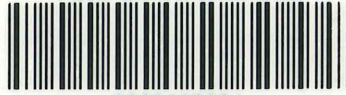


PERPUSTAKAAN UMP



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PERFORMANCE ANALYSIS ON VOIP FOR
KOLEJ KEDIAMAN 2 & 3, UNIVERSITI
MALAYSIA PAHANG

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BACHELOR OF COMPUTER SCIENCE
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BORANG PENGESAHAN STATUS TESIS

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SESI PENGAJIAN: 2014/ 2015

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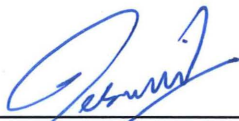
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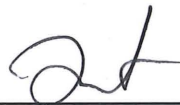
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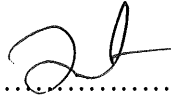
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PERFORMANCE ANALYSIS ON VOIP FOR KOLEJ KEDIAMAN 2 & 3,
UNIVERSITI MALAYSIA PAHANG

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This thesis submitted in fulfilment of the requirements
for the award of the degree of
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Faculty of Computer Systems & Software Engineering
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DECEMBER 2014

DEDICATION

This project is dedicated to my beloved parents,
Mr. Ko Son Hong and **Ms. Gan Chiw Yet**,
who always give me positive and moral support,
during the time I do my research.

To my supervisor,
Mr. Imran Edzereiq bin Kamarudin.
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To all my friends,
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ABSTRACT

Voice over Internet Protocol (VoIP) is a useful technology that allows users able to make unlimited free call no matter is local or international call. With this technology, the voice can be transported through over data networks. Unfortunately, the voice data will cope with some traffic that can affect the quality of VoIP calls. In this research, the maximum number of concurrent VoIP users will be estimated based on three factors which are packet jitter, packet end to end delays and MOS value. Besides, with the likes of Internet application such as video streaming, downloading of video and audio, online gaming which can directly affected the performance of VoIP. Therefore, two types of simulation scenarios can be done. First scenario is a campus network with VoIP application only. While the second scenario is a campus network with applying VoIP application and some external traffics such as HTTP and video streaming. The result these two scenarios will simulate by using OPNET modeler 14.5. In the testing phase, the number of clients in each scenario increased gradually until the threshold value for packet jitter, packet end to end delays and MOS are unacceptable. After that, the result that obtained from the simulation experiment will be analyzed and recommend the number of VoIP users for each factor. At the end, all the recommended number of users in each factor will combine in a table form and analyze again in order to determine the most recommended maximum number of VoIP users for each scenario. As a conclusion of this research, the recommended number of VoIP users for scenario 1 is less than 900 users while for scenario 2 is less 700 users.

ABSTRAK

Voice over Internet Protocol (VoIP) merupakan satu teknologi yang berguna yang membolehkan pengguna-pengguna dapat membuat panggilan percuma tanpa had tidak kira panggilan tempatan atau antarabangsa. Dengan teknologi ini, suara dapat dihantar melalui rangkaian data. Tetapi malangnya, suara dalam data akan menghadapi lalu lintas yang boleh menjejaskan kualiti panggilan VoIP. Dalam kajian ini, bilangan maksimum pengguna VoIP akan dianggarkan berdasarkan pada tiga faktor seperti ketar paket, paket kelewatan hujung ke hujung dan nilai MOS. Selain itu, aplikasi Internet seperti video streaming, memuat turun video dan audio, permainan online yang boleh menjejaskan prestasi VoIP secara langsung. Oleh itu, dua jenis simulasi scenario boleh dijalankan. Scenario pertama adalah rangkaian kampus dengan aplikasi VoIP sahaja. Manakala scenario kedua adalah rangkaian kampus dengan menggunakan aplikasi VoIP dan beberapa trafik luaran seperti HTTP dan video streaming. Hasil daripada kedua-dua senario akan disimulasikan dengan menggunakan OPNET modeler 14.5. Dalam fasa ujian, bilangan pelanggan dalam setiap senario akan meningkat secara beransur-ansur sehingga nilai ambang untuk ketar paket, paket kelewatan hujung ke hujung dan MOS tidak dapat diterima. Selepas itu, graf yang diperolehi daripada eksperimen simulasi akan dianalisis dan mengesyorkan bilangan pengguna VoIP bagi setiap faktor. Pada akhirnya, semua bilangan pengguna yang disarankan dari setiap faktor akan digabungkan dalam jadual and menganalisis semula untuk mendapatkan jumlah maksimum pengguna VoIP yang disyorkan bagi setiap senario. Sebagai kesimpulannya, jumlah pengguna VoIP yang disarankan untuk senario 1 adalah kurang daripada 900 pengguna manakala bagi senario 2 adalah kurang daripada 700 pengguna.

TABLE OF CONTENTS

	Page
SUPERVISOR’S DECLARATION	i
STUDENT’S DECLARATION	ii
DEDICATION	iii
ACKNOWLEDGMENT	iv
ABSTRACT	v
ABSTRAK	vi
TABLES OF CONTENTS	vii
LIST OF TABLES	x
LIST OF FIGURES	xi
CHAPTER 1 INTRODUCTION	
1.0 Introduction	1
1.1 Problem Statement	2
1.2 Objectives	3
1.3 Scopes	3
1.4 Thesis Organization	4
CHAPTER 2 LITERATURE REVIEW	
2.0 How VoIP Works	5
2.1 CODEC	7
2.1.1 MOS	8
2.1.2 R-Factor	9
2.2 Typical VoIP Problems	10
2.2.1 Packet Loss	10
2.2.2 Packet Jitter	12
2.2.3 Packet End-to-End Delays	13

2.3	Existing Research	14
2.3.1	A Comparison Of VoIP Performance Over Three Routing Protocols For IEEE 802.11s-based Wireless Mesh Networks (WLAN Mesh)	14
2.3.2	Analyzing Of MOS And Codec Selection For Voice Over IP Technology	15
2.3.3	Performance Studies Of VoIP Over Ethernet LANs	17
2.4	Comparison Between Existing Researches	18
2.4.1	Proposed Research	18
2.4.2	Comparison between Proposed Research and Existing Research	19
2.4.3	Comparison between Existing Researches	20
 CHAPTER 3 METHODOLOGY		
3.0	Introduction	22
3.1	Overview of Research Methods	24
3.1.1	Preliminary Study Phase	24
3.1.2	Research Planning Phase	25
3.1.3	Architecture Design Phase	26
3.1.4	Testing Phase	28
3.1.5	Data Analysis Phase	28
 CHAPTER 4 RESEARCH FINDINGS AND ANALYSIS		
4.0	Introduction	29
4.1	Experiment Network Design	30
4.2	Experiment Architecture	31
4.3	Testing Plan	33
 CHAPTER 5 RESULT AND DISCUSSION		
5.0	Introduction	35
5.1	Simulation Result and Comparisons	35
5.1.1	Scenario 1: VoIP Calls In LAN Without Traffic	35
5.1.1.1	Jitter in VoIP Calls	37
5.1.1.2	Packet End-to-End Delays	39
5.1.1.3	MOS	41

5.1.2 Scenario 2: VoIP Calls in LAN With External Traffic	43
5.1.2.1 Jitter in VoIP Calls	43
5.1.2.2 Packet End-to-End Delay	45
5.1.2.3 MOS	47
CHAPTER 6 CONCLUSION	
6.0 Conclusion	49
6.1 Recommendations	51
REFERENCES	52
APPENDICES	
A1 Configuration of OPNET Modeler	55
A2 Simulation Output	61
A3 Gantt Chart	77

LIST OF TABLES

Table No.	Title	Page
2.1	Comparison of several type of codec	7
2.2	Mean Opinion Score (MOS) values	8
2.3	R-Factor values	9
2.4	Acceptable level of packet loss	11
2.5	MOS Codec Measurement	16
2.6	Comparison between proposed research and existing research	19
2.7	Comparison between existing researches	20
3.1	Methodology of the research	23
3.2	Sample of testing table	28
4.1	Sample of testing table	33
4.2	Sample of testing table	33
5.1	Summary of Scenario 1	50
5.2	Summary of Scenario 2	50

LIST OF FIGURES

Figure No.	Title	Page
2.1	VoIP System Architecture	6
2.2	Jitter	12
2.3	Network Topology of KK2 and KK3	18
3.1	Network Topology of KK2 and KK3 in Simulation Mode	27
4.1	Network Topology of KK2 and KK3	32
4.2	Sample of result in graph	34
5.1	Simulation network topology	36
5.2	Configuration of Number of Clients	36
5.3	Simulation Result of Jitter	38
5.4	Simulation Result of Packet End to End Delay	40
5.5	Simulation Result of MOS	42
5.6	Simulation Result of Jitter	44
5.7	Simulation Result of Packet End to End Delays	46
5.8	Simulation Result of MOS	48
A1.1	Object Palette	55
A1.2	Application Definition for network without traffic	56
A1.3	Application Definition for network external traffic	57
A1.4	Voice Application Configuration	57
A1.5	Profile Definition of network without traffic	59
A1.6	Profile Definition of network with external traffic	60

CHAPTER 1

INTRODUCTION

1.0 Introduction

Many years ago, the people use conventional phone calls which known as Public Switched Telephone Network (PSTN) or also known as Plain Old Telephone Services (POTS) to keep in touch with their family members or friends. Unfortunately, by using PSTN services there are a lot of disadvantages especially is cost of calling. This is because by using PSTN services, there is no free call can be made and the cost of international calling is more expensive which can charge them up to RM100 just for a few minutes. With this reason, the technology has to be improved and during Feb of 1995, a new technology had been invented which is named as Voice over Internet Protocol (VoIP). [1]

VoIP is a technology that allows users to make voice or video calls over the Internet rather than phone companies' network. With this technology, it can help the people to reduce their expenses because of VoIP provide services such as making an unlimited free call no matter is local or international calling. VoIP services use Internet to transport voice from sender by "packetized" it. This mean the voice signal is fragmented into small pieces and send to the network one-by-one. After that, all the small packets will defragment at the other end to form the original voice that sent from sender earlier. In order to use VoIP services, the users can use a computer, smart

phone and phone adapter with telephone to connect to high speed Internet to contact with each other. Unfortunately, not all the VoIP services are provided to computer or phone adapter with telephone. If using a computer, the users are required a microphone and install a VoIP software such as Skype, FaceTime, Google Talk and so on. With VoIP, it helps many companies to make their work more efficient and effective such as make video conference to their main company which usually at oversea to discuss certain urgent issues so that they can handle it fast and effective.

Besides reducing the cost of calling, VoIP also allow the number of persons to speak more than two persons. Unlike traditional phone line which can only provide two persons speak at the same time. By setup a conferences, the whole team members can communicate with each other. With this way, the quality of work in the workstations will increase due to VoIP provide better services compare to PSTN services.

1.1 Problem Statement

In Universiti Malaysia Pahang (UMP), numbers of students are distributed to different Kolej Kediaman. There are about 1500 students in KK2 and 1600 students in KK3. VoIP performance can be affected by several factors such as packet end to end delays and so on, so how many number of concurrent VoIP users can be supported by using the network that provided by UMP between KK2 and KK3 .

Besides that, the quality of VoIP session can be affected when the network is get busy. In UMP, most of the students have no lecture after 8pm so most of them spend their time for entertainment such as watching video on Youtube, download video, plays online games and so on. So, this research also will focus on the maximum concurrent users can be supported when the network with external traffic.

1.2 Objectives

This study was conducted to meet three objectives:

- To study the current network setup between KK2 and KK3.
- To simulate the VoIP session with the network setup on OPNET.
- To analyze the performance based on packet end to end delays, packet jitter and mean opinion score (MOS) and recommend the maximum VoIP concurrent users.

1.3 Scopes

The research is limited to the following matter:

- This research will be tested using LAN architecture environment which is current network setup between KK2 and KK3 only.
- There are many types of factors can affect the performance of VoIP but only packet jitter, packet end to end delay and Means Opinion Score (MOS) will be analyzed.
- The research will be conducted by using software named OPNET Modeler.
- The quality of VoIP in the network will be measured based on the value of MOS.
- The end of this research, the number of concurrent users will be estimated for VoIP calls in network without traffic and network with external traffic.

1.4 Thesis Organization

The research consists of six (6) chapters:

Chapter 1 is about the overview of thesis. Here, problem statement will be introduced here. After that follow by the objective of the research that define based on the problem statement. Scopes of the research will be stated here too.

Chapter 2 is literature review which will introduce more about VoIP. After that, MOS will be introduced here.

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

Chapter 4 designs the network topology that will be used to perform the test in OPNET. The design phase of methodology will be more detail in this chapter.

Chapters 5 explain and discussion of the result. All the results will be produced by the simulator.

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

CHAPTER 2

LITERATURE REVIEW

2.0 How Does VoIP works

When using VoIP, it allows standard telephone signal convert into a digital signal so that it able to transmitted over the internet to the receiver end. Any standard telephone can use VoIP to make a free call just by plugging a special device called analog telephone adapter (ATA). This is because ATA is functioning as a converter to converts the analog signal from telephone. By applying ATA on the telephone, the user can use the internet connection to make telephone calls. Besides using standard telephone for VoIP telephone calls, the users also can use computer or smart phone to make VoIP telephone calls with install the software that supported VoIP services. Furthermore, for the users who are always on the move also can use a wireless “hot spot” in location such as cafe, hotel or airport to make and receive VoIP phone calls. [2]

In more detail, when the sender speaks to the receiver, the voice will be recorded and the computer will compress those voices so that the voice data just require minimum of space. For VoIP, codecs will used to encodes and optimize for compressing voice. After that, the voice that been compressed will pass through a packetizer for converting compressed voice into voice data packet for transmission over the IP network. At the receiver end, the packet will be depacketizer. Playout

buffer will take place for the precautions to check whether any jitter happen that will cause some of the packets stuck somewhere. After that, concealment will activated too to check whether any voice data packets lost. Next, the voice data packets will decoded by the same codec that used before and turn it into an original voice to the receiver so that he can understand the content.

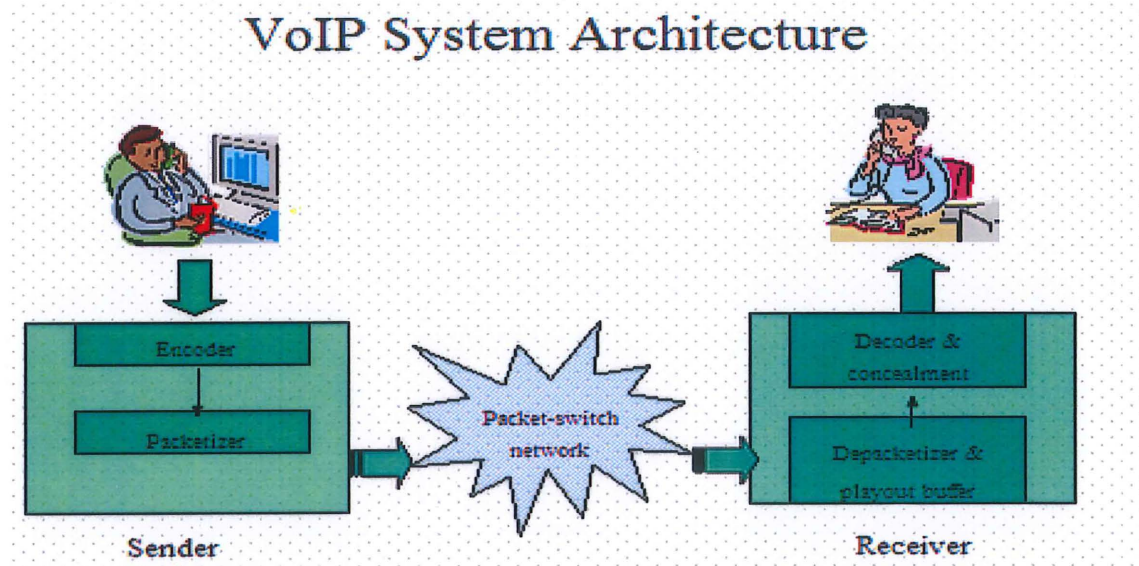


Figure 2.1: VoIP System Architecture

2.1 Codec

A codec is a type of algorithm that acts as software that installed within hardware in order to make it functioning. In VoIP, codec is functioning as a converter which convert voice signal to digital data so that can undergo transmission over the Internet during a VoIP call. After that, the process of decoding will be take place to decode the digital voice data at the receiver end back into its original sound so that the receiver can hear and understand the content. In next table, there are several types of known codecs will be compared.

Note: This table only contains calculations for default voice payload sizes in Cisco CallManager or Cisco IOS® Software H.323 gateways [3].

Table 2.1: Comparison of several type of Codec

Codec Information				Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth MP or FRF.12 (Kbps)	Bandwidth w/cRTP MP or FRF.12 (Kbps)	Bandwidth Ethernet (Kbps)
G.711 (64 Kbps)	80 Bytes	10 ms	4.1	160 Bytes	20 ms	50	82.8 Kbps	67.6 Kbps	87.2 Kbps
G.729 (8 Kbps)	10 Bytes	10 ms	3.92	20 Bytes	20 ms	50	26.8 Kbps	11.6 Kbps	31.2 Kbps
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	3.9	24 Bytes	30 ms	33.3	18.9 Kbps	8.8 Kbps	21.9 Kbps
G.723.1 (5.3 Kbps)	20 Bytes	30 ms	3.8	20 Bytes	30 ms	33.3	17.9 Kbps	7.7 Kbps	20.8 Kbps
G.726 (32 Kbps)	20 Bytes	5 ms	3.85	80 Bytes	20 ms	50	50.8 Kbps	35.6 Kbps	55.2 Kbps

Different type of codecs will obtain different value of MOS. So, the selection of codec must be considered whether it appropriate with the bandwidth of the network. The quality of voice call with selected of certain codec often can be determined by using MOS and R-factor.

2.1.1 Mean Opinion Score (MOS)

In voice and video communication, the quality of VoIP usually is decided whether the experience is either good or bad. Besides that, there are some qualitative descriptions such as ‘fairly good’ or ‘terrible’. In order to make the expression easier, a numerical method is used to state the voice and video quality. Based on ACR subjective test, a numbers of people rate the voice quality of experience on a scale from 1 to 5. 1 is stated as the worst while 5 is the best. The average of scores is called mean opinion score (MOS). MOS was considered as subjective because the result is based on what is experienced by the people and ranked the quality of the call during the tests. Therefore, there are some software applications that use MOS as a measurement for voice quality. The MOS values and level of user satisfaction level are shown in the table below [4].

Table 2.2: Mean Opinion Score (MOS) values

MOS	Quality	Description of Quality Impairment
5	Excellent	Imperceptible
4	Good	Just perceptible but not annoying
3	Fair	Perceptible and slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

2.1.2 R-Factor

R-Factor is a type of alternative methods to assess the quality of voice. The scale of R-factor is from 0 (the worst) to 100 (the best) for measuring the voice quality. With this scale, R-factor is considered as more precise tool to measure voice quality in VoIP. R-Factor can be divided into Network R-Factor and User R-factor. Network R-Factor is generated based on the physical equipment impairments while User R-Factor is additional of perceptual effects to the equipments impairments such as delay and recency [5]. Therefore, some users believe that R-Factor result is more reliable than MOS but in order to make better judgment for call quality, both MOS and R-Factor should be done. The table below shows the R-Factor values and level of user satisfaction [6].

Table 2.3: R-Factor values

User Satisfaction Level	R Factor Value Range
Maximum for G.711 codec	93
Very satisfied	90-100
Satisfied	80-89
Some users satisfied	70-79
Many users dissatisfied	60-69
Nearly all users dissatisfied	50-59
Not recommended	Less than 50

2.2 Typical VoIP Problems

2.2.1 Packet Loss

Packet loss in definition meant a failure of one or more transmitted packets to arrive at their destination. In general, packet loss is happened in every kind of networks. Therefore, each network transport protocols are playing an important role to face this problem with their own way. For example, Transmission Control Protocol (TCP) can provide reliable delivery of packets because during the event of packet loss, if the segments is lost or not yet been acknowledged, it will request from the sender to resend the segments that are missing. In data traffic, if one of the segments of data is missing, the transfer protocol still can request for retransmission, but in VoIP which is very sensitive to packet loss has no time to wait for packet retransmit. The reason is if the voice packets did not arrive as expected time at the receiver, it will assume to be lost. Therefore, it will affect the quality of VoIP and cause some parts of the voice cannot be heard by listeners.

In VoIP, packet loss are usually happens which will affect the performance and can occurs due to several reasons such as some other protocols like User Datagram Protocol (UDP) does not provide recovery for lost packet but must accept the missing, errors or duplication data. According to Colin Perkins, Orion Hodson and Vickey Hardman [7] *“Packet loss can occur due to the following reasons:*

- 1) *Delay in packet transmission from sender to receiver*
- 2) *Packet arriving too late at the receiver side*
- 3) *Heavy loading*
- 4) *Congestion of routers and gateways*
- 5) *The variation of packet inter arrival time which mean the difference between when the packet is expected and when it is actually received is jitter*
- 6) *The loss of voice packets from sender to receiver.*

In VoIP, there are two types of packet loss which known as receive packet loss and receive packet discard. Receive packet loss [8] is where a packet never delivered to

the receiving system which mean that the packet is dropped somewhere during transmitting in the network. Meanwhile, Receive packet discard [8] is where a packet is received at a time when it is not useable for generating audio playback.

In order to reduce or minimize the problem of packet loss in digital communication, one of the techniques is used which known as Packet Loss Concealment (PLC). In PLC, one of the simplest techniques called zero substitution is used to replace the losing speech with zero. With the use simple algorithms, the quality of sound provided usually is the lowest when some parts of the packets are discarded. Another technique named as waveform substitution functioning by substituting the missing frames with artificially generated substitute sound. That mean the artificial substitute sound is created by only repeats the last received packet. Unfortunately, not all the PCL techniques are matches the characteristics of particular environment because packets can be lost by several reasons as mention above.

Since packet loss in VoIP cannot be prevented, there are some researches about level of packet loss that can be acceptable. The table below is show the level of packet loss in percentage and comment [9].

Table 2.4: Acceptable level of packet loss

Level of packet loss	Rating
0 % -1%	Good
1% - 2.5%	Acceptable
2.5% - 5%	Poor
5%-12%	Very poor
>12%	Bad

[9] Based on the observation, above 4-6% packet loss video conferencing has become irritating and unable to communicate with each other. For above 10% -12%, the connections and video conferencing are unstable.

