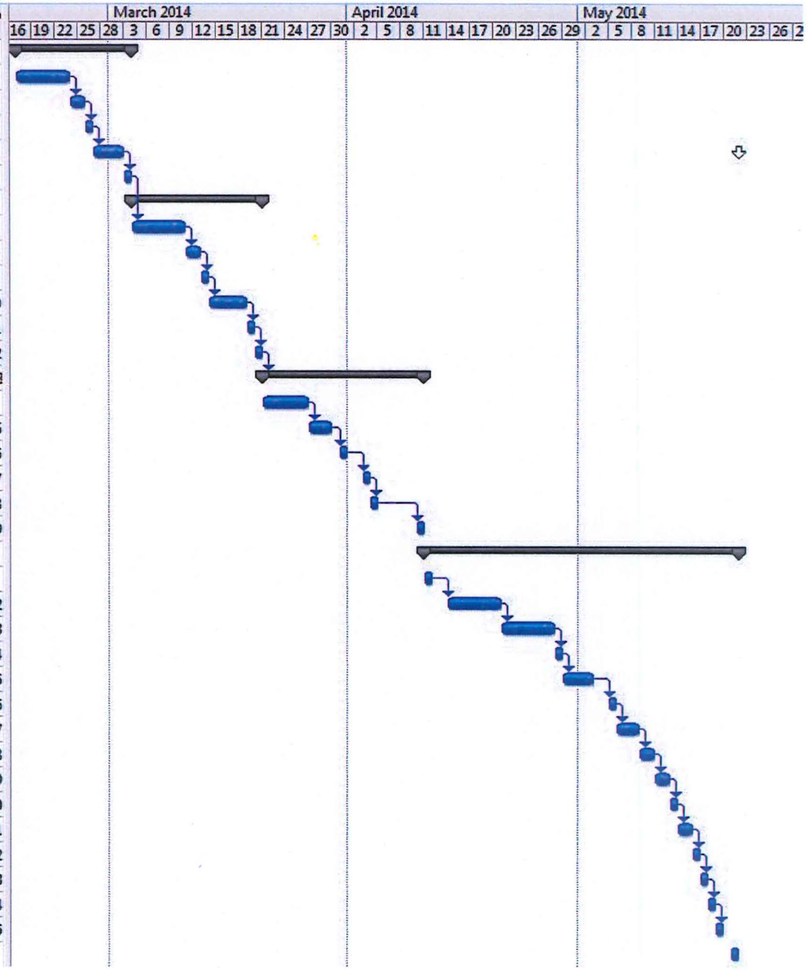
10	0.00038	3.055426	9.06E-06	0.152625
10.5	-1.4E-05	3.051174	1.09E-05	0.152649
11	0.000113	3.051174	1.19E-05	0.152866
11.5	1.26E-05	3.051174	1.27E-05	0.153243
12	1.79E-05	3.051174	1.33E-05	0.153334
12.5	1.64E-05	3.051174	1.37E-05	0.153567
13	2.3E-05	3.051174	1.42E-05	0.153994
13.5	-1.7E-05	3.051174	1.46E-05	0.153844
14	1.6E-05	3.051098	1.48E-05	0.154017
14.5	3.19E-05	3.052371	1.5E-05	0.154055
15	0.006195	3.05054	4.91E-05	0.181691
15.5	0.009074	2.748214	0.000409	0.247939
16	0.010432	2.158089	0.001612	0.332934
16.5	0.010269	1.644536	0.004048	0.41797
17	0.011343	1.301788	0.007892	0.504427
17.5	0.012712	1.033038	0.013643	0.602917
18	0.012733	1	0.021576	0.704884
18.5	0.013021	1	0.031663	0.805272
19	0.014193	1	0.044172	0.91473
19.5	0.016262	1	0.059616	1.032015
20	0.01941	1	0.077741	1.16513
20.5	0.021303	1	0.101152	1.310936
21	0.024132	1	0.118075	1.460906
21.5	0.029179	1	0.130636	1.641648
22	0.031346	1	0.163363	1.83487
22.5	0.028823	1	0.201951	2.014664
23	0.031614	1	0.252209	2.200922
23.5	0.028848	1	0.309254	2.388466
24	0.031708	1	0.370903	2.581886
24.5	0.030893	1	0.450735	2.766735
25	0.031124	1	0.531543	2.967137
25.5	0.031629	1	0.62646	3.15458
26	0.031234	1	0.718185	3.344353
26.5	0.031725	1	0.840007	3.538381
27	0.032254	1	0.951109	3.737019
27.5	0.031851	1	1.07182	3.927767
28	0.034812	1	1.21873	4.139631
28.5	0.032019	1	1.363406	4.337777
29	0.037021	1	1.394947	4.529724
29.5	0.037431	1	1.481543	4.740748
30	0.039681	1	1.519522	4.966366
30.5	0.041221	1	1.698545	5.19316
31	0.038969	1	1.817788	5.414459
31.5	0.041148	1	2.062899	5.638873
32	0.040453	1	2.191965	5.850339
32.5	0.04044	1	2.414359	6.097293
33	0.042135	1	2.61645	6.304483
33.5	0.040801	1	2.800359	6.541452
34	0.041553	1	3.116709	6.767622
34.5	0.043321	1	3.327687	6.991529
35	0.041864	1	3.520461	7.229746
35.5	0.042935	1	3.898903	7.449557
36	0.044916	1	4.086081	7.696287
36.5	0.043475	1	4.469243	7.934865
37	0.04437	1	4.655518	8.153227
37.5	0.045273	1	5.009792	8.399169
38	0.045454	1	5.221319	8.628781
38.5	0.048073	1	5.240028	8.868183
39	0.052023	1	5.221377	9.119769
39.5	0.052058	1	5.507562	9.374342
40	0.052174	1	5.820757	9.629161
40.5	0.05216	1	6.14333	9.88488
41	0.053368	1	6.383959	10.14586
41.5	0.052037	1	6.877421	10.39694
42	0.051596	1	7.245489	10.64999
42.5	0.051941	1	7.642843	10.90728