2.2.2 Packet Jitter

In definition, jitter is meant by [10] a variation of packet transit delay which caused by queuing, contention and serialization effects on the path via the network. In every networks, jitter is usually exist and this phenomenon is commonly did not cause any serious problem. This is because with the help of TCP/IP transport protocol, it wills makes the communication better by coping with jitter impact. On the other hand, in VoIP environment, jitter is one of the factors that can affect the quality of conversation. It is totally different compare to network delay, jitter does not happen due to packet delay but because of variation in packet delays. For example, when a person is sending VoIP communication with a normal interval (20ms for one frame), the voice packets that send for transmitting through the packet network might gets stuck somewhere and require longer period in order to reach the destination. This means that one of the packets' timing for arriving to destination is different with the others which known as jitter.

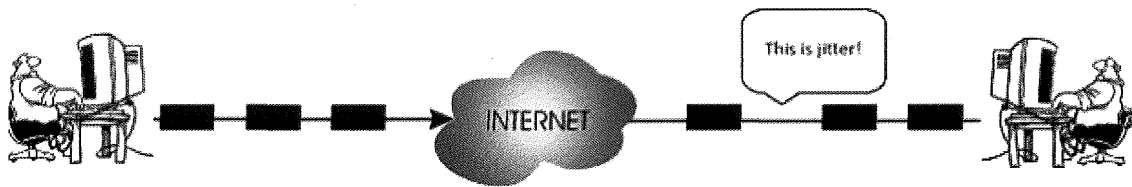


Figure 2.2: Jitter

In order to overcome or minimize this problem, there are several types of techniques can be used. One of the techniques commonly be used is called jitter buffer. In VoIP, jitter buffer [11] which is also known as playout buffers are used to change asynchronous packet arrivals into synchronous stream by turning variable network delays into constant delays at the destination end systems. Jitter can be occurs due to several reasons such as network congestion, types of firewall router used, routing table updates, time drift and so on. Therefore, jitter buffer normally will be located at the destination end of voice connection with the purpose of delaying the arriving packets and waits for the late packets to receive so that the packets will put

together in the proper timing and order. Although the voice packets are able to queue in proper order, the listener still will experience a clear connection with little sound distortion. With this reason, it does not mean that VoIP network will lots of jitter will cause increasing in total delay but in fact it mean the more jitter coping in VoIP network, the bigger jitter buffer is required.

If the jitter buffer is simply set by end user either too big or too small, it might cause unnecessary obstruction on the characteristics of the network. For example, if a jitter buffer set too large, the network will require to supports tighter delay. This is because if the jitter buffer is too large, it will take longer period for the delay. While if the jitter buffer is settled too small to accommodate the network jitter, underflows or overflows of buffer might occur. For underflows, when the coded is needed to play out for a sample, the buffer will empty. Meanwhile for overflows, if the jitter buffer is out of capacity and another packet arrives, then the packet will be dropped and the listener will not able to hear them. Since manually set of jitter buffer size will may cause voice quality degradation, another jitter buffers which known as adaptive jitter buffers will overcome this issue by dynamically changing the size to the most suitable value.

2.2.3 Packet End to End Delays

Packet end to end delay is one of the factors that can affect the quality of voice from sender to the receiver. Packet delay meant the length of time taken for a packet to transmit through the internet. The packets will get delay somewhere when the packets are sent through a network towards destination devices. This is because the reliability features in voice quality mechanism will not waits for a packet with no direction or lost direction. The packet that lost its' direction might come late. For example, when person A send a voice packets "hello" to person B, but due to affected by packet delay, person B might lost a "lo" in a speech. That's mean the receiver will hear the voice later than he suppose to.

If the network delay is exceeding 200 milliseconds (ms), the users will face difficulties in order to carry on a normal conversation. So, if delay is less than 200 ms

and remains constant, the conversation and connection is still can be acceptable for many users. [11][16] Unfortunately, the delay will not always staying constant and varies are also depending on some technical factors. The variation of delay which called jitter is one type of delay and it also can cause echo in VoIP calls. In a network, the addition of switches, routers, firewalls and jitter buffers will cause packet delay. Besides that, the packets delay happens in routers are not depending on hardware, but also depends the configuration that configure by the network engineer. The configurations include access lists, queuing methods and transmission modes. Furthermore, the distance travelled which mean the number of router hops that packets required to arrive the destination also can cause delay.

2.3 Existing Research

2.3.1 A Comparison of VoIP Performance Over Three Routing Protocols for IEEE 802.11s-based Wireless Mesh Networks (WLAN Mesh) [12]

In this research, it is about the investigation of wireless mesh networks (WMNs) capacity to support its' quality of service (QoS) before VoIP service deploying on it. When using VoIP, there are several factors such as end to end delay, packet loss and overhead that can affect the QoS of VoIP. Therefore, comparison of VoIP performance will use three types of different routing protocols in 802.11s-based WMN. The routing protocols that will considered in this comparison are: Adhoc On-demand Distance Vector Routing (AODV), Dynamic Source Routing (DSR) and Hybrid Wireless Mesh Protocol (HWMP) which is defined by the 802.11s group for the IEEE 802.11s standard.

The comparison of three protocols was experimented in the simulation based on Network Simulator 2 (NS2). The authors modified NS2 to implement HWMP protocol and Airmetric. The network was consisted maximum 50 nodes that placed uniformly at a distance from each other. These nodes were undergone VoIP connections with each other and no other background traffic was applied. Each node

has 2Mbps link while for default metrics was used by Airtime link metrics. The total simulation time was 300 seconds and each VoIP call only can last for 20 seconds. Besides that, the types of voice coders used were G711 and H323.

As a result, HWMP has better performance if compared to DSR and AODV when using VoIP application. However, HWMP will become more overhead when the number of nodes increased. The graph of End to End delay showed that it is nearly same as AODV. For HWMP, it did not choose the best path for every case. The path is selected based on next hop cost but not the entire path cost. In HWMP, RREQ packets are broadcasted to the network in order to find the best route to destination and finally the next best low metrics value is selected but unfortunately it did not always guarantee a low metrics value path.

2.3.2 Analyzing of MOS and Codec Selection for Voice over IP Technology [13]

In this research, it is about the analysis of VoIP Codec Selection and investigation of Mean Opinion Score (MOS) performance areas with the changes with quality of services (QoS) that received by soft phone and IP phone. The objectives of this study are on the quality of voice prediction such as accuracy of MOS between automated system and human perception and another one is different types of codec performance measurement through human perception based on MOS technique. Network management system (NMS) is used for monitoring and capturing the VoIP performance in campus environment.

This research was carried on in the real network environment to analyze and measure the implementation of VoIP service in a campus. In order to measure the quality of VoIP, VQnet (VoIP) management tools are used to collect the quality variation information, avoid the ignorance of unacceptable VoIP quality that caused by network failure or bandwidth bottleneck. IP phone and multiple soft phones performance measurement were implemented MOS technique. Some examples of soft phones that used in this study to evaluate and measurement of VoIP performance using MOS technique are Wengo Phone, 3CX, PhoneLite, Kapanga and SJPhone.

As a conclusion of this research, the measurement of MOS through LAN and WAN with different types of Codec shown that WAN can cause higher delay, packet loss, and CPU usage compare to LAN in campus environment.

Table 2.5: MOS Codec Measurement

Users	Codecs	MOS in LAN	MOS in WAN
User 1	G711 aLaw	4	3
	G711 uLaw	4	3
	iLBC	3	2
	Speex	3	2
User 2	G711 aLaw	4	3
	G711 uLaw	3	2
	iLBC	2	1
	Speex	3	2
User 3	G711 aLaw	3	2
	G711 uLaw	4	3
	iLBC	2	1
	Speex	3	2
User 4	G711 aLaw	3	2
	G711 uLaw	4	3
	iLBC	3	2
	Speex	3	2
User 5	G711 aLaw	4	3
	G711 uLaw	3	2
	iLBC	2	1
	Speex	2	1

2.3.3 Performance Studies of VoIP over Ethernet LANs [14]

This research is study about the performance of VoIP traffic characteristics over Ethernet LANs. In this investigation, the performance of VoIP must considered by several factors such as the increasing of number of VoIP clients, the selection of voice codec schemes and traffic distribution on system. This research will investigate through various experiments under realistic networking scenarios, such as small offices home offices (SOHO) and campus network.

This research consists of four different scenarios. First scenario is the impact of increasing number of VoIP clients. This experiment is tested because when there is only one node in an Ethernet LAN, the transmission rate of the Ethernet LAN can use up to the maximum rate which is 100 Mbps, but after increasing the number of nodes, the effective of transmissions rate getting lower. Scenario two is the impact of wireless nodes. WLAN is test due to many organizations are using WLAN within their networks. So, it is a good reason to investigate VoIP performance limitations over WLAN since WLAN is more convenient. WLAN is tested by using IEEE 802.11b and 802.11g. Next scenario is the impact of voice encoder schemes. In this scenario, different types of codec schemes such as G.711, G.723 and G.729 are tested to measure the VoIP performance and voice quality. This is because the process of compression and decompression of voice signals have negative impacts of voice quality. Last scenario is the impact of traffic arrival distribution. In this scenario, the VoIP calls will be generated in constant fixed rate or have fix length of VoIP calls in a real network. VoIP performance will be assessed based on Poisson distributed, exponential distribution and constant distribution.

As a conclusion, the major factors that affected VoIP quality are delay, jitter and packet loss. The simulation results presented in this dissertation can give a better understanding to organizations how well VoIP can be perform on local network prior to adopt VoIP. It also helps researchers and designers to design network for VoIP deployment.

2.4 Comparison between Existing Researches

2.4.1 Proposed Research

This research is about the analysis of VoIP performance in Kolej Kediaman (KK) 2 and 3 in University Malaysia Pahang. In the diagram below is the network topology between KK2 and KK3.

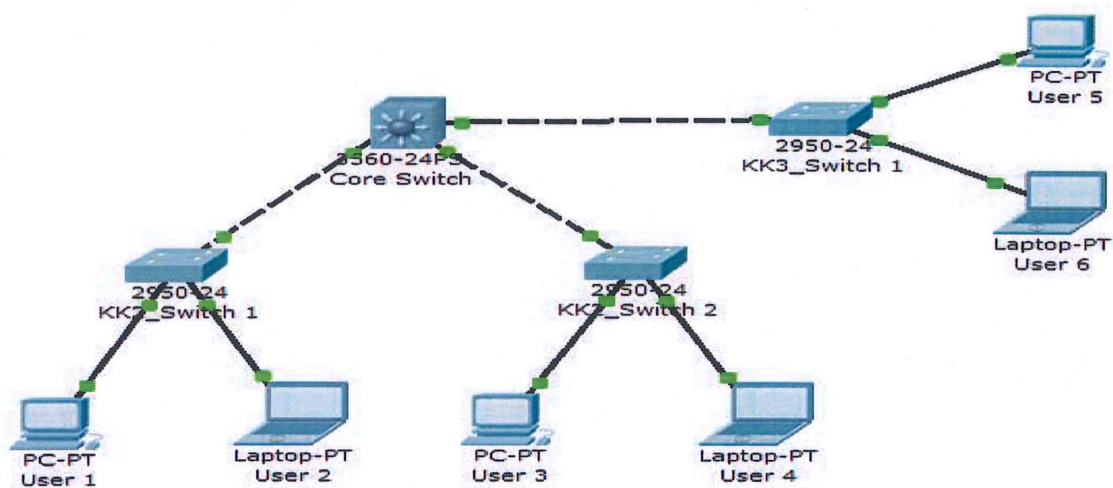


Figure 2.3: Network Topology of KK2 and KK3

The topology in this research is using LAN architecture. In KK2, there are two layer 2 of distributed switches for connecting to end users and in KK3 only used one layer 2 distributed switch. Three of the distributed switches will connected to layer 3 core switch with 10 Gbps of bandwidth while for the bandwidth between distributed switch and end users is 1 Gbps. In this research, the codec for supporting VoIP QoS will be used is G.711 which is popular and other research has proved that the best codec among all the codecs. The maximum number of concurrent users can be supported by this network will be estimated based on jitter and packet end to end delay.

2.4.2 Comparison between Proposed Research and Existing Research

Table 2.6: Comparison between proposed research and existing research

Proposed Research		Existing Research
Similarity		
<p>1) The research use OPNET Modeler to simulate the results.</p> <p>2) Jitter, packet end to end delays and MOS are observed.</p>		
<p>1) Implementation of VoIP in a LAN network.</p> <p>2) CODEC G.711 is used and fixed during the research.</p> <p>3) No routing protocol is used since in a LAN network.</p>	Differences	<p>1) Implementation VoIP in LAN, and WLAN network.</p> <p>2) Different types of CODEC are used to get different results of VoIP performance.</p> <p>3) Various types of protocol used to VoIP performance.</p>

2.4.3 Comparison between Existing Researches

Table 2.7: Comparison between existing researches

Research title	Author	Method	Result/Analysis
Comparison of VoIP Performance Over Three Routing Protocols for IEEE 802.11s-based Wireless Mesh Networks (WLAN Mesh)	- Adlen Ksentini - Obaid Abassi	Evaluating the VoIP performance using AODV, DSR, HWMP routing protocol based on several factors such as delay, packet loss and overhead.	- HWMP has more advantages than other MANET protocols (AODV and DSR). - AODV not very effective because it use hop-count metrics. - Layer 2 routing is more effective in IEEE 802.11s compare to traditional routing protocols
Analyzing of MOS and Codec Selection for Voice over IP Technology	- Mohd Nazri Ismail	Evaluating the quality of voice based on MOS by automated system and human perception, the use of different types of codecs such as G.711 aLaw, G.711 uLaw, iLBC, Speex.	-MOS system, both automated and manual measurement produce good performance evaluation. -LAN has higher quality of voice than WAN.
Performance Studies of VoIP over Ethernet LANs	- Di Wu	1) Impact of increasing number of VoIP clients	1) The number of VoIP clients increases, the impact of the VoIP performance over the network becomes

		<p>2) Impact of Wireless Nodes (802.11b, 802.11g)</p> <p>3) Impact of voice encoder schemes (G.711, G.723, G.729)</p> <p>4) Impact of traffic arrival distributions</p>	<p>significant.</p> <p>2) 802.11g protocol gives a better performance than 802.11b but it still cannot support VoIP services under this network as the packet loss ratios are too high.</p> <p>3) G.711 codec scheme is a good choice to the VoIP network model with 20 VoIP clients.</p> <p>4) VoIP services can be performed best under exponential traffic distribution.</p>
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CHAPTER 3

RESEARCH METHODOLOGY

3.0 INTRODUCTION

Research methodology is defined as a systematic way that used to collect relevant information and data for research projects. It also can be defined as a process that how the researcher to solve certain unanswered problems and also study how the research can be conducted. In the research, the collection of relevant data and information can be done in theoretical procedures, through the study of others related experimental or research, interviews, surveys and so on. After gathering the collection of data and information, the analysis and interpretation will be involved. Therefore, in order to ensure the objectives of this research can be achieved and the research methodology in the right path, there are a few guidelines need to be followed. The respective phases are stated as the following:

- I. Collection of relevant information and data about this research.
- II. Installation and configuration of the software which is required in this research.
- III. Design the network topology in simulation mode based on the real network in the software tool.
- IV. Run the test in the software and obtain the result.

- V. Gather the graph and data from the respective experiment and analyze the result.

Table 3.1: Methodology of the research

INFORMATION GATHERING	PRELIMINARY STUDY PHASE	
	PRELIMINARY STUDY	Offline: Books, Journals, Thesis Projects, Research Papers, Appointments Online: Articles, Journals, Proceeding materials
PLANNING AND IDENTIFYING HARDWARE AND SOFTWARE TOOLS	RESEARCH PLANNING PHASE	
	HARDWARE	Laptop
	SOFTWARE TOOLS	OPNET Modeler 14.5 Microsoft Visual Studio 2012
SOFTWARE CONFIGURATION AND EXPERIMENT DESIGN	ARCHITECTURE DESIGN PHASE	
	INSTALLATION	Software tools
	SETTINGS AND CONFIGURATIONS	OPNET Modeler 14.5
	ARCHITECTURE (SIMULATION MODE)	1. One core switch 2. Two distributed switches 3. Three access switches 4. Three 1000 Base X LANs 5. One Profile Definition 6. One Application Definition
IMPLEMENTATION AND SIMULATION	TESTING PHASE	
	<ul style="list-style-type: none"> To test the current network setup between KK2 and KK3 by increasing the number of VoIP user in the network topology without traffic and with traffic. 	
DATA ANALYSIS	DATA ANALYSIS PHASE	
	<ul style="list-style-type: none"> Things to consider are the results of jitter, packet end to end delays and MOS generated by OPNET Modeler. 	

3.1 Overview of Research Methods

In this research project, there are five main phases that defined to be used to ensure this research can be done and completed. The methodology of this research is performed step by step accordingly which are stated as below:

- I. Preliminary Study Phase
- II. Research Planning Phase
- III. Architecture Design Phase
- IV. Testing Phase
- V. Data Analysis Phase

3.1.1 Preliminary Study Phase

This is the initial step of the research which is searching for the relevant and appropriate information that can solve the problem statement of this research. This step is taken to ensure that the research project will undergo the correct way but not out of the topic. Besides, searching for information can increase the knowledge of the title focus on and provide some solutions how to solve the problem statement of the research project. The searching and gathering of information can be done through the study of multiple resources such as research journals, books, articles, case studies and other online materials.

Next, the network topology that will be used in this research project is real network topology. Therefore, the appointments with PTMK staff to obtain the real network topology are required. In the appointments, the network topology's information is explained and provided by the staff such as model of switches used, current bandwidth and types of cable used.

3.1.2 Research Planning Phase

The second phase will be research planning phase. In order to ensure the research project carried out successfully and smoothly, the research project need to be planned carefully. In this phase, it will ensure that the defined steps and activities will guide the flow of this research project in order to obtain the required data for analysis at a later stage in this research project. The planning of this research must be made carefully so that the time is sufficient for the research to be completed successfully. The necessary hard and software tool also needed for the research project to be carried out is defined in this phase.

In this phase, there is necessary hardware and software require for the research project to be carried out. The hardware in this research project require is a laptop or computer and the software is Opnet Modeler 14.5. Next, the information of installation for Opnet Modeler 14.5 is required because the installation is not as simple as others software. In addition, the study of how to use the Opnet Modeler 14.5 is needed too because the software can be implemented for many scenarios and most of the components can be manipulated for different scenarios. So, in order to design the network topology in simulation mode, the skills for using this software is required too.

3.1.3 Architecture Design Phase

In this phase, the network topology will be setup based on the network topology that provided by PTMK staff. This network topology will setup in simulation mode by using the software which is Opnet Modeler 14.5. The network topology in simulation mode is as below:

- i. A laptop that contained Opnet Modeler 14.5
- ii. The network topology in simulation mode is as below:
 - ✓ One core switch
 - ✓ Two distributed switches
 - ✓ Three access switches
 - ✓ Two 1000 Base X LANs which consist of 1500 end users in KK2 and one Two 1000 Base X LAN with 1600 end users in KK3.
 - ✓ One Profile Definition
 - ✓ One Application Definition
 - ✓ 3 UTP 1 Gbps link
 - ✓ 3 Single Mode Fiber in 1 Gbps
 - ✓ 2 10 Gbps Fiber

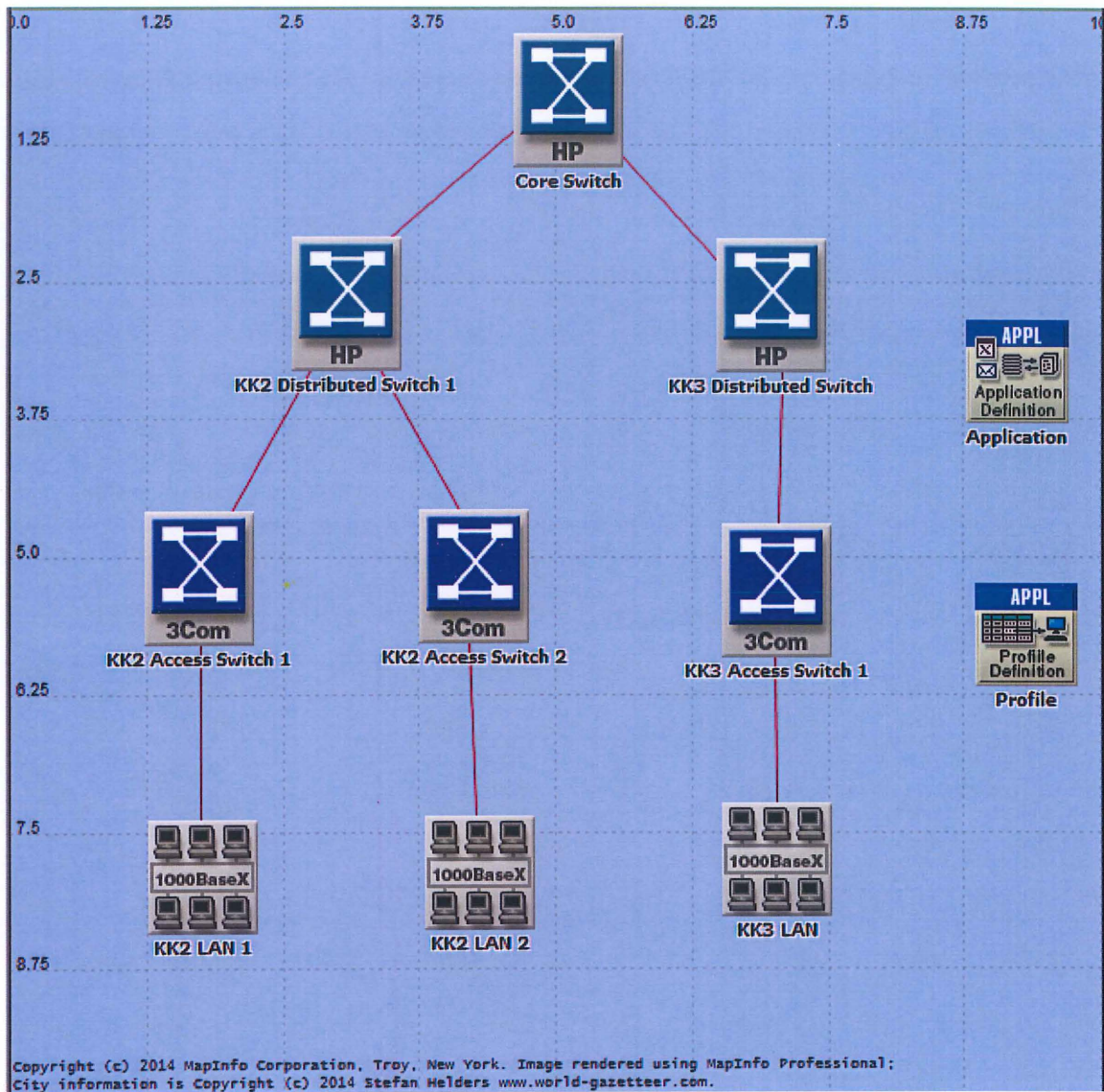


Figure 3.1: Network Topology of KK2 and KK3 in Simulation Mode

3.1.4 Testing Phase

After the architecture design phase, the testing phase is needed to be implemented to test the simulation and obtain the result in graph. In the testing phase, the number of users will be increased gradually as the bandwidth is remains the same. With the increase the number of VoIP users, the factors such as packet end to end delay and jitter will increase while the value of MOS will decrease.

Table 3.2: Sample of testing table

Number of Clients	Jitter	Packet ETE Delay	MOS

3.1.5 Data Analysis Phase

Data analysis phase is a step where the data that can be obtained or gathered from the experiments and the data or result are used to analyzed and interpreting. This is very vital to the research project because interpretation and analysis of data can support researchers to make the final decision and conclusion of the research project. Besides that, the researcher can get the result whether they achieve the objectives of the research or solve the problem of the research. In this research, the analysis of several factors of VoIP is very important analytical report for making the conclusion of this research.

CHAPTER 4

DESIGN AND IMPLEMENTATION

4.0 INTRODUCTION

In design and implementing phase, it is very important to plan and state all the procedures in detail and carefully so that the outcome of this research project can be achieved its objectives. In this phase, it must be defined according based on the objectives that stated in the beginning, this is to make sure that this research project can reach the main objective at the end of the research. In this research, the network topology setup will use wired network which means all the network components are connected with cable or fiber. Based on the methodology's design phase, there is a requirement for a laptop that can be used for running Opnet Modeler 14.5 with smoothly. This is because Opnet Modeler 14.5 is not easily to install and sometimes it can appear with several errors due to wrong installation methods or some mistakes such as in designing and configuring of network components.

4.1 EXPERIMENTAL NETWORK DESIGN

In this phase, the installation and configuration of software settings are done and have been tested can be running smoothly without any errors. Therefore, the network topology will be setup in the simulation mode by using Opnet Modeler 14.5. The network topology that setups in simulation mode consist of network components as shown as the following:

- ✓ One core switch
- ✓ Two distributed switches
- ✓ Three access switches
- ✓ Two 1000 Base X LANs which consist of 1500 end users in KK2 and one 1000 Base X LAN with 1600 end users in KK3.
- ✓ One profile definition and one application definition
- ✓ 3 UTP 1 Gbps link
- ✓ 3 Single Mode Fiber in 1 Gbps
- ✓ 2 10 Gbps Fiber

4.2 EXPERIMENT ARCHITECTURE

In this network topology, the network scale for the simulation mode that has been chosen for this research is Campus since this network topology is Universiti Malaysia Pahang (UMP) real network topology. After that, the size of map that used to design the network topology for simulation is 10 Kilometers for both X-span and Y-span.

In this topology, the number of end users that requested from Kolej Kediaman (KK) which including staff, foreign lecturers and students in KK2 is 1500 while for KK3 consist of 1600 end users. As shown in the Figure 4.1 below, the core switch of network topology currently is connected with one distributed switch in KK2 and one distributed switch in KK3 with the bandwidth of 10Gbps. The connection that used to link these distributed switches and core switch is fiber single mode. After that, the distributed switches connected with another switch which is known as access switch. The connection that used to link these two different switches is fiber optic. The bandwidth that provided for the connection between distributed switch and access switch is 1Gbps. After that is the access switches will connect to LAN. In this topology, there are two access switches in KK2. Therefore, each LAN in KK2 will consist of 750 end users. The bandwidth that used to connect access switches and LAN is 1Gbps by using twister pairs cable.

In this network topology, a Hewlett-Packard's ProCurve 8000M Ethernet Switch is used because in Opnet Modeler 14.5 there is no a 6500 series switch with the brand of Cisco. Therefore, in order to replace Cisco 6500 series, another brand is used. This switch is chosen because of it is able to support up to 10Gbps which is required for the core switch. While for distributed switch, the original switch which is Cisco catalyst 4500 also cannot be found in Opnet Modeler 14.5, so Hewlett-Packard's ProCurve 4000M Ethernet Switch is used. This brand of switch is chosen due to it can support up to 10Gbps for connection. The switch also fulfill the requirements which are able to support 10Gbps for the connection between core switch and distributed switch and 1Gbps connection for distributed switch and access switch. Next, a 3Com's SupterStack II Switch 3900 is used as access switch which connected with distributed switch and end users. This switch is selected for access

switch because it is the same brand as the original switch that used in UMP and it also consists of 3 ports that can support to 1Gbps. Therefore, this switch has reached the minimum requirements because in this network topology, an access switch must support 1Gbps in order to connect with distributed switch and end users in LAN.

Since there are two access switches in KK2, so the LAN is separated into two. This meant each LAN in KK2 consists of 750 end users. While for KK3 LAN, it is still remains for one LAN which consists of 1600 end users. In this topology, an application definition and a profile definition are required for the configuration of VoIP application.

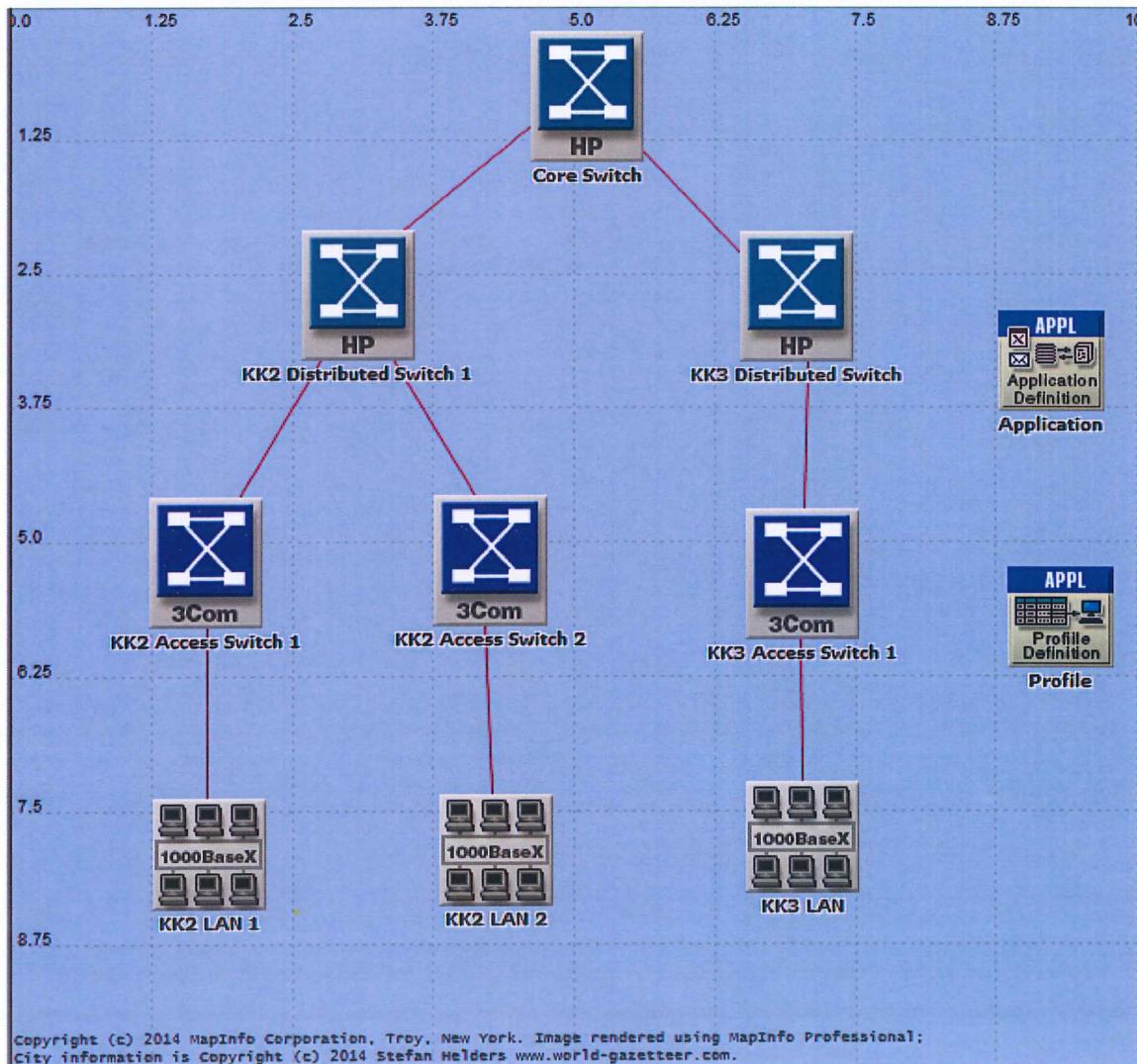


Figure 4.1: Network Topology of KK2 and KK3

4.3 TESTING PLAN

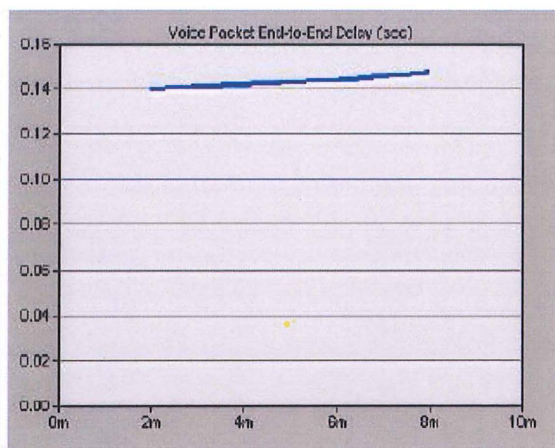
After the architecture design phase, the testing phase is needed to be implemented to test the simulation and obtain the result in graph. In the testing phase, the number of users will be increased gradually as the bandwidth is remains the same. With the increase the number of VoIP users, the factors such as packet delay and jitter will increase while the MOS value will decrease. There are two types of scenarios can be tested which are the current UMP network topology without any traffic and another scenarios is with the addition of HTTP and video streaming external traffic.

Table 4.1: Sample of testing table

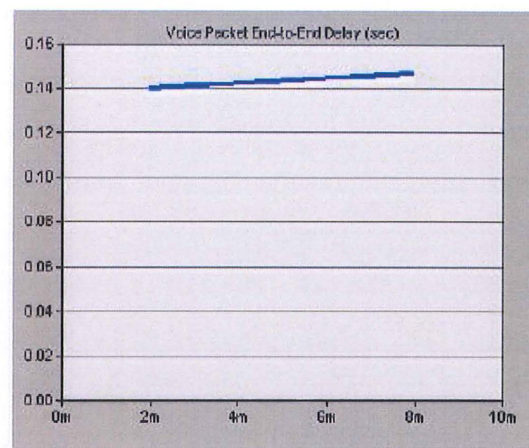
Network Topology – Without Traffic			
Number of Clients	Jitter	Packet ETE Delay	MOS
200			
400			
600			
700			
800			
1000			

Table 4.2: Sample of testing table

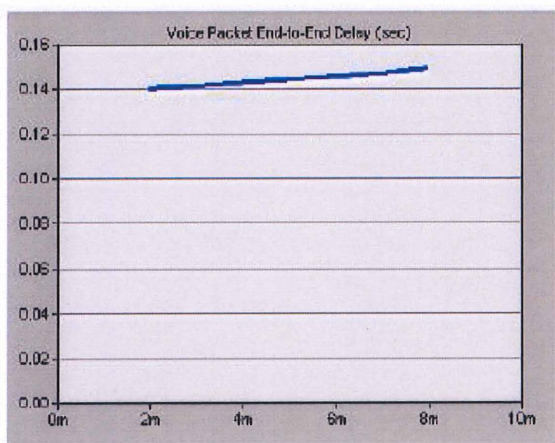
Network Topology – With HTTP and Video Streaming External Traffic			
Number of Clients	Jitter	Packet ETE Delay	MOS
200			
400			
600			
700			
800			
1000			



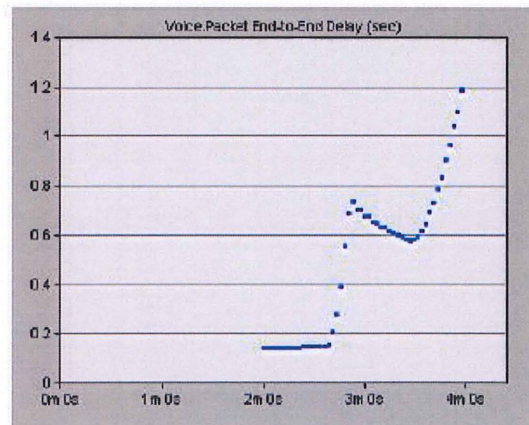
(g) End-to-End Packet Delay
(Wired Nodes = 40)



(h) End-to-End Packet Delay
(Wired Nodes = 80)



(i) End-to-End Packet Delay
(Wired Nodes = 120)



(j) End-to-End Packet Delay
(Wired Nodes = 200)

Figure 4.2: Sample of result in graph

CHAPTER 5

RESULT AND DISCUSSION

5.0 Introduction

This research was completed in order to estimate the number of VoIP concurrent users with the current network topology. The estimation of VoIP users is based on three factors which are jitter, packet end to end delays and MOS. The experiment was done in simulation mode so that can different types of scenarios can be done.

5.1 Simulation Result and Comparisons

5.1.1 Scenario 1: VoIP calls in LAN without traffic

The first scenario is tested the performance of VoIP calls in LAN that have no external traffic. The VoIP calls will be made from KK2 LAN 1 and KK2 LAN 2 to KK3 LAN. The number of client starts from 200 which divided into half for KK2 LAN and KK3 LAN. The number of client for both sides will increase up to 1000. Each testing was started conducting with 100 VoIP sessions. [15]

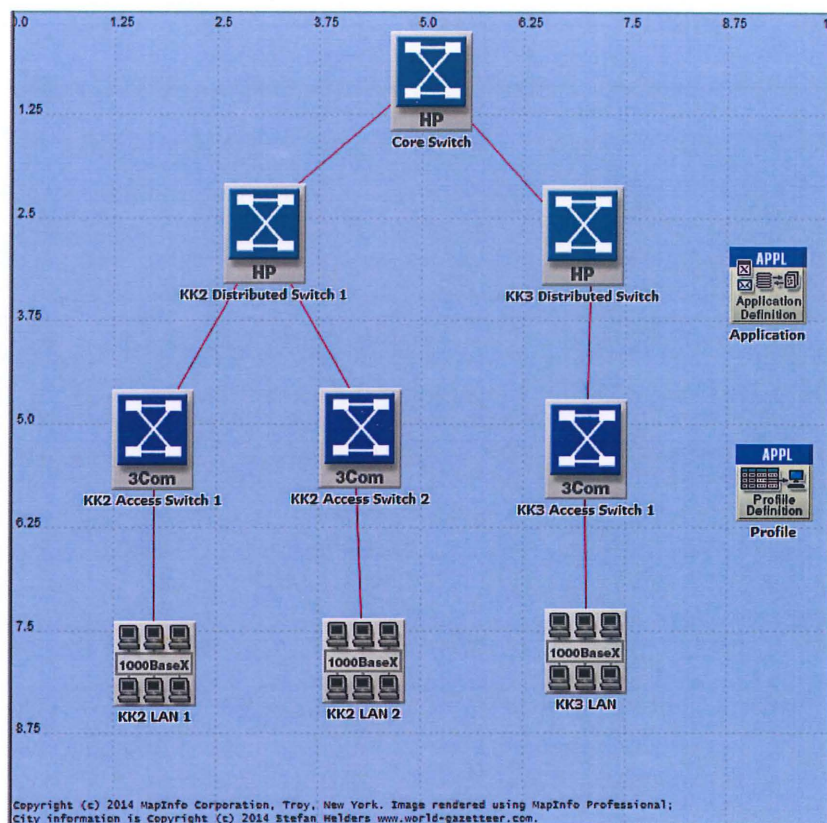


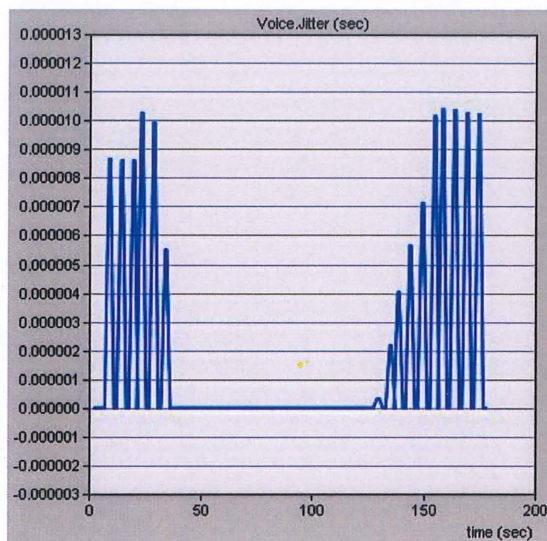
Figure 5.1: Simulation network topology

Application: Supported Profiles	(...)	Application: Supported Profiles	(...)
Number of Rows	1	Number of Rows	1
VoIP Profile		VoIP Profile	
Profile Name	VoIP Profile	Profile Name	VoIP Profile
Number of Clients	50	Number of Clients	100
Traffic Type	All Discrete	Traffic Type	All Discrete
Application Delay Tracking	(...)	Application Delay Tracking	(...)
Start Time (seconds)	Start of Simulation	Start Time (seconds)	Start of Simulation
End Time (seconds)	End of Simulation	End Time (seconds)	End of Simulation
Sample Every N Applications	All	Sample Every N Applications	All
Maximum Samples	Tracking Disabled	Maximum Samples	Tracking Disabled
Application: Supported Services	All	Application: Supported Services	All

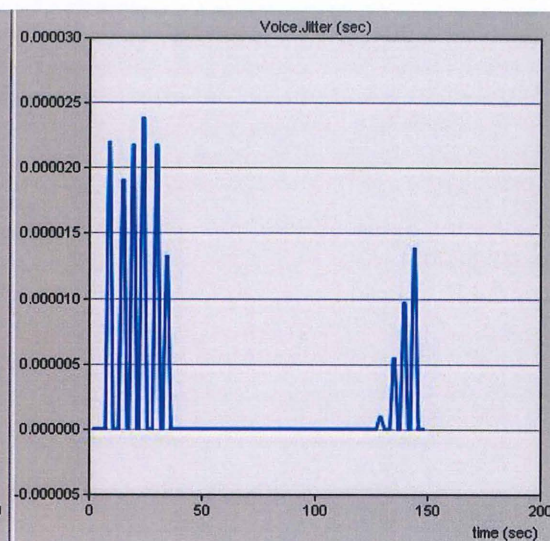
Figure 5.2: Configuration of Number of Clients

5.1.1.1 Jitter in VoIP calls

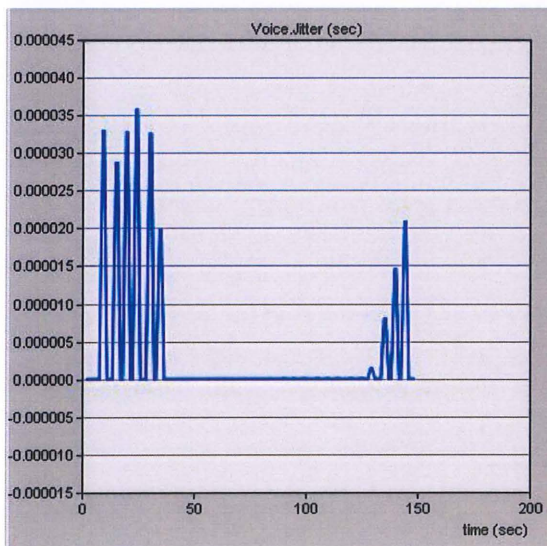
Threshold value of jitter = 0.0075ms



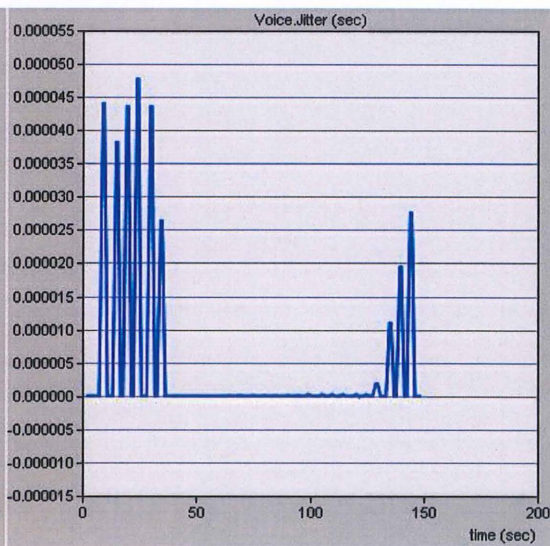
a. 200 Clients



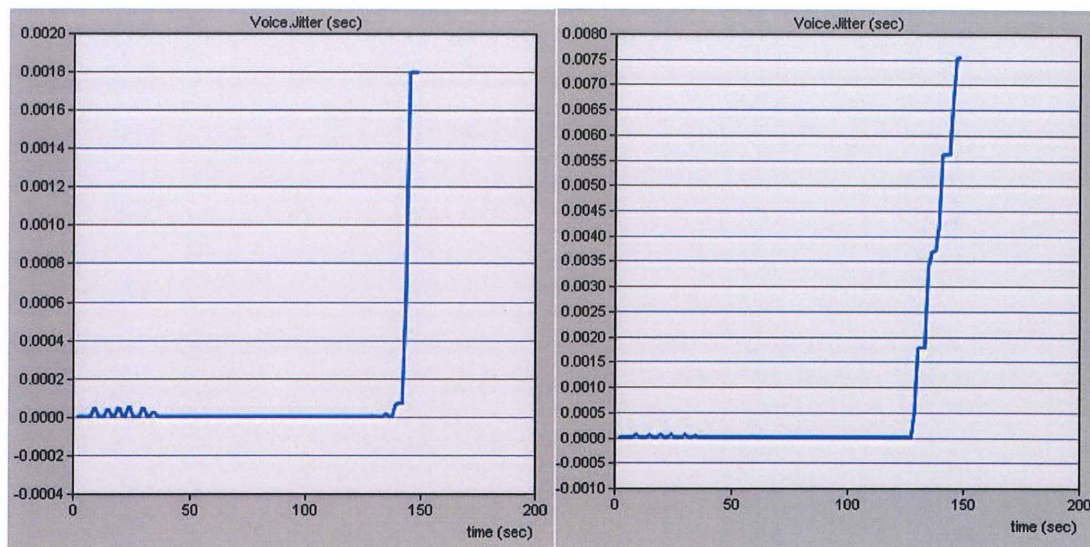
b. 400 Clients



c. 600 Clients



d. 800 clients



e. 900 clients

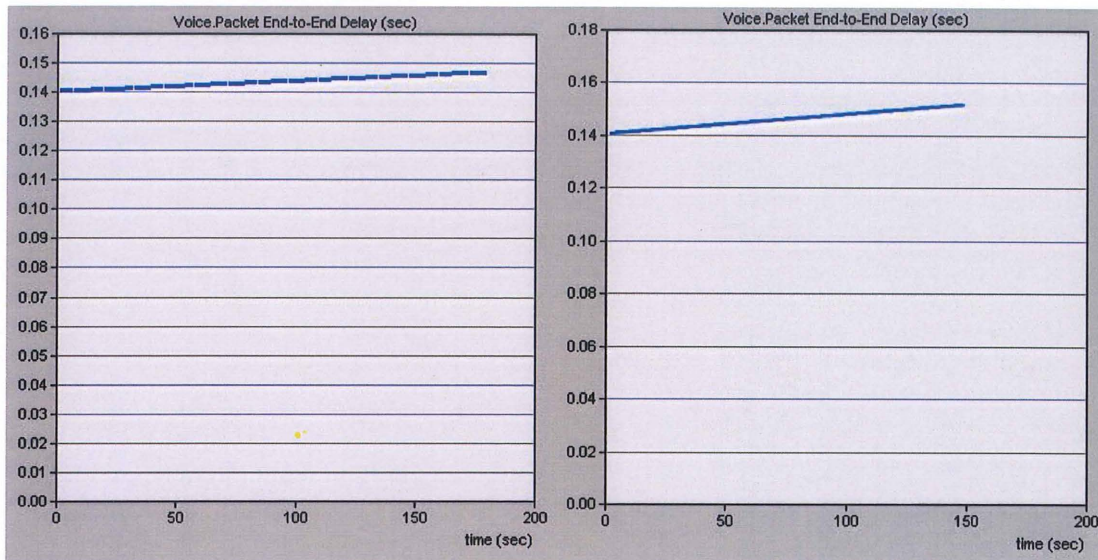
f. 1000 clients

Figure 5.3: Simulation Result of Jitter

From Figure 5.3, the value of jitter increases as the number of client increases. From the number of client from 200 to 800, the value of voice jitter is very small and the highest value is 0.0000479 seconds only while the maximum acceptable voice jitter is 0.075 seconds. However, when the number of client increases to 900, the voice jitter is clearly showed that largely difference compare to the previous number of clients. The value of voice jitter increases dramatically to 0.001796 seconds but still can be accepted. When the number of client increases to 1000 users, the value of jitter showed largely difference compare to 900 users. The value of jitter increased to 0.007532 seconds in 150 seconds of simulation time. In conclusion, the packet jitter was below the threshold value when tested for 1000 users. Therefore, with the current setup, the number of VoIP clients can go up to 1000 users.