43	0.053685	1	8.023121	11.16621
43.5	0.051842	1	8.440627	11.42763
44	0.051719	1	8.818372	11.68601
44.5	0.051925	1	9.363854	11.93989
45	0.053296	1	9.841341	12.18978
45.5	0.052412	1	10.24483	12.43689
46	0.052874	1	10.78035	12.69731
46.5	0.052842	1	11.39105	12.95458
47	0.052697	1	11.81273	13.21163
47.5	0.055738	1	12.4533	13.47724
48	0.054396	1	12.83584	13.72986
48.5	0.05675	1	12.99218	13.98332
49	0.057975	1	12.80845	14.26553
49.5	0.060622	1	12.78993	14.53383
50	#N/A	#N/A	#N/A	#N/A

APPENDIX C

GANTT CHART

		Task Name	Duration	Start	Finish	Pr
1		**Preparation of Final Year Project**	**14 days**	**Mon 17/2/14**	**Mon 3/3/14**	
2		Search for project title	7 days	Mon 17/2/14	Sun 23/2/14	
3		Wrtie proposal	2 days	Mon 24/2/14	Tue 25/2/14	2
4		Propose research title to supervisor	1 day	Wed 26/2/14	Wed 26/2/14	3
5		Correcton of Proposal	3 days	Thu 27/2/14	Sun 2/3/14	4
6		Submission of Proposal	1 day	Mon 3/3/14	Mon 3/3/14	5
7		**PRELIMINARY STUDY PHASE**	**13 days**	**Tue 4/3/14**	**Thu 20/3/14**	
8		Search the related information about Chapter 1	5 days	Tue 4/3/14	Mon 10/3/14	6
9		Start writing Chapter 1	2 days	Tue 11/3/14	Wed 12/3/14	8
10		Submit the draft Chapter 1 to supervisor	1 day	Thu 13/3/14	Thu 13/3/14	9
11		Correction of Chapter 1	3 days	Fri 14/3/14	Tue 18/3/14	10
12		Submit the corrected Chapter 1	1 day	Wed 19/3/14	Wed 19/3/14	11
13		Interview with PTMK	1 day	Thu 20/3/14	Thu 20/3/14	12
14		**RESEARCH PLANNING PHASE**	**16 days**	**Fri 21/3/14**	**Thu 10/4/14**	13
15		Search the information of Chapter 2	4 days	Fri 21/3/14	Wed 26/3/14	
16		Start Writing Of Chapter 2	3 days	Thu 27/3/14	Sat 29/3/14	15
17		Submission Chapter 2 to supervisor	1 day	Mon 31/3/14	Mon 31/3/14	16
18		Correction Chapter 2	1 day	Thu 3/4/14	Thu 3/4/14	17
19		Submission Finalize Chapter 2 to Supervisor	1 day	Fri 4/4/14	Fri 4/4/14	18
20		Search the OPNET Modeler software	1 day	Thu 10/4/14	Thu 10/4/14	19
21		**ARCHITECTURE DESIGN PHASE**	**35 days**	**Fri 11/4/14**	**Wed 21/5/14**	
22		Installation of OPNET into laptop	1 day	Fri 11/4/14	Fri 11/4/14	
23		Search the relevant information of Chapter 3	6 days	Mon 14/4/14	Sun 20/4/14	22
24		Define and Start writing Chapter 3	6 days	Mon 21/4/14	Sun 27/4/14	23
25		Submission of Chapter 3	1 day	Mon 28/4/14	Mon 28/4/14	24
26		Correction of Chapter 3	4 days	Tue 29/4/14	Fri 2/5/14	25
27		Resubmission of Chapter 3	1 day	Mon 5/5/14	Mon 5/5/14	26
28		Design the network topology using Opnet	3 days	Tue 6/5/14	Thu 8/5/14	27
29		Search the relevant information of Chapter 4	2 days	Fri 9/5/14	Sat 10/5/14	28
30		Start Writing of Chapter 4	2 days	Sun 11/5/14	Mon 12/5/14	29
31		Submission of Chapter 4	1 day	Tue 13/5/14	Tue 13/5/14	30
32		Correction of Chapter 4	2 days	Wed 14/5/14	Thu 15/5/14	31
33		Submission of Chapter 4	1 day	Fri 16/5/14	Fri 16/5/14	32
34		Prepare PSM 1 Presentation slide	1 day	Sat 17/5/14	Sat 17/5/14	33
35		Submit Slide to Supervisor	1 day	Sun 18/5/14	Sun 18/5/14	34
36		Correction of PSM 1 Presentation slide	1 day	Mon 19/5/14	Mon 19/5/14	35
37		PSM 1 Presentation	1 day	Wed 21/5/14	Wed 21/5/14	