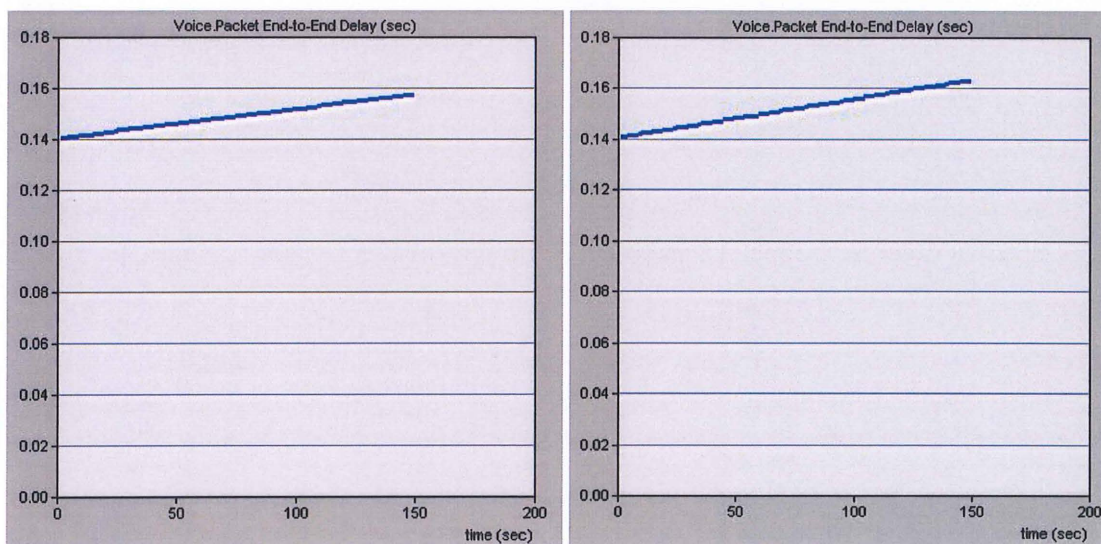
5.1.1.2 Packet End to End Delay

Threshold value of packet end to end delay = 200ms



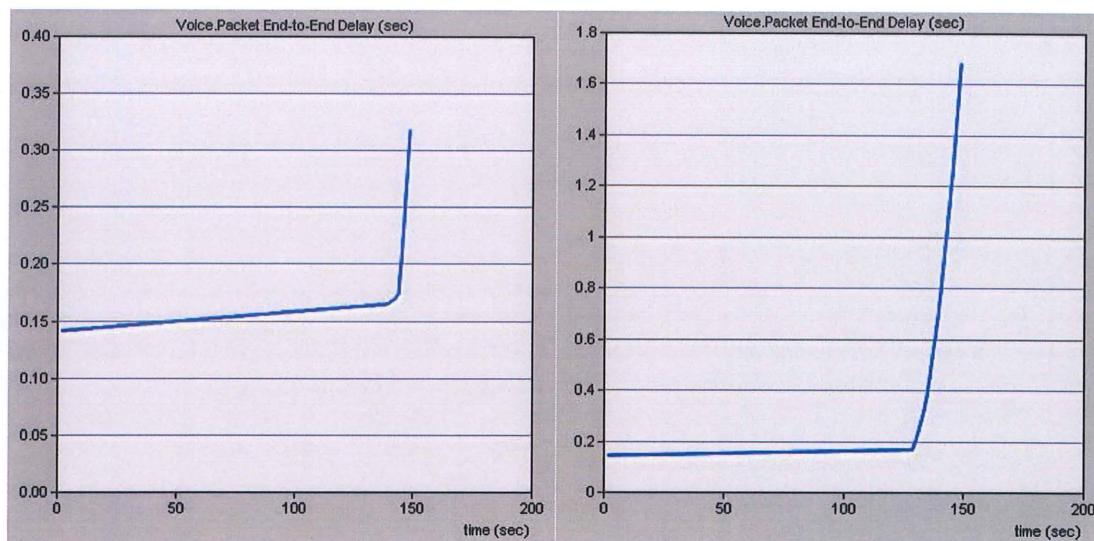
a. 200 clients

b. 400 clients



c. 600 clients

d. 800 clients



e. 900 clients

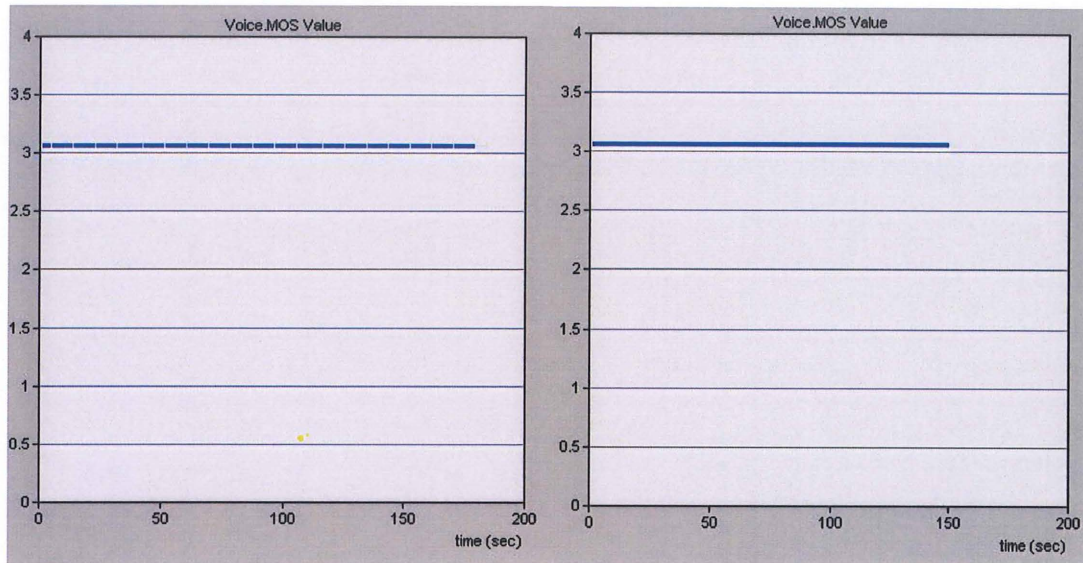
f. 1000 clients

Figure 5.4: Simulation Result of Packet End to End Delay

From Figure 5.4, the value of packet end to end (ETE) delay increases as the number of client increases. Starting from 200 to 800 clients, the value of packet ETE delay increased gradually when the simulation time increased but it did not reach the threshold value of packet ETE delay which is 0.2 seconds. As the number of client increased to 900 clients, the graph showed the value of packet ETE delays increased gradually until 144 seconds. After 144 seconds, the value increased dramatically to 0.31838 seconds until the end of the simulation time. The threshold value was reached when after 1.5 seconds, which meant at 145.5 seconds the value of packet ETE delays exceeded 0.2 seconds. When tested for 1000 clients, logically the graph showed that the value of packet ETE delays exceeded 0.2 seconds earlier which at 130.5 seconds of simulation time. In conclusion, the packet ETE delay was above threshold value when tested for 900 users. Therefore, with the current setup, the number of clients should less than 900.

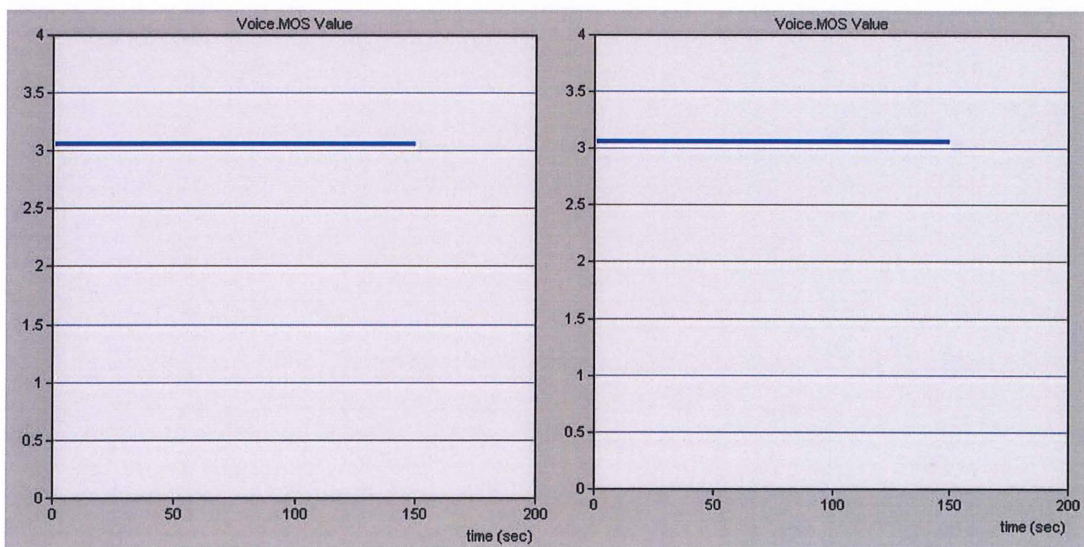
5.1.1.3 MOS

Threshold value of MOS = 3



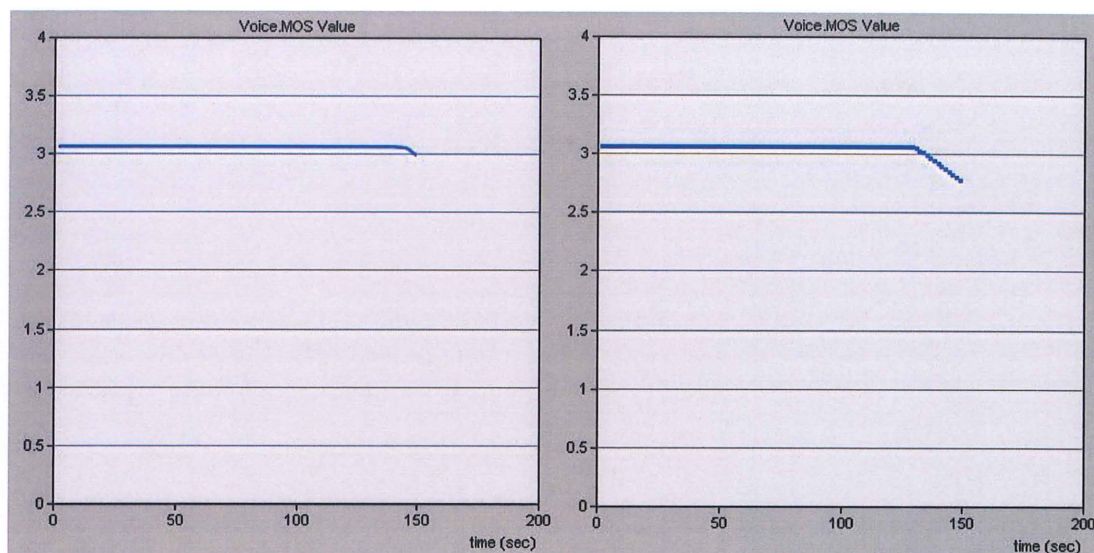
a. 200 clients

b. 400 clients



c. 600 clients

d. 800 clients



e. 900 clients

f. 1000 clients

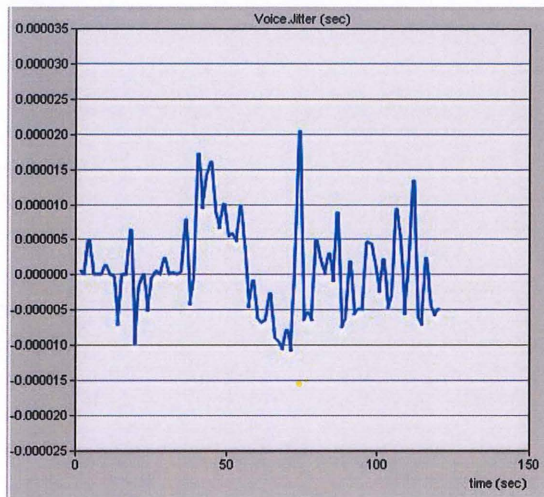
Figure 5.5: Simulation Result of MOS

From Figure 5.5, the value of MOS decreases as the number of client increases. Starting from 200 until 800 clients, the value of MOS showed a straight line at the value of 3.0599 after 150 seconds of simulation time. This showed that the quality of VoIP is very stable and satisfied. As the number of client increased to 900 clients, the graph remained a straight line until 145.5 seconds of simulation time. After that, the MOS value drop slightly to 2.9943 which is in the range of unacceptable value. This showed that the number of clients can be supported by the current bandwidth for 145.5 seconds and the quality will start degrading after 145.5 seconds. While for the simulation of 1000 clients, the MOS value started drop at 135 seconds of simulation time. In conclusion, the MOS was under the threshold value when tested for 900 users. Therefore, with the current setup, the number of clients should less than 900.

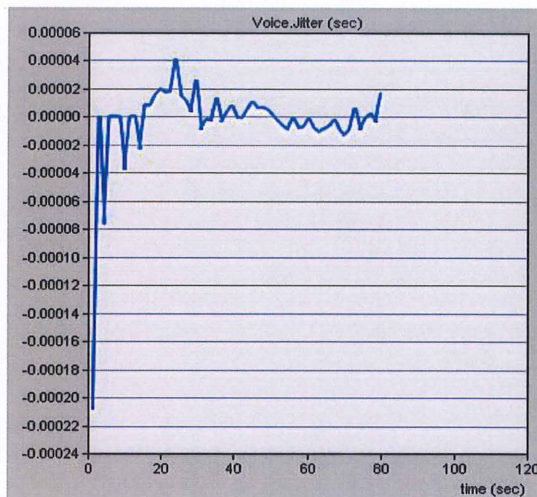
5.1.2 Scenario 2: VoIP calls in LAN with external traffic

5.1.2.1 Jitter in VoIP calls

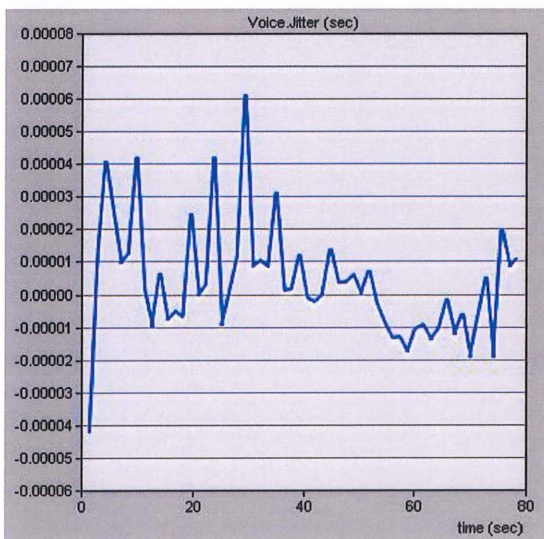
Threshold value of jitter = 75ms



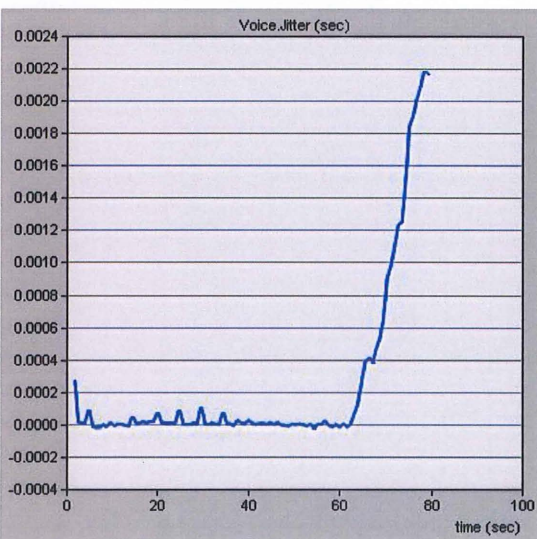
a. 200 clients



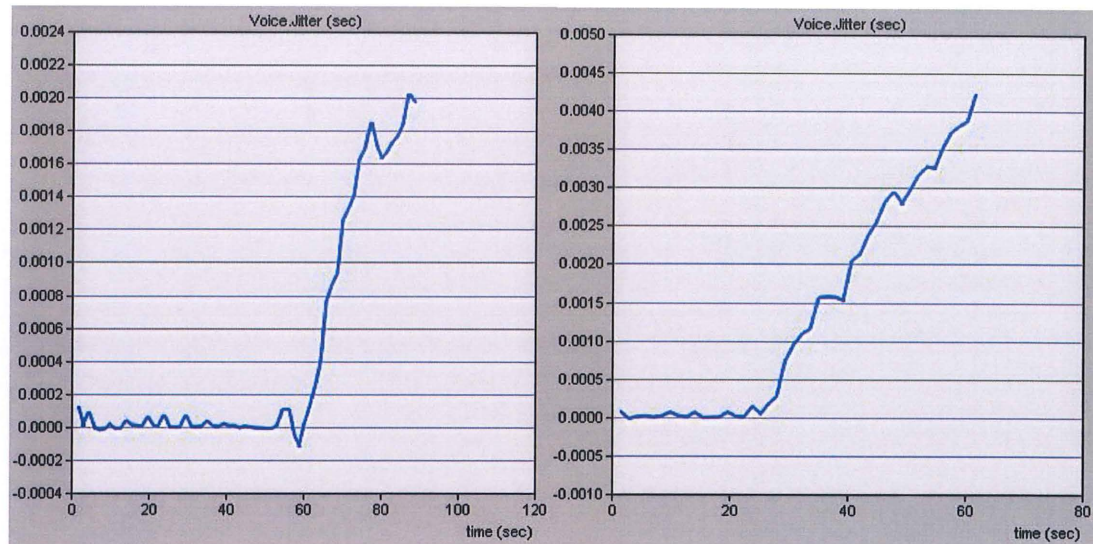
b. 400 clients



c. 600 clients



d. 700 clients



e. 800 clients

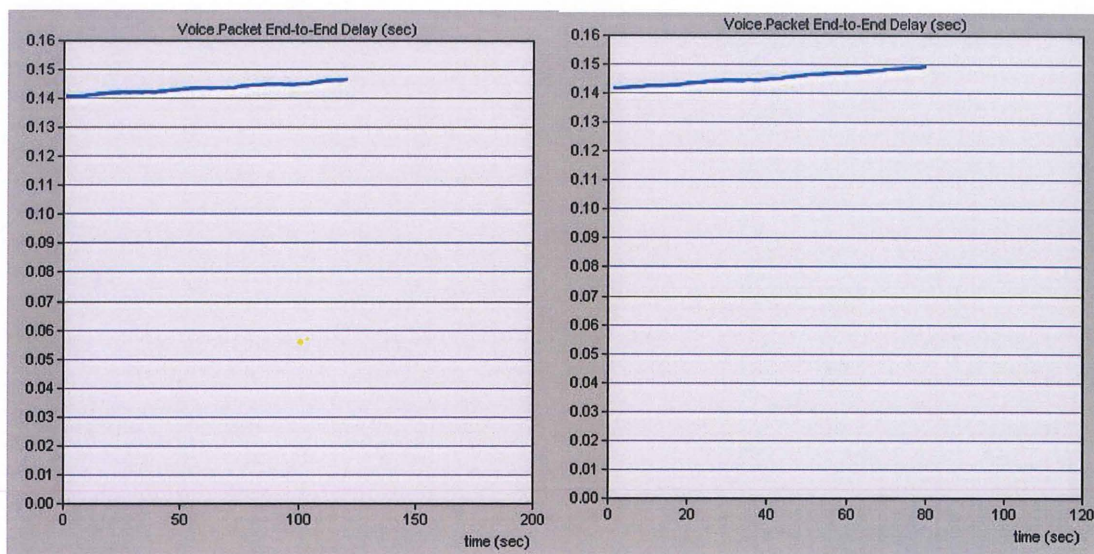
f. 1000 clients

Figure 5.6: Simulation Result of Jitter

From Figure 5.6, the graph showed that the value of jitter increases as the number of client increases. Starting from 200 clients to 600 clients, the value of voice jitter is very small and unnoticeable. From these graphs, it showed that the value of voice jitter is unstable which the graph showed the value of voice jitter went up and down but no matter how unstable the graph showed, the value of jitter is still acceptable since the value of voice jitter did not exceed 0.075 seconds. While testing for 700 clients, the graph started to change and the value of jitter increased dramatically but it still under the maximum acceptable value. This showed the same things while testing for 800 clients and 1000 clients, the value of jitter still can be accepted. In conclusion, packet jitter was still below the threshold value when tested for 1000 users. Therefore, with the current setup, the number of user can go up to 1000.

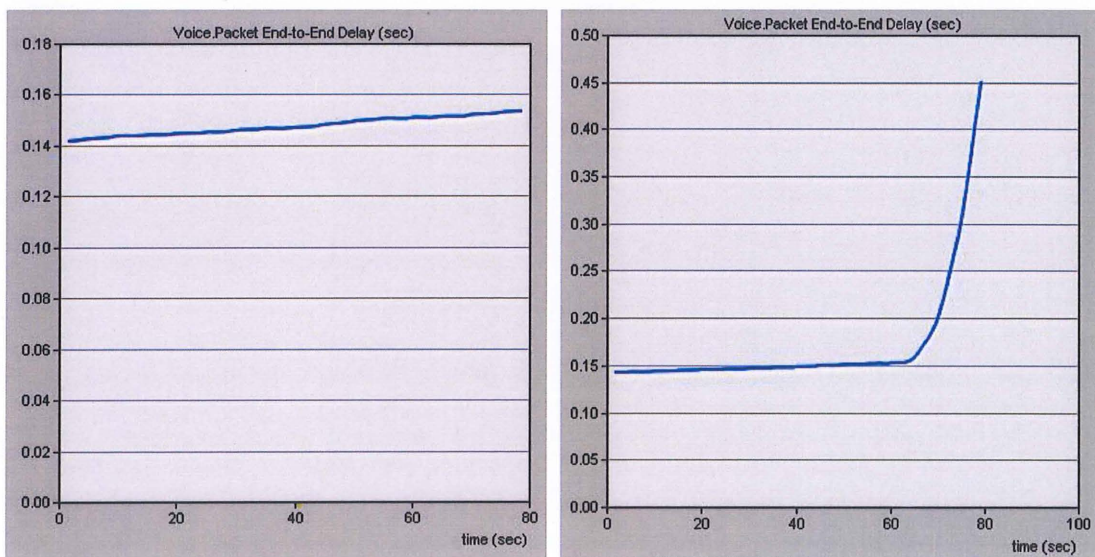
5.1.2.2 Packet End to End Delay

Threshold value of packet end to end delay = 200ms



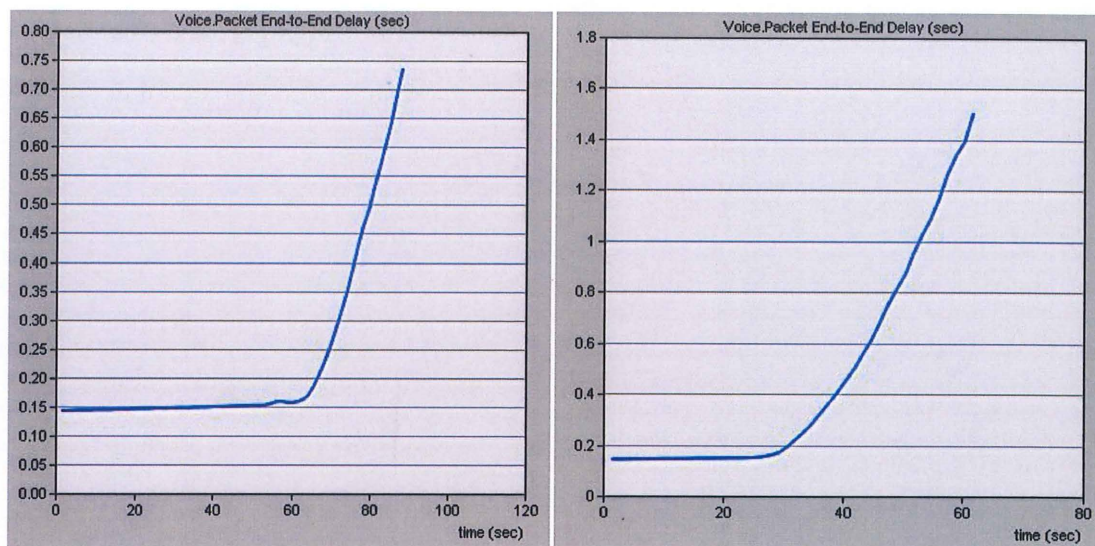
a. 200 clients

b. 400 clients



c. 600 clients

d. 700 clients



e. 800 clients

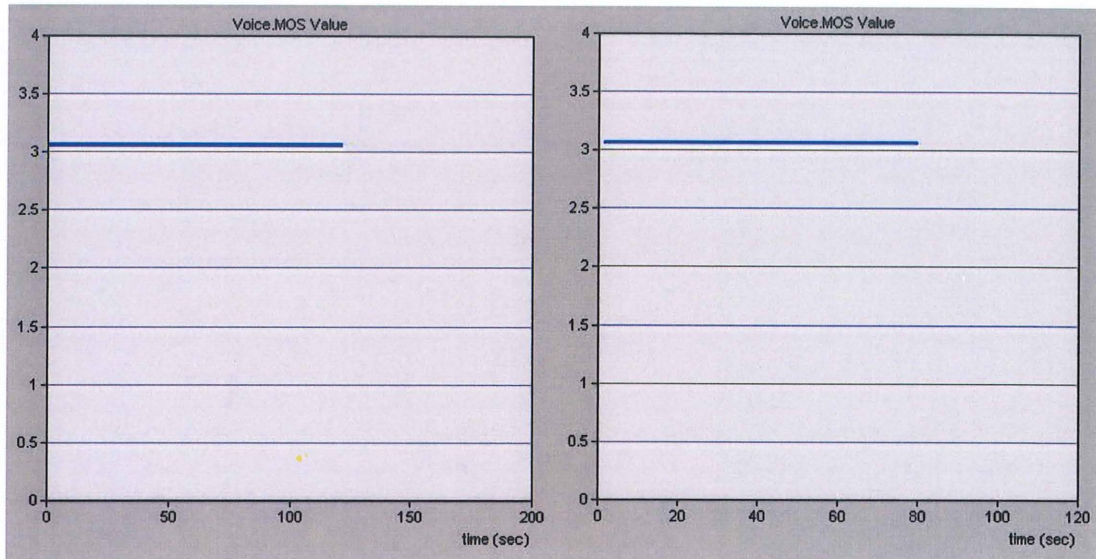
f. 1000 clients

Figure 5.7: Simulation Result of Packet End to End Delays

From the graph showed above, the value of packet ETE delay increases as the number of client increases. Starting from 200 to 600 clients, the graph showed that value of packet ETE delay increase gradually but it did not exceeds 0.2 seconds after the simulation time was completed. While testing for 700 clients, the value of packet ETE delay increased gradually which same as the previous number of clients but after 70.4 seconds, the value of packet ETE delay increased sharply until the end of the simulation time. At 70.4 seconds, the value of packet ETE delays had reached the maximum accepted value which is 0.2 seconds. At the end of simulation time, the value of packet ETE delays reached 0.450995 seconds which is twice of the acceptable value. While for testing 800 clients and 1000 clients, obviously it is not acceptable and it showed the value of packet ETE delays reached the acceptable value earlier than 700 clients. In conclusion, the packet ETE delay was above threshold value when tested for 700 users. Therefore, with the current setup, the number of user should less than 700.

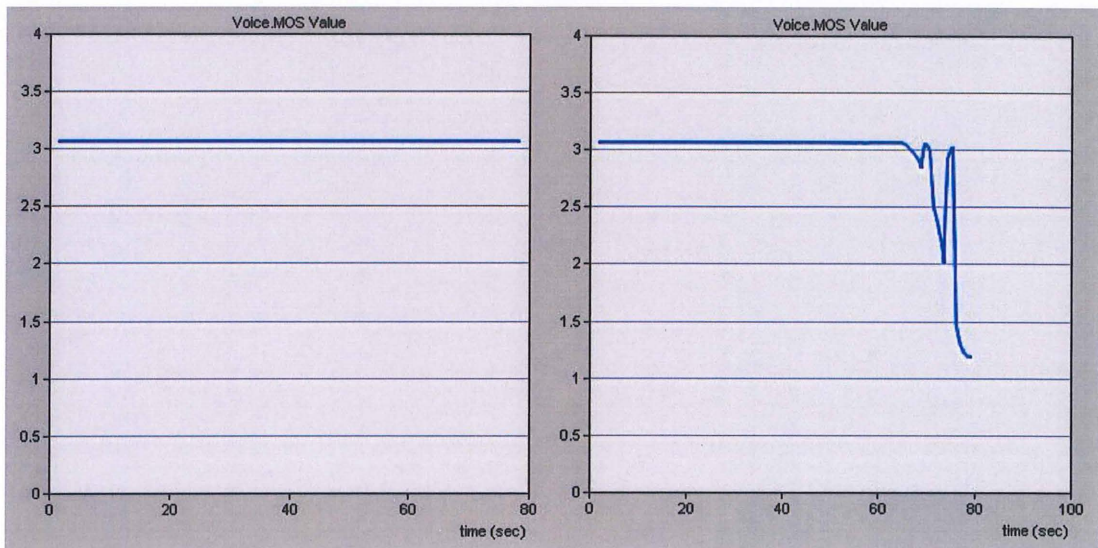
5.1.2.3 MOS

Threshold value of MOS = 3



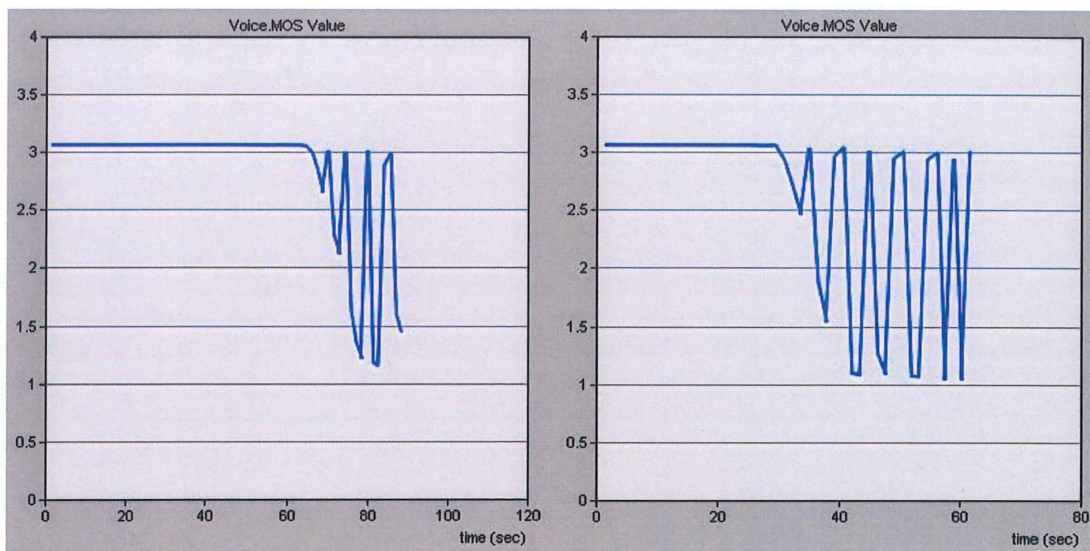
a. 200 clients

b. 400 clients



c. 600 clients

d. 700 clients



e. 800 clients

f. 1000 clients

Figure 5.8: Simulation Result of MOS

From Figure 5.8, the value of MOS decreases as the number of clients increases. As shown above, the graph from 200 clients to 600 clients showed the value of MOS in a straight line at the value of 3.0599. This showed that the quality of VoIP is stable and satisfied. While proceeding to 700 clients, the value of MOS showed a straight initially but at 67.2 seconds of simulation time, the value started to drop to 2.9707 which is not acceptable for a good quality of VoIP calls. After a few seconds, the MOS value went back to 3.0597 but it did not last for a long time and the value of MOS was dropped again to 2.0235 which lower than previous. This showed that for 700 clients, the MOS value is very stable until 67.2 seconds, after that the MOS value will started to unstable and the quality of VoIP calls are strongly affected by the traffic. While for testing of 800 clients and 1000 clients, the graph showed that the duration of time that can be supported by the network topology getting shorter. In conclusion, the MOS was under threshold value when tested for 700 users. Therefore, with the current setup, the number of user should less than 700.

CHAPTER 6

CONCLUSION

6.0 Conclusion

This research was conducted two types of scenarios which are VoIP calls in LAN without any traffic and VoIP calls in LAN with traffic. In each scenario, there were three factors such as jitter, packet end to end delay and MOS in order to estimate the maximum number of VoIP concurrent users. In each scenario, the graph of each factor was analysed and a minor conclusion was made for recommending maximum number of VoIP concurrent users.

For the first scenario, the maximum number of VoIP concurrent users from each factor is more than 1000 for jitter, 900 for packet end to end delay and 900 for MOS. Since there are differences in recommendation of maximum number of VoIP concurrent users, an accurate maximum number of VoIP users cannot be made. Therefore, a combination of three factors is required in order to obtain an accurate the maximum number of VoIP concurrent users. The table below showed the summary of scenario 1.

Table 5.1: Summary of Scenario 1

Number of Clients	Jitter	Packet ETE Delay	MOS
200	Acceptable	Acceptable	Acceptable
400	Acceptable	Acceptable	Acceptable
600	Acceptable	Acceptable	Acceptable
800	Acceptable	Acceptable	Acceptable
900	Acceptable	Not Acceptable	Not Acceptable
1000	Acceptable	Not Acceptable	Not Acceptable

From Table 5.1, the summary showed that maximum number of concurrent users for scenario 1 is less than 900. As shown in the table, two factors which are packet end to end delay and MOS showed unacceptable due to the value exceeded the maximum acceptable value.

For the second scenario, the maximum number of VoIP concurrent users from each factor is more than 1000 for jitter, 700 for packet end to end delay and 700 for MOS. The result showed the same case as scenario 1, there are several of recommendations of maximum number of VoIP concurrent users. So, there is no accuracy of maximum number of VoIP users can be made. Therefore, a combination of three factors is required in order to obtain an accurate the maximum number of VoIP concurrent users. The table below showed the summary of the scenario 2.

Table 5.2: Summary of Scenario 2

Number of Clients	Jitter	Packet ETE Delay	MOS
200	Acceptable	Acceptable	Acceptable
400	Acceptable	Acceptable	Acceptable
600	Acceptable	Acceptable	Acceptable
700	Acceptable	Not Acceptable	Not Acceptable
800	Acceptable	Not Acceptable	Not Acceptable
1000	Acceptable	Not Acceptable	Not Acceptable

From Table 5.2, the summary showed that maximum number of concurrent users for scenario 2 is less than 700. As shown in the table, two factors which are packet end to end delay and MOS showed unacceptable due to the value exceeded the maximum acceptable value.

As a conclusion, the maximum number of VoIP concurrent users for VoIP calls in LAN without any traffic is less than 900 users. While for the maximum number of VoIP concurrent users for VoIP calls in LAN with traffic is less than 700 users.

6.1 Recommendations

As stated in the first chapter of this research project, there are limitations regarding to this research. By considering the scopes and outcomes of the result, there are some proposals are recommended for the future works that related to estimation of maximum number of VoIP concurrent users. Some recommendations for future work of the research:

- The research can be completed by using various types of codec such as G.729, G.723, GSM FR and so on.
- The research also can be tested the VoIP calls in wireless local area network (WLAN).
- The extensive experiments can be done by applying different kinds of external traffic such as database, file transfer protocol (FTP), email, remote login and so on with a certain percentage of traffic.
- This research also encouraged the experiment to be done by estimate the number of VoIP concurrent users by making the VoIP calls from internal network to external network or vice versa.

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APPENDIX

Appendix A1: Configuration of Opnet Modeler

The experimental network setup was completed in OPNET modeller 14.5. A network node can be selected under the object palette as shown in Figure A1.1. The required network nodes can be dragging and dropping to the project editor in OPNET modeller.

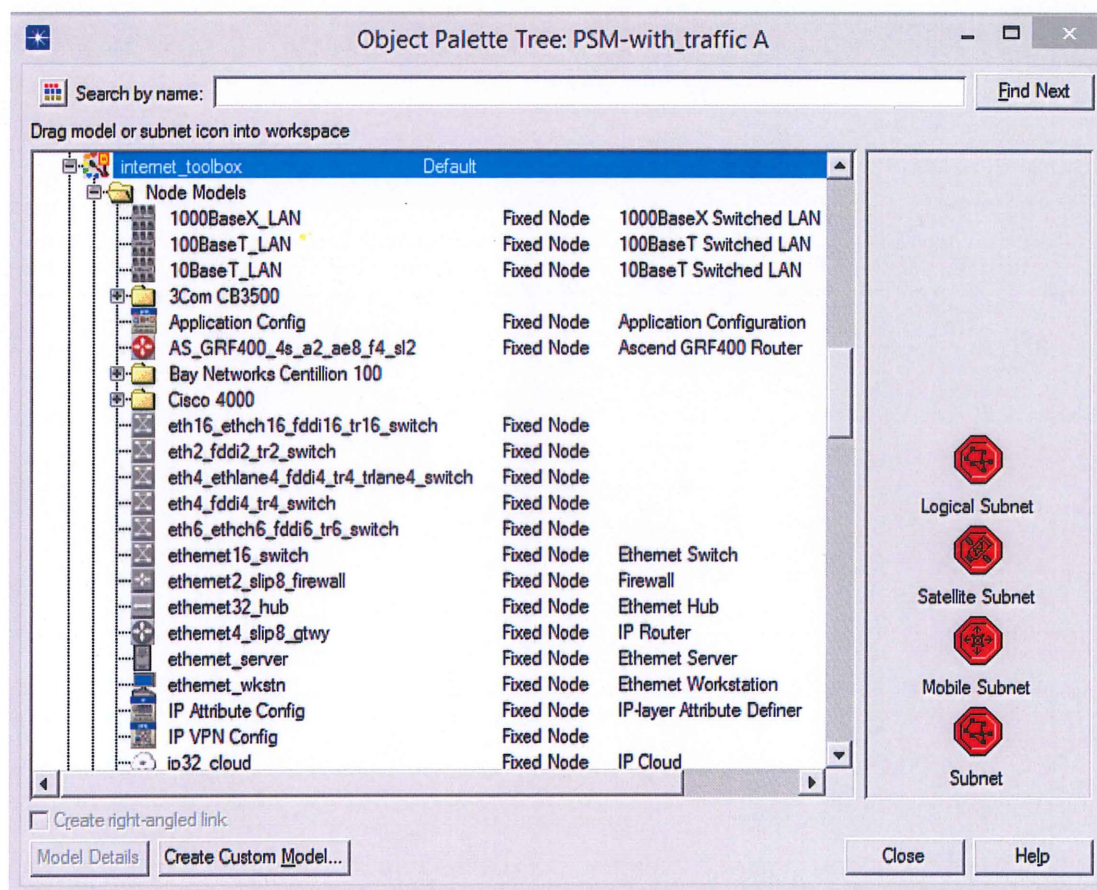


Figure A1.1: Object Palette

Configuration of Application Definition

The application definition in this version of OPNET modeller has several types of applications such as Database, Email, FTP, HTTP, Print, Remote Login, Video Conferencing and Voice. The user can edit the specification of each of the application. Figure A1.2 showed the settings for choosing the voice application in application definition.

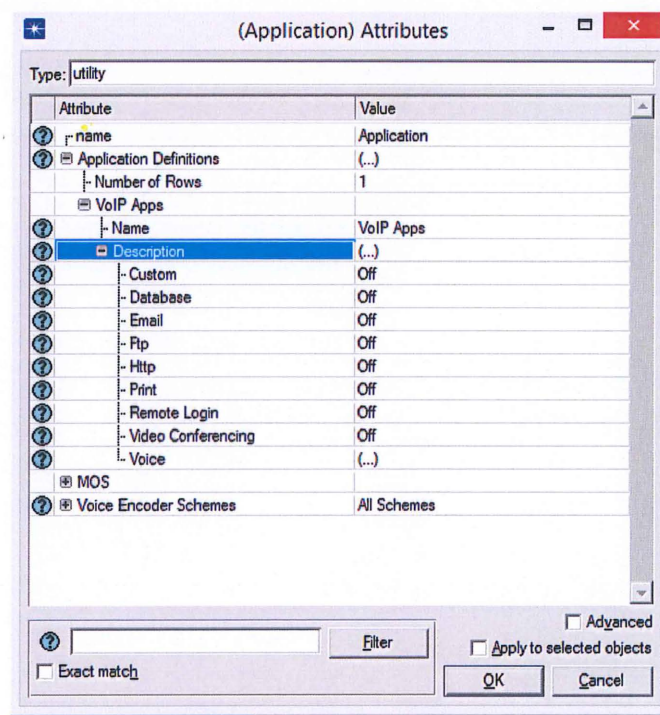


Figure A1.2: Application Definition for network without traffic

For example, in the scenario without traffic, the application selected showed in Figure A1.2. Since there is not traffic, therefore only VoIP traffic is selected.

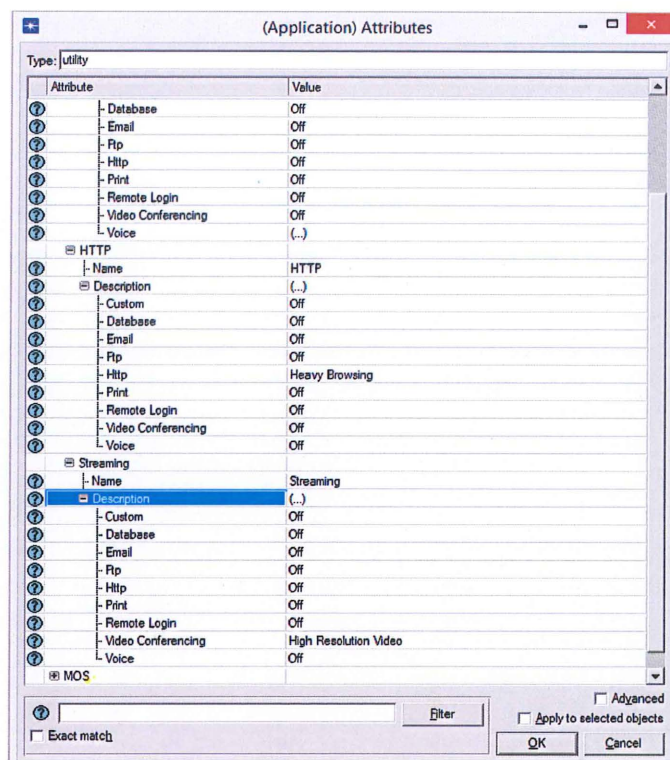


Figure A1.3: Application Definition for network external traffic

Another scenario is the network topology with external traffic which are HTTP and Video Streaming traffic are added. The configuration is showed in Figure A1.3. As showed in Figure A1.3, HTTP traffic is configured as heavy browsing while for Video Conferencing is configured as High Resolution Video.