		Task Name	Duration	Start	Finish	Pr
38		**TESTING PHASE**	**56 days**	**Thu 10/7/14**	**Thu 25/9/14**	
39		Design the network topology using Opnet	10 days	Thu 10/7/14	Wed 23/7/14	
40		Testing the network topology	6 days	Thu 24/7/14	Thu 31/7/14	39
41		Solve the error in Opnet	5 days	Fri 1/8/14	Thu 7/8/14	40
42		Manipulate the design network topology	10 days	Fri 8/8/14	Thu 21/8/14	41
43		Testing the network topology	2 days	Fri 22/8/14	Mon 25/8/14	42
44		Solve the error	3 days	Tue 26/8/14	Thu 28/8/14	43
45		Complete the network topology with no error	5 days	Fri 29/8/14	Thu 4/9/14	44
46		Meeting With Supervisor	1 day	Mon 8/9/14	Mon 8/9/14	
47		Continue the progress of Chapter 4	6 days	Tue 9/9/14	Tue 16/9/14	46
48		Submission of Chapter 4	1 day	Wed 17/9/14	Wed 17/9/14	47
49		Correction of Chapter 4	5 days	Thu 18/9/14	Wed 24/9/14	48
50		Submission of Chapter 4	1 day	Thu 25/9/14	Thu 25/9/14	49
51		**DATA ANALYSIS PHASE**	**58 days**	**Fri 26/9/14**	**Tue 16/12/14**	
52		Get the relevant information of Chapter 5	4 days	Fri 26/9/14	Wed 1/10/14	
53		Start writing of Chapter 5	20 days	Thu 2/10/14	Wed 29/10/14	52
54		Submission of Chapter 5	1 day	Thu 30/10/14	Thu 30/10/14	53
55		Correction of Chapter 5	8 days	Fri 31/10/14	Tue 11/11/14	54
56		Submission of Chapter 5	1 day	Wed 12/11/14	Wed 12/11/14	55
57		Search the relevant information of Chapter 6	3 days	Thu 13/11/14	Mon 17/11/14	56
58		Start writing of Chapter 6	3 days	Tue 18/11/14	Thu 20/11/14	57
59		Submission of Chapter 6	1 day	Fri 21/11/14	Fri 21/11/14	58
60		Correction of Chapter 6	2 days	Mon 24/11/14	Tue 25/11/14	59
61		Submission of Chapter 6	1 day	Wed 26/11/14	Wed 26/11/14	60
62		Meet Supervisor	1 day	Thu 27/11/14	Thu 27/11/14	61
63		Write the abstract of this research	3 days	Fri 28/11/14	Tue 2/12/14	62
64		Last check of all chapters	4 days	Wed 3/12/14	Mon 8/12/14	63
65		Submit PSM 2	1 day	Tue 9/12/14	Tue 9/12/14	64
66		Preparation of the PSM 2 presentation slides	2 days	Wed 10/12/14	Thu 11/12/14	65
67		Submission of PSM presentation slides	1 day	Fri 12/12/14	Fri 12/12/14	66
68		Present PSM 2	1 day	Tue 16/12/14	Tue 16/12/14