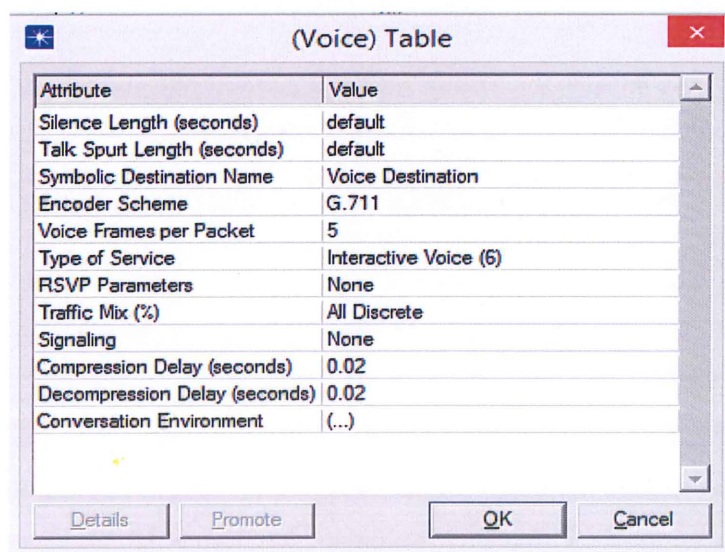


Figure A1.4: Voice Application Configuration

Figure A1.4 showed the voice table where the G.711 codec is selected from all the list of codecs. The type of service used in voice application is Interactive Voice. While for the silence length and talk spurt length is set as Default. The symbolic destination name is set as Voice Destination which will be used in configuration of LAN later.

Configuration of Profile Definition

The profile definition is used to create a user profile. These user profiles are able to be specified on different nodes in the network to generate application layer traffic. A profile can be added with different applications which are selected in application definition. Every profile has its own configurable parameters. The parameters will be shown as below:

- 1) **Operation mode:** It defines how the application will start. If it is 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.

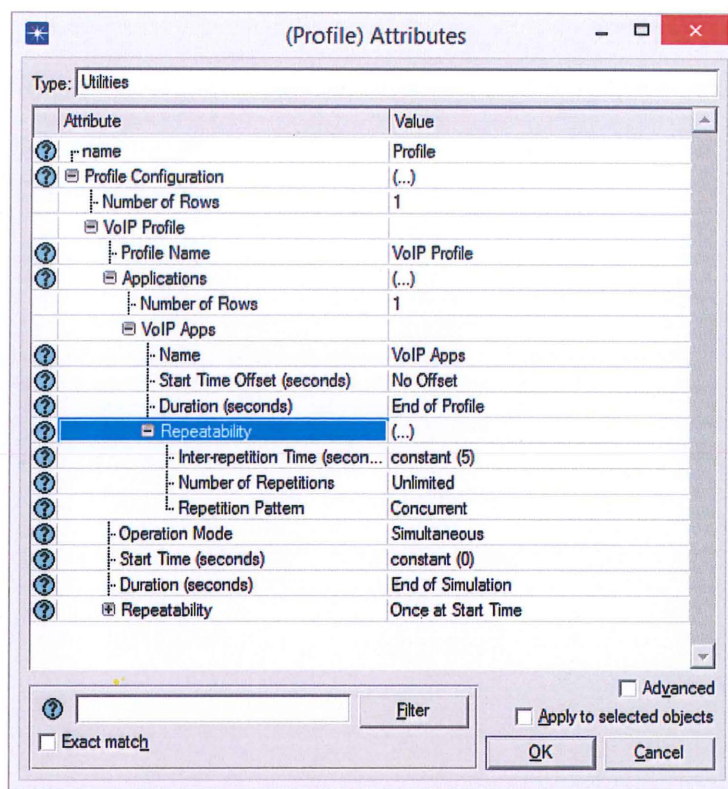


Figure A1.5: Profile Definition of network without traffic

In Figure A1.5, it showed the profile definition for scenario without traffic. Therefore, only one VoIP application is showed.

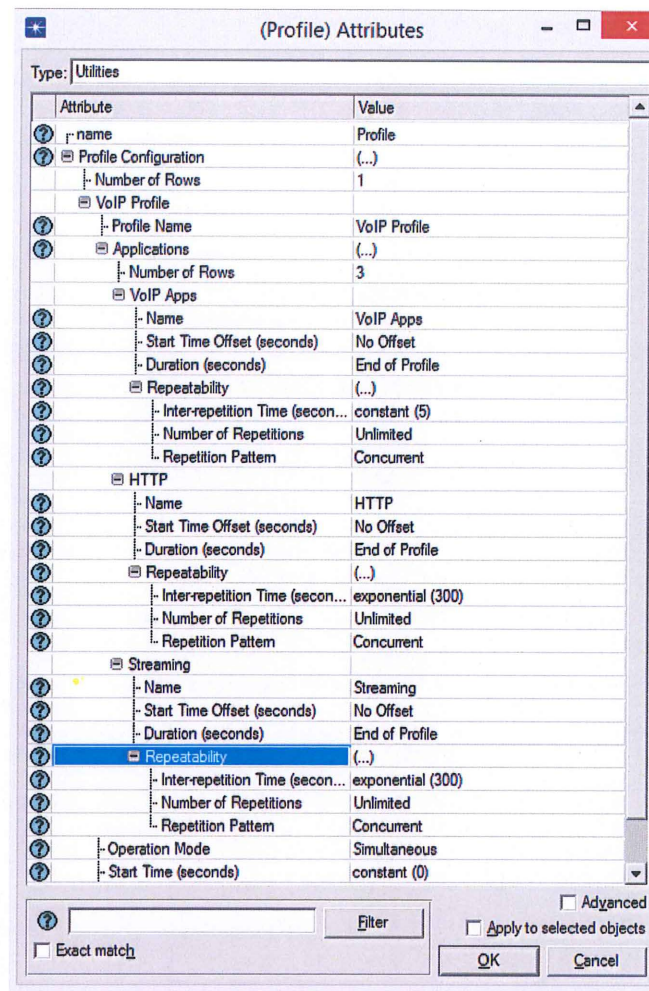


Figure A1.6: Profile Definition of network with external traffic

In Figure A1.6, it is shown the profile definition in scenario with external traffic. In this profile, there are VoIP application, HTTP application and Streaming application as shown in the figure.

Appendix A2: Simulation Output

$$*1.16E-08 = 1.16 \times 10^{-8} = 0.0000000116$$

Scenario 1: VoIP calls in LAN without traffic

a) 200 Clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.5	-3.3E-10	3.059935	1.16E-08	0.140298
3	0	3.059935	1.2E-08	0.140298
4.5	3.77E-10	3.059935	1.18E-08	0.140449
6	0	3.059935	1.2E-08	0.140486
7.5	0	3.059935	1.2E-08	0.140486
9	1.09E-05	3.059935	1.33E-08	0.140567
10.5	0	3.059935	2.46E-08	0.140675
12	0	3.059935	3.17E-08	0.140675
13.5	0	3.059935	3.45E-08	0.140675
15	9.5E-06	3.059935	4.17E-08	0.140863
16.5	0	3.059935	5.78E-08	0.140863
18	0	3.059935	6.6E-08	0.140863
19.5	1.08E-05	3.059935	6.89E-08	0.140998
21	0	3.059935	8.84E-08	0.141051
22.5	0	3.059935	1.03E-07	0.141051
24	1.18E-05	3.059935	1.08E-07	0.141122
25.5	0	3.059935	1.23E-07	0.14124
27	0	3.059935	1.45E-07	0.14124
28.5	0	3.059935	1.59E-07	0.14124
30	1.08E-05	3.059935	1.61E-07	0.141428
31.5	0	3.059935	1.91E-07	0.141428
33	0	3.059935	2.11E-07	0.141428
34.5	6.58E-06	3.059935	2.11E-07	0.141559
36	0	3.059935	2.28E-07	0.141617
37.5	0	3.059935	2.49E-07	0.141617
39	7.4E-11	3.059935	2.54E-07	0.141684
40.5	0	3.059935	2.47E-07	0.141805
42	0	3.059935	2.55E-07	0.141805
43.5	0	3.059935	2.61E-07	0.141805
45	8.4E-11	3.059935	2.4E-07	0.141993
46.5	0	3.059935	2.43E-07	0.141993
48	0	3.059935	2.44E-07	0.141993
49.5	6.6E-11	3.059935	2.3E-07	0.142123
51	0	3.059935	2.24E-07	0.142182
52.5	0	3.059935	2.24E-07	0.142182
54	5.9E-11	3.059935	2.17E-07	0.142248
55.5	0	3.059935	2.05E-07	0.14237
57	0	3.059935	2.04E-07	0.14237
58.5	0	3.059935	2.02E-07	0.14237
60	4.3E-11	3.059935	1.86E-07	0.142558
61.5	0	3.059935	1.85E-07	0.142558
63	0	3.059935	1.83E-07	0.142558
64.5	4.5E-11	3.059935	1.73E-07	0.142687
66	0	3.059935	1.68E-07	0.142747
67.5	0	3.059935	1.66E-07	0.142747
69	5.1E-11	3.059935	1.61E-07	0.142813
70.5	0	3.059935	1.53E-07	0.142935
72	0	3.059935	1.52E-07	0.142935
73.5	0	3.059935	1.5E-07	0.142935
75	4.4E-11	3.059935	1.4E-07	0.143124
76.5	0	3.059935	1.39E-07	0.143124
78	0	3.059935	1.37E-07	0.143124
79.5	4.4E-11	3.059935	1.31E-07	0.143252
81	0	3.059935	1.27E-07	0.143312
82.5	0	3.059935	1.26E-07	0.143312
84	3.7E-11	3.059935	1.22E-07	0.143377
85.5	0	3.059935	1.17E-07	0.1435
87	0	3.059935	1.16E-07	0.1435

88.5	0	3.059935	1.15E-07	0.1435
90	2.8E-11	3.059935	1.08E-07	0.143689
91.5	0	3.059935	1.07E-07	0.143689
93	0	3.059935	1.06E-07	0.143689
94.5	3E-11	3.059935	1.02E-07	0.143816
96	0	3.059935	9.94E-08	0.143877
97.5	0	3.059935	9.84E-08	0.143877
99	3.3E-11	3.059935	9.61E-08	0.143942
100.5	0	3.059935	9.26E-08	0.144065
102	0	3.059935	9.17E-08	0.144065
103.5	0	3.059935	9.08E-08	0.144065
105	3.3E-11	3.059935	8.64E-08	0.144254
106.5	0	3.059935	8.57E-08	0.144254
108	0	3.059935	8.49E-08	0.144254
109.5	3.2E-11	3.059935	8.2E-08	0.144381
111	0	3.059935	8.03E-08	0.144442
112.5	0	3.059935	7.96E-08	0.144442
114	3.7E-11	3.059935	7.79E-08	0.144507
115.5	0	3.059935	7.55E-08	0.144631
117	0	3.059935	7.48E-08	0.144631
118.5	0	3.059935	7.42E-08	0.144631
120	3.4E-11	3.059935	7.11E-08	0.144819
121.5	0	3.059935	7.06E-08	0.144819
123	0	3.059935	7E-08	0.144819
124.5	3.4E-11	3.059935	6.79E-08	0.144946
126	0	3.059935	6.67E-08	0.145007
127.5	0	3.059935	6.62E-08	0.145007
129	4.78E-07	3.059935	6.51E-08	0.145072
130.5	0	3.059935	6.4E-08	0.145196
132	0	3.059935	6.39E-08	0.145196
133.5	0	3.059935	6.36E-08	0.145196
135	2.69E-06	3.059935	6.31E-08	0.145384
136.5	0	3.059935	6.52E-08	0.145384
138	0	3.059935	6.62E-08	0.145384
139.5	4.82E-06	3.059935	6.7E-08	0.145511
141	0	3.059935	7.22E-08	0.145572
142.5	0	3.059935	7.63E-08	0.145572
144	6.86E-06	3.059935	7.89E-08	0.145637
145.5	0	3.059935	8.68E-08	0.145761
147	0	3.059935	9.56E-08	0.145761
148.5	0	3.059935	1.02E-07	0.145761
150	#N/A	#N/A	#N/A	#N/A

b) 400 Clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.5	-1.7E-10	3.059935	4.65E-08	0.140489
3	0	3.059935	4.82E-08	0.140489
4.5	4.4E-10	3.059935	4.76E-08	0.140793
6	0	3.059935	4.83E-08	0.140869
7.5	0	3.059935	4.83E-08	0.140869
9	2.2E-05	3.059935	5.36E-08	0.141031
10.5	0	3.059935	9.98E-08	0.141248
12	0	3.059935	1.28E-07	0.141248
13.5	0	3.059935	1.4E-07	0.141248
15	1.91E-05	3.059935	1.68E-07	0.141628
16.5	0	3.059935	2.33E-07	0.141628
18	0	3.059935	2.67E-07	0.141628
19.5	2.18E-05	3.059935	2.78E-07	0.141899
21	0	3.059935	3.57E-07	0.142007
22.5	0	3.059935	4.16E-07	0.142007
24	2.39E-05	3.059935	4.34E-07	0.14215
25.5	0	3.059935	4.96E-07	0.142387
27	0	3.059935	5.83E-07	0.142387
28.5	0	3.059935	6.4E-07	0.142387
30	2.18E-05	3.059935	6.5E-07	0.142766
31.5	0	3.059935	7.69E-07	0.142766
33	0	3.059935	8.5E-07	0.142766
34.5	1.33E-05	3.059935	8.5E-07	0.143031

36	0	3.059935	9.2E-07	0.143146
37.5	0	3.059935	1E-06	0.143146
39	7.6E-11	3.059935	1.03E-06	0.143283
40.5	0	3.059935	9.96E-07	0.143526
42	0	3.059935	1.03E-06	0.143526
43.5	0	3.059935	1.05E-06	0.143526
45	6.1E-11	3.059935	9.67E-07	0.143905
46.5	0	3.059935	9.77E-07	0.143905
48	0	3.059935	9.84E-07	0.143905
49.5	5.8E-11	3.059935	9.29E-07	0.144166
51	0	3.059935	9.03E-07	0.144285
52.5	0	3.059935	9.02E-07	0.144285
54	7.4E-11	3.059935	8.74E-07	0.144419
55.5	0	3.059935	8.25E-07	0.144664
57	0	3.059935	8.2E-07	0.144664
58.5	0	3.059935	8.15E-07	0.144664
60	4.6E-11	3.059935	7.51E-07	0.145044
61.5	0	3.059935	7.45E-07	0.145044
63	0	3.059935	7.39E-07	0.145044
64.5	5.3E-11	3.059935	6.99E-07	0.145303
66	0	3.059935	6.77E-07	0.145423
67.5	0	3.059935	6.71E-07	0.145423
69	5E-11	3.059935	6.5E-07	0.145556
70.5	0	3.059935	6.17E-07	0.145803
72	0	3.059935	6.11E-07	0.145803
73.5	0	3.059935	6.05E-07	0.145803
75	3.6E-11	3.059935	5.64E-07	0.146182
76.5	0	3.059935	5.58E-07	0.146182
78	0	3.059935	5.53E-07	0.146182
79.5	4.7E-11	3.059935	5.27E-07	0.146441
81	0	3.059935	5.12E-07	0.146562
82.5	0	3.059935	5.07E-07	0.146562
84	4.6E-11	3.059935	4.93E-07	0.146693
85.5	0	3.059935	4.72E-07	0.146941
87	0	3.059935	4.67E-07	0.146941
88.5	0	3.059935	4.62E-07	0.146941
90	3.8E-11	3.059935	4.36E-07	0.147321
91.5	0	3.059935	4.32E-07	0.147321
93	0	3.059935	4.28E-07	0.147321
94.5	3.4E-11	3.059935	4.11E-07	0.147578
96	0	3.059935	4.01E-07	0.147701
97.5	0	3.059935	3.97E-07	0.147701
99	3.6E-11	3.059935	3.87E-07	0.147831
100.5	0	3.059935	3.73E-07	0.14808
102	0	3.059935	3.7E-07	0.14808
103.5	0	3.059935	3.66E-07	0.14808
105	3.3E-11	3.059935	3.48E-07	0.14846
106.5	0	3.059935	3.45E-07	0.14846
108	0	3.059935	3.42E-07	0.14846
109.5	3.2E-11	3.059935	3.31E-07	0.148717
111	0	3.059935	3.24E-07	0.148839
112.5	0	3.059935	3.21E-07	0.148839
114	3.2E-11	3.059935	3.14E-07	0.148969
115.5	0	3.059935	3.04E-07	0.149219
117	0	3.059935	3.02E-07	0.149219
118.5	0	3.059935	2.99E-07	0.149219
120	2.4E-11	3.059935	2.87E-07	0.149598
121.5	0	3.059935	2.84E-07	0.149598
123	0	3.059935	2.82E-07	0.149598
124.5	3E-11	3.059935	2.74E-07	0.149855
126	0	3.059935	2.69E-07	0.149978
127.5	0	3.059935	2.67E-07	0.149978
129	9.93E-07	3.059935	2.62E-07	0.150108
130.5	0	3.059935	2.58E-07	0.150358
132	0	3.059935	2.58E-07	0.150358
133.5	0	3.059935	2.56E-07	0.150358
135	5.47E-06	3.059935	2.55E-07	0.150739
136.5	0	3.059935	2.63E-07	0.150739
138	0	3.059935	2.67E-07	0.150739
139.5	9.77E-06	3.059935	2.71E-07	0.150996
141	0	3.059935	2.91E-07	0.15112

142.5	0	3.059935	3.08E-07	0.15112
144	1.39E-05	3.059935	3.19E-07	0.15125
145.5	0	3.059935	3.51E-07	0.151501
147	0	3.059935	3.87E-07	0.151501
148.5	0	3.059935	4.13E-07	0.151501
150	#N/A	#N/A	#N/A	#N/A

c) 600 Clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.5	-7.5E-11	3.059935	1.04E-07	0.14068
3	0	3.059935	1.08E-07	0.14068
4.5	3.92E-10	3.059935	1.07E-07	0.141137
6	0	3.059935	1.08E-07	0.141251
7.5	0	3.059935	1.08E-07	0.141251
9	3.31E-05	3.059935	1.21E-07	0.141496
10.5	0	3.059935	2.25E-07	0.141822
12	0	3.059935	2.9E-07	0.141822
13.5	0	3.059935	3.15E-07	0.141822
15	2.88E-05	3.059935	3.8E-07	0.142393
16.5	0	3.059935	5.26E-07	0.142393
18	0	3.059935	6.01E-07	0.142393
19.5	3.28E-05	3.059935	6.27E-07	0.1428
21	0	3.059935	8.04E-07	0.142963
22.5	0	3.059935	9.37E-07	0.142963
24	3.59E-05	3.059935	9.78E-07	0.143177
25.5	0	3.059935	1.12E-06	0.143534
27	0	3.059935	1.32E-06	0.143534
28.5	0	3.059935	1.44E-06	0.143534
30	3.28E-05	3.059935	1.47E-06	0.144105
31.5	0	3.059935	1.73E-06	0.144105
33	0	3.059935	1.92E-06	0.144105
34.5	1.99E-05	3.059935	1.92E-06	0.144502
36	0	3.059935	2.08E-06	0.144676
37.5	0	3.059935	2.26E-06	0.144676
39	7.8E-11	3.059935	2.31E-06	0.144881
40.5	0	3.059935	2.25E-06	0.145246
42	0	3.059935	2.32E-06	0.145246
43.5	0	3.059935	2.37E-06	0.145246
45	7E-11	3.059935	2.18E-06	0.145817
46.5	0	3.059935	2.2E-06	0.145817
48	0	3.059935	2.22E-06	0.145817
49.5	6.4E-11	3.059935	2.1E-06	0.146209
51	0	3.059935	2.04E-06	0.146388
52.5	0	3.059935	2.03E-06	0.146388
54	5.5E-11	3.059935	1.97E-06	0.146589
55.5	0	3.059935	1.86E-06	0.146958
57	0	3.059935	1.85E-06	0.146958
58.5	0	3.059935	1.84E-06	0.146958
60	6E-11	3.059935	1.69E-06	0.147529
61.5	0	3.059935	1.68E-06	0.147529
63	0	3.059935	1.67E-06	0.147529
64.5	4.7E-11	3.059935	1.58E-06	0.147919
66	0	3.059935	1.53E-06	0.1481
67.5	0	3.059935	1.51E-06	0.1481
69	5.1E-11	3.059935	1.47E-06	0.148299
70.5	0	3.059935	1.39E-06	0.148671
72	0	3.059935	1.38E-06	0.148671
73.5	0	3.059935	1.36E-06	0.148671
75	4.4E-11	3.059935	1.27E-06	0.149242
76.5	0	3.059935	1.26E-06	0.149241
78	0	3.059935	1.25E-06	0.149241
79.5	4E-11	3.059935	1.19E-06	0.14963
81	0	3.059935	1.16E-06	0.149812
82.5	0	3.059935	1.14E-06	0.149812
84	2.09E-08	3.059935	1.11E-06	0.15001
85.5	0	3.059935	1.06E-06	0.150384
87	0	3.059935	1.05E-06	0.150384
88.5	0	3.059935	1.04E-06	0.150384

90	4.36E-08	3.059935	9.84E-07	0.150957
91.5	0	3.059935	9.75E-07	0.150957
93	0	3.059935	9.65E-07	0.150957
94.5	4.53E-08	3.059935	9.27E-07	0.151345
96	0	3.059935	9.05E-07	0.151529
97.5	0	3.059935	8.96E-07	0.151529
99	4.71E-08	3.059935	8.75E-07	0.151726
100.5	0	3.059935	8.43E-07	0.152102
102	0	3.059935	8.35E-07	0.152102
103.5	0	3.059935	8.28E-07	0.152102
105	4.62E-08	3.059935	7.88E-07	0.152674
106.5	0	3.059935	7.81E-07	0.152674
108	0	3.059935	7.74E-07	0.152674
109.5	4.49E-08	3.059935	7.48E-07	0.153062
111	0	3.059935	7.33E-07	0.153247
112.5	0	3.059935	7.27E-07	0.153247
114	4.36E-08	3.059935	7.13E-07	0.153443
115.5	0	3.059935	6.9E-07	0.153819
117	0	3.059935	6.85E-07	0.153819
118.5	0	3.059935	6.8E-07	0.153819
120	4.07E-08	3.059935	6.52E-07	0.154391
121.5	0	3.059935	6.48E-07	0.154391
123	0	3.059935	6.43E-07	0.154391
124.5	4.22E-08	3.059935	6.25E-07	0.154777
126	0	3.059935	6.14E-07	0.154963
127.5	0	3.059935	6.1E-07	0.154963
129	1.5E-06	3.059935	6.01E-07	0.15514
130.5	0	3.059935	5.92E-07	0.155535
132	0	3.059935	5.92E-07	0.155535
133.5	0	3.059935	5.89E-07	0.155535
135	8.23E-06	3.059935	5.85E-07	0.156089
136.5	0	3.059935	6.05E-07	0.156108
138	0	3.059935	6.15E-07	0.156108
139.5	1.47E-05	3.059935	6.23E-07	0.156475
141	0	3.059935	6.7E-07	0.15668
142.5	0	3.059935	7.09E-07	0.15668
144	2.09E-05	3.059935	7.33E-07	0.156856
145.5	0	3.059935	8.05E-07	0.157253
147	0	3.059935	8.87E-07	0.157253
148.5	0	3.059935	9.48E-07	0.157253
150	#N/A	#N/A	#N/A	#N/A

d) 800 Clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.5	-1.7E-10	3.059935	1.87E-07	0.140871
3	0	3.059935	1.94E-07	0.140871
4.5	4.03E-10	3.059935	1.91E-07	0.141481
6	0	3.059935	1.94E-07	0.141633
7.5	0	3.059935	1.94E-07	0.141633
9	4.42E-05	3.059935	2.15E-07	0.14196
10.5	0	3.059935	4.01E-07	0.142395
12	0	3.059935	5.16E-07	0.142395
13.5	0	3.059935	5.61E-07	0.142395
15	3.84E-05	3.059935	6.76E-07	0.143157
16.5	0	3.059935	9.37E-07	0.143157
18	0	3.059935	1.07E-06	0.143157
19.5	4.39E-05	3.059935	1.12E-06	0.143702
21	0	3.059935	1.43E-06	0.143919
22.5	0	3.059935	1.67E-06	0.143919
24	4.79E-05	3.059935	1.74E-06	0.144205
25.5	0	3.059935	1.99E-06	0.144681
27	0	3.059935	2.34E-06	0.144681
28.5	0	3.059935	2.57E-06	0.144681
30	4.37E-05	3.059935	2.61E-06	0.145443
31.5	0	3.059935	3.09E-06	0.145443
33	0	3.059935	3.41E-06	0.145443
34.5	2.66E-05	3.059935	3.41E-06	0.145973
36	0	3.059935	3.69E-06	0.146205

37.5	0	3.059935	4.03E-06	0.146205
39	9.2E-11	3.059935	4.12E-06	0.146479
40.5	0	3.059935	4E-06	0.146967
42	0	3.059935	4.13E-06	0.146967
43.5	0	3.059935	4.22E-06	0.146967
45	7E-11	3.059935	3.88E-06	0.147729
46.5	0	3.059935	3.92E-06	0.147729
48	0	3.059935	3.95E-06	0.147729
49.5	6.8E-11	3.059935	3.73E-06	0.148253
51	0	3.059935	3.63E-06	0.148491
52.5	0	3.059935	3.62E-06	0.148491
54	5.9E-11	3.059935	3.51E-06	0.14876
55.5	0	3.059935	3.31E-06	0.149253
57	0	3.059935	3.29E-06	0.149253
58.5	0	3.059935	3.27E-06	0.149253
60	5.6E-11	3.059935	3.01E-06	0.150015
61.5	0	3.059935	2.99E-06	0.150015
63	0	3.059935	2.97E-06	0.150015
64.5	4.79E-08	3.059935	2.81E-06	0.150537
66	0	3.059935	2.72E-06	0.150779
67.5	0	3.059935	2.69E-06	0.150779
69	6.12E-08	3.059935	2.61E-06	0.151046
70.5	0	3.059935	2.48E-06	0.151543
72	0	3.059935	2.45E-06	0.151543
73.5	0	3.059935	2.43E-06	0.151543
75	5.57E-08	3.059935	2.27E-06	0.152307
76.5	0	3.059935	2.24E-06	0.152307
78	0	3.059935	2.22E-06	0.152307
79.5	6.03E-08	3.059935	2.12E-06	0.152827
81	0	3.059935	2.06E-06	0.153072
82.5	0	3.059935	2.04E-06	0.153072
84	5.88E-08	3.059935	1.98E-06	0.153336
85.5	0	3.059935	1.9E-06	0.153836
87	0	3.059935	1.88E-06	0.153836
88.5	0	3.059935	1.86E-06	0.153836
90	5.36E-08	3.059935	1.76E-06	0.154599
91.5	0	3.059935	1.74E-06	0.154599
93	0	3.059935	1.72E-06	0.154599
94.5	5.18E-08	3.059935	1.66E-06	0.155105
96	0	3.059935	1.62E-06	0.155363
97.5	0	3.059935	1.6E-06	0.155363
99	2.42E-07	3.059935	1.57E-06	0.155601
100.5	0	3.059935	1.51E-06	0.156132
102	0	3.059935	1.5E-06	0.156132
103.5	0	3.059935	1.48E-06	0.156132
105	2.22E-07	3.059935	1.42E-06	0.156876
106.5	0	3.059935	1.4E-06	0.156901
108	0	3.059935	1.39E-06	0.156901
109.5	2.15E-07	3.059935	1.35E-06	0.157396
111	0	3.059935	1.32E-06	0.15767
112.5	0	3.059935	1.31E-06	0.15767
114	2.09E-07	3.059935	1.28E-06	0.157907
115.5	0	3.059935	1.24E-06	0.158438
117	0	3.059935	1.24E-06	0.158438
118.5	0	3.059935	1.23E-06	0.158438
120	1.95E-07	3.059935	1.18E-06	0.159182
121.5	0	3.059935	1.17E-06	0.159207
123	0	3.059935	1.16E-06	0.159207
124.5	1.9E-07	3.059935	1.13E-06	0.159701
126	0	3.059935	1.11E-06	0.159975
127.5	1E-12	3.059935	1.1E-06	0.159975
129	2.13E-06	3.059935	1.09E-06	0.160212
130.5	0	3.059935	1.07E-06	0.160744
132	0	3.059935	1.07E-06	0.160744
133.5	0	3.059935	1.07E-06	0.160744
135	1.12E-05	3.059935	1.06E-06	0.161489
136.5	0	3.059935	1.1E-06	0.161514
138	0	3.059935	1.12E-06	0.161514
139.5	1.96E-05	3.059935	1.13E-06	0.162004
141	0	3.059935	1.21E-06	0.162278
142.5	0	3.059935	1.28E-06	0.162278

144	2.79E-05	3.059935	1.33E-06	0.162513
145.5	0	3.059935	1.46E-06	0.163043
147	0	3.059935	1.6E-06	0.163043
148.5	0	3.059935	1.71E-06	0.163043
150	#N/A	#N/A	#N/A	#N/A

e) 900 Clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.5	-7.5E-11	3.059935	2.37E-07	0.140967
3	0	3.059935	2.46E-07	0.140967
4.5	4.39E-10	3.059935	2.42E-07	0.141653
6	0	3.059935	2.46E-07	0.141825
7.5	0	3.059935	2.46E-07	0.141825
9	4.97E-05	3.059935	2.73E-07	0.142192
10.5	0	3.059935	5.08E-07	0.142682
12	0	3.059935	6.54E-07	0.142682
13.5	0	3.059935	7.1E-07	0.142682
15	4.32E-05	3.059935	8.56E-07	0.14354
16.5	0	3.059935	1.19E-06	0.14354
18	0	3.059935	1.36E-06	0.14354
19.5	4.94E-05	3.059935	1.41E-06	0.144152
21	0	3.059935	1.81E-06	0.144397
22.5	0	3.059935	2.11E-06	0.144397
24	5.39E-05	3.059935	2.2E-06	0.144719
25.5	0	3.059935	2.52E-06	0.145255
27	0	3.059935	2.97E-06	0.145255
28.5	0	3.059935	3.25E-06	0.145255
30	4.92E-05	3.059935	3.31E-06	0.146112
31.5	0	3.059935	3.91E-06	0.146112
33	0	3.059935	4.32E-06	0.146112
34.5	3E-05	3.059935	4.32E-06	0.146709
36	0	3.059935	4.68E-06	0.14697
37.5	0	3.059935	5.1E-06	0.14697
39	7.3E-11	3.059935	5.21E-06	0.147279
40.5	0	3.059935	5.06E-06	0.147827
42	0	3.059935	5.23E-06	0.147827
43.5	0	3.059935	5.35E-06	0.147827
45	6.9E-11	3.059935	4.92E-06	0.148685
46.5	0	3.059935	4.97E-06	0.148685
48	0	3.059935	5E-06	0.148685
49.5	7.1E-11	3.059935	4.72E-06	0.149275
51	0	3.059935	4.59E-06	0.149542
52.5	0	3.059935	4.59E-06	0.149542
54	2.14E-08	3.059935	4.44E-06	0.149846
55.5	0	3.059935	4.19E-06	0.150401
57	0	3.059935	4.17E-06	0.150401
58.5	0	3.059935	4.14E-06	0.150401
60	6.22E-08	3.059935	3.82E-06	0.151261
61.5	0	3.059935	3.79E-06	0.151261
63	0	3.059935	3.76E-06	0.151261
64.5	6.53E-08	3.059935	3.55E-06	0.151849
66	0	3.059935	3.44E-06	0.152121
67.5	0	3.059935	3.41E-06	0.152121
69	6.53E-08	3.059935	3.31E-06	0.152421
70.5	0	3.059935	3.14E-06	0.152981
72	0	3.059935	3.11E-06	0.152981
73.5	0	3.059935	3.08E-06	0.152981
75	6.37E-08	3.059935	2.87E-06	0.153841
76.5	0	3.059935	2.84E-06	0.153841
78	0	3.059935	2.82E-06	0.153841
79.5	6.11E-08	3.059935	2.69E-06	0.154426
81	0	3.059935	2.61E-06	0.154701
82.5	0	3.059935	2.59E-06	0.154701
84	5.89E-08	3.059935	2.52E-06	0.154979
85.5	0	3.059935	2.41E-06	0.15556
87	0	3.059935	2.38E-06	0.15556
88.5	0	3.059935	2.36E-06	0.15556
90	2.36E-07	3.059935	2.23E-06	0.156397

91.5	0	3.059935	2.21E-06	0.156425
93	0	3.059935	2.19E-06	0.156425
94.5	2.22E-07	3.059935	2.11E-06	0.156982
96	0	3.059935	2.05E-06	0.157289
97.5	0	3.059935	2.04E-06	0.157289
99	2.15E-07	3.059935	1.99E-06	0.157557
100.5	0	3.059935	1.92E-06	0.158153
102	0	3.059935	1.9E-06	0.158153
103.5	0	3.059935	1.89E-06	0.158153
105	1.99E-07	3.059935	1.8E-06	0.15899
106.5	0	3.059935	1.78E-06	0.159018
108	0	3.059935	1.77E-06	0.159018
109.5	1.93E-07	3.059935	1.71E-06	0.159574
111	0	3.059935	1.68E-06	0.159882
112.5	0	3.059935	1.66E-06	0.159882
114	1.67E-07	3.059935	1.63E-06	0.160149
115.5	0	3.059935	1.58E-06	0.160746
117	0	3.059935	1.57E-06	0.160746
118.5	0	3.059935	1.56E-06	0.160746
120	1.44E-07	3.059935	1.5E-06	0.161582
121.5	0	3.059935	1.49E-06	0.16161
123	0	3.059935	1.48E-06	0.16161
124.5	1.37E-07	3.059935	1.44E-06	0.162165
126	0	3.059935	1.41E-06	0.162474
127.5	1E-12	3.059935	1.4E-06	0.162474
129	2.32E-06	3.059935	1.38E-06	0.16274
130.5	0	3.059935	1.36E-06	0.163338
132	0	3.059935	1.36E-06	0.163338
133.5	0	3.059935	1.35E-06	0.163338
135	1.25E-05	3.059935	1.34E-06	0.164175
136.5	0	3.059935	1.39E-06	0.164203
138	0	3.059935	1.41E-06	0.164203
139.5	6.58E-05	3.059935	1.44E-06	0.165178
141	6.95E-05	3.057968	1.86E-06	0.167427
142.5	6.95E-05	3.055022	2.89E-06	0.169514
144	0.000587	3.052072	4.99E-06	0.174188
145.5	0.001796	3.042398	4.29E-05	0.214362
147	0.001796	3.018392	0.000236	0.266364
148.5	0.001796	2.99434	0.000683	0.31838
150	#N/A	#N/A	#N/A	#N/A

f) 1000 Clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.5	-1.4E-10	3.059935	2.93E-07	0.141063
3	0	3.059935	3.04E-07	0.141063
4.5	4.37E-10	3.059935	2.99E-07	0.141825
6	0	3.059935	3.04E-07	0.142016
7.5	0	3.059935	3.04E-07	0.142016
9	5.53E-05	3.059935	3.37E-07	0.142424
10.5	0	3.059935	6.27E-07	0.142969
12	0	3.059935	8.07E-07	0.142969
13.5	0	3.059935	8.78E-07	0.142969
15	4.81E-05	3.059935	1.06E-06	0.143922
16.5	0	3.059935	1.47E-06	0.143922
18	0	3.059935	1.67E-06	0.143922
19.5	5.49E-05	3.059935	1.75E-06	0.144603
21	0	3.059935	2.24E-06	0.144875
22.5	0	3.059935	2.61E-06	0.144875
24	5.99E-05	3.059935	2.72E-06	0.145233
25.5	0	3.059935	3.11E-06	0.145828
27	0	3.059935	3.66E-06	0.145828
28.5	0	3.059935	4.02E-06	0.145828
30	5.47E-05	3.059935	4.08E-06	0.146782
31.5	0	3.059935	4.83E-06	0.146782
33	0	3.059935	5.33E-06	0.146782
34.5	3.33E-05	3.059935	5.34E-06	0.147445
36	0	3.059935	5.78E-06	0.147735

37.5	0	3.059935	6.3E-06	0.147735
39	8.2E-11	3.059935	6.44E-06	0.148078
40.5	0	3.059935	6.25E-06	0.148688
42	0	3.059935	6.46E-06	0.148688
43.5	0	3.059935	6.61E-06	0.148688
45	7.2E-11	3.059935	6.07E-06	0.149641
46.5	0	3.059935	6.14E-06	0.149641
48	0	3.059935	6.18E-06	0.149641
49.5	3.48E-08	3.059935	5.83E-06	0.150297
51	0	3.059935	5.67E-06	0.150596
52.5	0	3.059935	5.66E-06	0.150596
54	7.33E-08	3.059935	5.49E-06	0.150933
55.5	0	3.059935	5.18E-06	0.151552
57	0	3.059935	5.15E-06	0.151552
58.5	0	3.059935	5.12E-06	0.151552
60	6.87E-08	3.059935	4.71E-06	0.152508
61.5	0	3.059935	4.68E-06	0.152508
63	0	3.059935	4.64E-06	0.152508
64.5	7.01E-08	3.059935	4.39E-06	0.15316
66	0	3.059935	4.26E-06	0.153463
67.5	0	3.059935	4.22E-06	0.153463
69	7.12E-08	3.059935	4.09E-06	0.153797
70.5	0	3.059935	3.88E-06	0.154419
72	0	3.059935	3.84E-06	0.154419
73.5	0	3.059935	3.8E-06	0.154419
75	6.38E-08	3.059935	3.55E-06	0.155363
76.5	0	3.059935	3.52E-06	0.155374
78	0	3.059935	3.48E-06	0.155374
79.5	2.52E-07	3.059935	3.33E-06	0.155996
81	0	3.059935	3.23E-06	0.156335
82.5	0	3.059935	3.2E-06	0.156335
84	2.32E-07	3.059935	3.12E-06	0.156634
85.5	0	3.059935	2.98E-06	0.157295
87	0	3.059935	2.95E-06	0.157295
88.5	0	3.059935	2.92E-06	0.157295
90	2.09E-07	3.059935	2.76E-06	0.158224
91.5	0	3.059935	2.73E-06	0.158255
93	0	3.059935	2.71E-06	0.158255
94.5	2.02E-07	3.059935	2.61E-06	0.158874
96	0	3.059935	2.54E-06	0.159214
97.5	0	3.059935	2.52E-06	0.159214
99	1.95E-07	3.059935	2.47E-06	0.159512
100.5	0	3.059935	2.37E-06	0.160174
102	0	3.059935	2.35E-06	0.160174
103.5	0	3.059935	2.34E-06	0.160174
105	1.47E-07	3.059935	2.23E-06	0.161103
106.5	0	3.059935	2.21E-06	0.161134
108	0	3.059935	2.19E-06	0.161134
109.5	1.41E-07	3.059935	2.12E-06	0.161751
111	0	3.059935	2.07E-06	0.162093
112.5	0	3.059935	2.06E-06	0.162093
114	1.32E-07	3.059935	2.02E-06	0.162389
115.5	0	3.059935	1.96E-06	0.163052
117	0	3.059935	1.94E-06	0.163052
118.5	0	3.059935	1.93E-06	0.163052
120	1.23E-07	3.059935	1.85E-06	0.163981
121.5	0	3.059935	1.83E-06	0.164012
123	0	3.059935	1.82E-06	0.164012
124.5	1.2E-07	3.059935	1.77E-06	0.164628
126	0	3.059935	1.74E-06	0.164971
127.5	2E-12	3.059935	1.73E-06	0.164971
129	0.000512	3.059935	1.86E-06	0.167664
130.5	0.001779	3.052952	2.99E-05	0.2073
132	0.001779	3.031372	0.000202	0.258833
133.5	0.001779	3.009726	0.000625	0.310367
135	0.003391	2.988205	0.001392	0.381301
136.5	0.003697	2.965867	0.003118	0.484177
138	0.003697	2.94337	0.006191	0.58739
139.5	0.004204	2.920817	0.010695	0.692612
141	0.005614	2.898051	0.017003	0.831776
142.5	0.005614	2.874643	0.02687	0.983187

144	0.005614	2.851242	0.040509	1.1346
145.5	0.006691	2.827826	0.056885	1.293498
147	0.007531	2.803911	0.07827	1.485848
148.5	0.007532	2.77969	0.107232	1.682202
150	#N/A	#N/A	#N/A	#N/A

Scenario 2: VoIP calls in LAN with external traffic

a) 200 clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.8	#N/A	#N/A	#N/A	#N/A
1.6	-2.9E-06	3.059935	2.54E-07	0.140512
2.4	1.87E-06	3.059935	2.72E-07	0.140489
3.2	-1.9E-06	3.059935	2.47E-07	0.140463
4	1.87E-06	3.059935	2.3E-07	0.140459
4.8	7.18E-06	3.059935	1.99E-07	0.140559
5.6	0	3.059935	2.07E-07	0.140547
6.4	0	3.059935	2.08E-07	0.140528
7.2	1.36E-08	3.059935	2.11E-07	0.140531
8	0	3.059935	2.15E-07	0.140531
8.8	0	3.059935	2.2E-07	0.140571
9.6	2.94E-06	3.059935	2.4E-07	0.140756
10.4	7.53E-09	3.059935	3E-07	0.140892
11.2	0	3.059935	3.46E-07	0.140954
12	0	3.059935	3.98E-07	0.141022
12.8	0	3.059935	4.54E-07	0.141041
13.6	0	3.059935	5.02E-07	0.14107
14.4	-1.3E-05	3.059935	5.69E-07	0.141151
15.2	0	3.059935	7.16E-07	0.141313
16	0	3.059935	7.64E-07	0.141334
16.8	0	3.059935	8.16E-07	0.141351
17.6	-4.1E-07	3.059935	8.67E-07	0.141374
18.4	8.61E-06	3.059935	9.19E-07	0.141407
19.2	-1.7E-06	3.059935	9.7E-07	0.141451
20	-1.2E-05	3.059935	1.17E-06	0.141654
20.8	-2.3E-06	3.059935	1.24E-06	0.141689
21.6	-1.3E-06	3.059935	1.31E-06	0.141717
22.4	-9.3E-09	3.059935	1.38E-06	0.141714
23.2	-3E-11	3.059935	1.44E-06	0.141726
24	0	3.059935	1.5E-06	0.141712
24.8	-8.1E-06	3.059935	1.62E-06	0.14187
25.6	9.97E-09	3.059935	1.67E-06	0.141888
26.4	-8.3E-09	3.059935	1.67E-06	0.141862
27.2	0	3.059935	1.67E-06	0.141831
28	-1.6E-09	3.059935	1.65E-06	0.141799
28.8	1.57E-09	3.059935	1.63E-06	0.141767
29.6	4.29E-06	3.059935	1.55E-06	0.14185
30.4	-4.5E-07	3.059935	1.5E-06	0.141915
31.2	4.54E-07	3.059935	1.47E-06	0.141886
32	2.18E-07	3.059935	1.44E-06	0.141856
32.8	1.77E-07	3.059935	1.41E-06	0.141825
33.6	-9.3E-07	3.059935	1.38E-06	0.141789
34.4	1.44E-07	3.059935	1.31E-06	0.141816
35.2	7.21E-07	3.059935	1.2E-06	0.141991
36	6.77E-07	3.059935	1.17E-06	0.141972
36.8	1.37E-05	3.059935	1.14E-06	0.141936
37.6	-3.1E-06	3.059935	1.12E-06	0.141891
38.4	-4.2E-06	3.059935	1.1E-06	0.141822
39.2	-3.9E-06	3.059935	1.07E-06	0.141769
40	8.38E-06	3.059935	9.5E-07	0.142055
40.8	1.89E-05	3.059935	9.38E-07	0.142141
41.6	9.93E-06	3.059935	9.3E-07	0.142154
42.4	1E-05	3.059935	9.25E-07	0.142161
43.2	8.69E-06	3.059935	9.24E-07	0.142177
44	1.81E-05	3.059935	9.26E-07	0.142247
44.8	1.95E-05	3.059935	8.97E-07	0.142471

45.6	9.28E-06	3.059935	9.19E-07	0.142565
46.4	9.61E-06	3.059935	9.4E-07	0.142554
47.2	9.59E-06	3.059935	9.58E-07	0.142592
48	6.63E-06	3.059935	9.82E-07	0.142649
48.8	2.96E-06	3.059935	1.01E-06	0.142668
49.6	1.53E-05	3.059935	1.02E-06	0.14281
50.4	5.83E-06	3.059935	1.09E-06	0.142942
51.2	6.51E-06	3.059935	1.12E-06	0.142959
52	5.39E-06	3.059935	1.15E-06	0.143044
52.8	4.88E-06	3.059935	1.19E-06	0.143091
53.6	3.45E-06	3.059935	1.22E-06	0.143052
54.4	1.43E-05	3.059935	1.25E-06	0.143123
55.2	4.92E-06	3.059935	1.35E-06	0.143272
56	5.52E-06	3.059935	1.4E-06	0.143241
56.8	1.36E-06	3.059935	1.45E-06	0.143221
57.6	-2.7E-07	3.059935	1.48E-06	0.143235
58.4	-1E-05	3.059935	1.51E-06	0.143283
59.2	-5.9E-06	3.059935	1.52E-06	0.143248
60	1.4E-06	3.059935	1.47E-06	0.143368
60.8	-6.1E-06	3.059935	1.5E-06	0.143307
61.6	-7.4E-06	3.059935	1.5E-06	0.143249
62.4	-5.9E-06	3.059935	1.5E-06	0.143196
63.2	-6.4E-06	3.059935	1.49E-06	0.143185
64	-6.8E-06	3.059935	1.49E-06	0.143187
64.8	5.65E-07	3.059935	1.43E-06	0.14334
65.6	-8E-06	3.059935	1.42E-06	0.14342
66.4	-9.4E-06	3.059935	1.41E-06	0.1434
67.2	-9.6E-06	3.059935	1.38E-06	0.143394
68	-9.6E-06	3.059935	1.35E-06	0.143334
68.8	-1E-05	3.059935	1.33E-06	0.143342
69.6	-6.1E-06	3.059935	1.26E-06	0.143388
70.4	-1.1E-05	3.059935	1.21E-06	0.143452
71.2	-1.1E-05	3.059935	1.18E-06	0.143403
72	-1.1E-05	3.059935	1.16E-06	0.143345
72.8	1.1E-06	3.059935	1.14E-06	0.143415
73.6	5.73E-06	3.059935	1.12E-06	0.143558
74.4	2.95E-05	3.059935	1.09E-06	0.143766
75.2	-9.1E-07	3.059935	1.05E-06	0.144029
76	-6.2E-06	3.059935	1.05E-06	0.144026
76.8	-5.8E-06	3.059935	1.05E-06	0.144056
77.6	-5.1E-06	3.059935	1.07E-06	0.144073
78.4	-5.1E-06	3.059935	1.08E-06	0.144103
79.2	-7.3E-06	3.059935	1.1E-06	0.144158
80	#N/A	#N/A	#N/A	#N/A

b) 400 clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.4	-0.00021	3.059935	3.34E-06	0.141501
2.8	1.08E-07	3.059935	3.73E-06	0.141497
4.2	-7.6E-05	3.059935	3.96E-06	0.141674
5.6	0	3.059935	4.37E-06	0.141825
7	2.72E-09	3.059935	4.64E-06	0.141873
8.4	3.69E-09	3.059935	4.88E-06	0.141896
9.8	-3.7E-05	3.059935	5.54E-06	0.142313
11.2	4.73E-08	3.059935	5.83E-06	0.142356
12.6	2.64E-07	3.059935	5.94E-06	0.142312
14	-2.3E-05	3.059935	5.89E-06	0.142384
15.4	8.32E-06	3.059935	5.79E-06	0.142607
16.8	7.99E-06	3.059935	5.63E-06	0.142637
18.2	1.59E-05	3.059935	5.46E-06	0.142764
19.6	1.96E-05	3.059935	5.05E-06	0.14306
21	1.75E-05	3.059935	4.87E-06	0.143226
22.4	1.79E-05	3.059935	4.78E-06	0.143343
23.8	4.09E-05	3.059935	4.63E-06	0.143522
25.2	1.53E-05	3.059935	4.43E-06	0.143897
26.6	1.13E-05	3.059935	4.55E-06	0.143921
28	3.95E-06	3.059935	4.66E-06	0.143993

29.4	2.64E-05	3.059935	4.52E-06	0.144146
30.8	-8.3E-06	3.059935	4.6E-06	0.144197
32.2	-1.6E-06	3.059935	4.72E-06	0.144114
33.6	-2.5E-06	3.059935	4.8E-06	0.144078
35	1.34E-05	3.059935	4.51E-06	0.144392
36.4	-2.9E-06	3.059935	4.63E-06	0.144259
37.8	2.98E-06	3.059935	4.68E-06	0.144234
39.2	8.4E-06	3.059935	4.54E-06	0.144461
40.6	-3.5E-07	3.059935	4.41E-06	0.144732
42	-7.2E-07	3.059935	4.47E-06	0.144719
43.4	5.03E-06	3.059935	4.53E-06	0.144843
44.8	1.11E-05	3.059935	4.3E-06	0.145383
46.2	6.57E-06	3.059935	4.34E-06	0.145555
47.6	6.64E-06	3.059935	4.39E-06	0.145741
49	5.11E-06	3.059935	4.37E-06	0.145924
50.4	9.72E-07	3.059935	4.29E-06	0.146246
51.8	-2.9E-06	3.059935	4.32E-06	0.146305
53.2	-6.6E-06	3.059935	4.35E-06	0.146335
54.6	-8.6E-06	3.059935	4.26E-06	0.146552
56	-4.5E-07	3.059935	4.24E-06	0.146656
57.4	-7.8E-06	3.059935	4.29E-06	0.14674
58.8	-7.2E-06	3.059935	4.33E-06	0.146685
60.2	-8.5E-07	3.059935	4.28E-06	0.147043
61.6	-7.7E-06	3.059935	4.33E-06	0.146962
63	-1E-05	3.059935	4.37E-06	0.146969
64.4	-8.3E-06	3.059935	4.37E-06	0.147259
65.8	-6.7E-06	3.059935	4.4E-06	0.147501
67.2	-1.6E-06	3.059935	4.45E-06	0.147587
68.6	-9.4E-06	3.059935	4.46E-06	0.147654
70	-1.3E-05	3.059935	4.35E-06	0.148067
71.4	-8.3E-06	3.059935	4.38E-06	0.148181
72.8	6.73E-06	3.059935	4.4E-06	0.1483
74.2	-8.6E-06	3.059935	4.41E-06	0.148433
75.6	-7.3E-07	3.059935	4.42E-06	0.148615
77	2.71E-06	3.059935	4.37E-06	0.148709
78.4	-3.1E-06	3.059935	4.34E-06	0.148702
79.8	1.69E-05	3.059935	4.23E-06	0.149117
81.2	#N/A	#N/A	#N/A	#N/A

c) 600 clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.4	-4.2E-05	3.059935	4.69E-06	0.141689
2.8	1.3E-05	3.059934	4.04E-06	0.141834
4.2	4.1E-05	3.059935	5.03E-06	0.142303
5.6	2.52E-05	3.059935	5.36E-06	0.142667
7	9.86E-06	3.059935	5.8E-06	0.142905
8.4	1.26E-05	3.059931	6.24E-06	0.143019
9.8	4.23E-05	3.059935	6.9E-06	0.143602
11.2	1.26E-06	3.059933	7.26E-06	0.143726
12.6	-9.5E-06	3.059933	7.45E-06	0.143666
14	6.49E-06	3.059935	7.37E-06	0.143825
15.4	-7.8E-06	3.059933	7.54E-06	0.144061
16.8	-5.3E-06	3.059935	7.54E-06	0.14406
18.2	-7E-06	3.059933	7.47E-06	0.144072
19.6	2.48E-05	3.059935	7.14E-06	0.144461
21	-1.5E-07	3.059933	7.3E-06	0.144583
22.4	3.03E-06	3.059935	7.25E-06	0.144672
23.8	4.23E-05	3.059935	6.99E-06	0.144722
25.2	-9.4E-06	3.059935	6.81E-06	0.145157
26.6	9.21E-07	3.059935	6.84E-06	0.145096
28	1.2E-05	3.059935	6.77E-06	0.145177
29.4	6.14E-05	3.059933	6.36E-06	0.145693
30.8	8.7E-06	3.059933	6.45E-06	0.146038
32.2	1.03E-05	3.059935	6.65E-06	0.146112
33.6	8.41E-06	3.059935	6.72E-06	0.146117
35	3.12E-05	3.059935	6.23E-06	0.146588
36.4	1.24E-06	3.059935	6.51E-06	0.146516
37.8	1.41E-06	3.059935	6.75E-06	0.146461

39.2	1.22E-05	3.059935	6.77E-06	0.146735
40.6	-1.1E-06	3.059935	6.78E-06	0.147063
42	-2.3E-06	3.059935	6.9E-06	0.147165
43.4	-1.6E-07	3.059935	7E-06	0.147361
44.8	1.41E-05	3.059935	6.66E-06	0.148075
46.2	3.56E-06	3.059935	6.7E-06	0.148397
47.6	3.68E-06	3.059935	6.77E-06	0.148585
49	5.85E-06	3.059935	6.7E-06	0.148926
50.4	4.71E-08	3.059935	6.54E-06	0.149586
51.8	7.46E-06	3.059935	6.62E-06	0.149766
53.2	-2.8E-06	3.059935	6.72E-06	0.149888
54.6	-8.5E-06	3.059935	6.58E-06	0.150417
56	-1.3E-05	3.059935	6.62E-06	0.150526
57.4	-1.3E-05	3.059935	6.62E-06	0.150427
58.8	-1.8E-05	3.059935	6.56E-06	0.150407
60.2	-1.1E-05	3.059935	6.44E-06	0.150863
61.6	-9.4E-06	3.059935	6.48E-06	0.15082
63	-1.4E-05	3.059935	6.47E-06	0.150738
64.4	-1E-05	3.059935	6.34E-06	0.151045
65.8	-1.6E-06	3.059935	6.3E-06	0.151329
67.2	-1.2E-05	3.059935	6.33E-06	0.151308
68.6	-5.8E-06	3.059935	6.41E-06	0.15128
70	-1.9E-05	3.059935	6.45E-06	0.152031
71.4	-6.7E-06	3.059935	6.6E-06	0.152119
72.8	5.2E-06	3.059935	6.8E-06	0.152231
74.2	-1.9E-05	3.059935	7.01E-06	0.15254
75.6	1.99E-05	3.059933	7.27E-06	0.152974
77	8.37E-06	3.059935	7.36E-06	0.153164
78.4	1.12E-05	3.059935	7.44E-06	0.153326
79.8	#N/A	#N/A	#N/A	#N/A

d) 700 clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
0.8	#N/A	#N/A	#N/A	#N/A
1.6	0.000271	3.059935	6.75E-06	0.14253
2.4	1.14E-05	3.059935	6.79E-06	0.142563
3.2	7.21E-06	3.059935	7.05E-06	0.142571
4	7.54E-06	3.059935	7.39E-06	0.142627
4.8	8.99E-05	3.059935	7.82E-06	0.143031
5.6	-1.5E-05	3.059935	8.22E-06	0.143083
6.4	-2.2E-05	3.059935	8.35E-06	0.143061
7.2	-2.1E-05	3.059935	8.45E-06	0.143052
8	-5.4E-06	3.059935	8.55E-06	0.143052
8.8	-1.6E-05	3.059935	8.64E-06	0.143065
9.6	1.49E-05	3.059935	8.33E-06	0.14337
10.4	-1.3E-05	3.059935	8.43E-06	0.143566
11.2	1.37E-07	3.059935	8.4E-06	0.143598
12	-1.4E-06	3.059935	8.29E-06	0.143678
12.8	-8.4E-06	3.059935	8.26E-06	0.14371
13.6	-1.3E-05	3.059935	8.2E-06	0.143741
14.4	4.95E-05	3.059935	7.84E-06	0.143949
15.2	3.36E-06	3.059935	7.81E-06	0.144431
16	5.75E-06	3.059935	7.83E-06	0.144464
16.8	1.32E-05	3.059935	7.78E-06	0.144522
17.6	8.12E-06	3.059935	7.71E-06	0.144581
18.4	1.48E-05	3.059935	7.62E-06	0.144627
19.2	1.84E-05	3.059935	7.52E-06	0.144754
20	7.56E-05	3.059935	6.86E-06	0.145467
20.8	7.16E-06	3.059935	7.1E-06	0.145465
21.6	5.82E-06	3.059935	7.17E-06	0.1455
22.4	4.9E-06	3.059935	7.19E-06	0.145526
23.2	6.67E-06	3.059935	7.15E-06	0.14561
24	4.37E-06	3.059935	7.15E-06	0.145614
24.8	9.01E-05	3.059935	6.66E-06	0.146074
25.6	-2.6E-06	3.059935	6.9E-06	0.146223
26.4	-2.3E-06	3.059935	7.08E-06	0.146178
27.2	-1.1E-06	3.059935	7.19E-06	0.146112
28	6.66E-06	3.059935	7.26E-06	0.146111

28.8	-2.9E-06	3.059935	7.3E-06	0.146119
29.6	0.000105	3.059935	7.05E-06	0.146448
30.4	2.36E-07	3.059935	7.32E-06	0.1468
31.2	-2.6E-06	3.059935	7.58E-06	0.146806
32	5.52E-06	3.059935	7.77E-06	0.146783
32.8	-5.1E-07	3.059935	7.92E-06	0.146784
33.6	-2.7E-06	3.059935	8.03E-06	0.146829
34.4	7.21E-05	3.059935	7.88E-06	0.147024
35.2	1.61E-06	3.059935	7.63E-06	0.147648
36	1.74E-06	3.059935	7.84E-06	0.147705
36.8	-9.8E-06	3.059935	8.04E-06	0.147654
37.6	2.23E-05	3.059935	8.18E-06	0.147719
38.4	8.29E-06	3.059935	8.34E-06	0.147743
39.2	3.12E-06	3.059935	8.47E-06	0.147761
40	2.3E-05	3.059935	8.12E-06	0.148559
40.8	6.13E-06	3.059935	8.34E-06	0.148571
41.6	-2.2E-06	3.059935	8.38E-06	0.148644
42.4	3.75E-06	3.059935	8.43E-06	0.148808
43.2	4.16E-06	3.059935	8.45E-06	0.148878
44	-7.3E-07	3.059935	8.44E-06	0.148953
44.8	8.29E-06	3.059935	7.97E-06	0.149722
45.6	-1.3E-06	3.059935	7.86E-06	0.149894
46.4	9.61E-06	3.059935	7.95E-06	0.149996
47.2	-5.9E-06	3.059933	7.99E-06	0.150028
48	-8E-06	3.059935	7.99E-06	0.150096
48.8	-1.2E-05	3.059935	8.04E-06	0.150156
49.6	-1.4E-06	3.059931	7.74E-06	0.150698
50.4	-4E-06	3.059549	7.56E-06	0.151063
51.2	8.2E-07	3.059711	7.69E-06	0.151175
52	-4.7E-06	3.059112	7.83E-06	0.151195
52.8	-1.2E-05	3.059699	7.86E-06	0.151196
53.6	2.2E-07	3.05867	7.91E-06	0.151186
54.4	-2.7E-05	3.058608	7.82E-06	0.151361
55.2	-3.3E-07	3.05783	7.56E-06	0.151969
56	-7.2E-07	3.059879	7.56E-06	0.151931
56.8	2.12E-05	3.057891	7.63E-06	0.152092
57.6	-8.7E-06	3.057668	7.74E-06	0.152375
58.4	-1.7E-05	3.059866	7.8E-06	0.152012
59.2	-3.4E-06	3.059844	7.82E-06	0.151854
60	-2.1E-05	3.05986	7.36E-06	0.152689
60.8	2.25E-06	3.059842	7.46E-06	0.152828
61.6	-1.7E-05	3.059862	7.5E-06	0.152939
62.4	-4.2E-06	3.059871	7.56E-06	0.152854
63.2	5.92E-05	3.059871	7.67E-06	0.153368
64	0.00012	3.059861	8.09E-06	0.154974
64.8	0.000249	3.059874	9.03E-06	0.157667
65.6	0.000385	3.044245	1.3E-05	0.162584
66.4	0.000412	3.010869	2.32E-05	0.168589
67.2	0.000382	2.970728	4.06E-05	0.174199
68	0.000468	2.925017	6.33E-05	0.179424
68.8	0.00053	2.850495	9.61E-05	0.186312
69.6	0.000649	3.059715	0.000138	0.194781
70.4	0.000907	3.002305	0.000197	0.206996
71.2	0.000985	2.512518	0.000298	0.220764
72	0.001102	2.383137	0.000456	0.236795
72.8	0.001238	2.222103	0.000673	0.254399
73.6	0.001251	2.023558	0.000976	0.272388
74.4	0.001502	2.92627	0.001343	0.290062
75.2	0.001846	3.023607	0.001724	0.312355
76	0.001908	1.470473	0.002338	0.338111
76.8	0.002013	1.304675	0.003139	0.364915
77.6	0.002089	1.228513	0.004222	0.394891
78.4	0.002188	1.190362	0.005481	0.422659
79.2	0.002165	1.185584	0.006986	0.450955
80	#N/A	#N/A	#N/A	#N/A

e) 800 clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.4	0.000127	3.059931	6.1E-06	0.142541
2.8	9.18E-06	3.059931	6.29E-06	0.142629
4.2	8.78E-05	3.059935	7.05E-06	0.143054
5.6	-6.5E-06	3.059935	7.81E-06	0.14333
7	-1.9E-05	3.059933	8.12E-06	0.143354
8.4	-1.4E-05	3.059933	8.33E-06	0.143312
9.8	1.59E-05	3.059935	8.2E-06	0.143839
11.2	-1.1E-05	3.059935	8.31E-06	0.143892
12.6	-8.6E-06	3.059931	8.33E-06	0.143854
14	3.93E-05	3.059935	7.91E-06	0.144171
15.4	1.15E-05	3.059931	7.73E-06	0.144888
16.8	7.92E-06	3.059931	7.73E-06	0.145028
18.2	4.8E-06	3.059935	7.66E-06	0.145102
19.6	6.2E-05	3.059935	7.25E-06	0.145871
21	8.19E-06	3.059935	7.5E-06	0.146116
22.4	6.51E-06	3.059935	7.67E-06	0.146121
23.8	7.3E-05	3.059935	7.53E-06	0.146378
25.2	-4.6E-07	3.059935	7.5E-06	0.147046
26.6	-2E-06	3.059935	7.95E-06	0.146926
28	-9.4E-08	3.059935	8.18E-06	0.146834
29.4	7.12E-05	3.059931	8.08E-06	0.147408
30.8	6.43E-07	3.059931	8.66E-06	0.147803
32.2	1.04E-06	3.059931	9.14E-06	0.147747
33.6	2.76E-06	3.059931	9.41E-06	0.147702
35	3.88E-05	3.059931	8.89E-06	0.1485
36.4	5.04E-06	3.059935	9.29E-06	0.148513
37.8	1.73E-06	3.059931	9.61E-06	0.14856
39.2	1.74E-05	3.059935	9.62E-06	0.14905
40.6	8.34E-06	3.059935	9.63E-06	0.14978
42	8.07E-06	3.059935	9.85E-06	0.149931
43.4	-5.2E-06	3.059935	1E-05	0.150145
44.8	3.75E-06	3.059931	9.56E-06	0.151001
46.2	-1.5E-06	3.059781	9.68E-06	0.151193
47.6	-6.9E-06	3.059781	9.72E-06	0.151331
49	-7.1E-06	3.0598	9.56E-06	0.151696
50.4	-1E-05	3.059802	9.31E-06	0.152302
51.8	-8.9E-06	3.059761	9.37E-06	0.152225
53.2	9.93E-06	3.059639	9.22E-06	0.152334
54.6	0.000106	3.058129	9.28E-06	0.154865
56	0.000106	3.058052	1.17E-05	0.158093
57.4	-5.2E-05	3.059715	1.62E-05	0.158918
58.8	-0.00012	3.059715	1.81E-05	0.157227
60.2	2.13E-05	3.059715	1.77E-05	0.156922
61.6	0.000122	3.058311	1.82E-05	0.158247
63	0.00024	3.05828	2.17E-05	0.162704
64.4	0.000397	3.051266	3.54E-05	0.170451
65.8	0.000767	3.00534	8.38E-05	0.185869
67.2	0.00087	2.856193	0.000201	0.205452
68.6	0.000938	2.657684	0.000423	0.227554
70	0.001261	3.015081	0.000753	0.254359
71.4	0.00133	2.288318	0.001356	0.285266
72.8	0.001408	2.12575	0.002291	0.316093
74.2	0.00163	3.007284	0.003515	0.349181
75.6	0.001698	1.625575	0.004869	0.386012
77	0.001853	1.349475	0.006945	0.42776
78.4	0.001711	1.226914	0.009369	0.464103
79.8	0.001633	3.00944	0.011844	0.492822
81.2	0.001671	1.181435	0.01487	0.533453
82.6	0.001731	1.150728	0.018436	0.569501
84	0.001772	2.889965	0.022689	0.606103
85.4	0.00184	2.985879	0.026341	0.642911
86.8	0.002027	1.608229	0.031685	0.691536
88.2	0.001977	1.44343	0.037865	0.736021
89.6	#N/A	#N/A	#N/A	#N/A

f) 1000 clients

time (sec)	Jitter (sec)	MOS Value	Packet Delay Variation	Packet End-to-End Delay (sec)
0	#N/A	#N/A	#N/A	#N/A
1.4	6.72E-05	3.059931	7.68E-06	0.142714
2.8	-1.8E-05	3.059935	8.32E-06	0.142616
4.2	6.93E-06	3.059935	7.93E-06	0.142946
5.6	1.66E-05	3.059935	7.46E-06	0.143388
7	-2.4E-07	3.059935	7.39E-06	0.143666
8.4	1.88E-05	3.059935	7.44E-06	0.143885
9.8	6.46E-05	3.059935	7.5E-06	0.144988
11.2	1.89E-05	3.059935	7.9E-06	0.145291
12.6	4.04E-06	3.059935	8.19E-06	0.145388
14	6.48E-05	3.059935	8.02E-06	0.145703
15.4	-3.9E-06	3.059935	8.36E-06	0.146482
16.8	6.68E-07	3.059935	8.76E-06	0.146529
18.2	2.21E-06	3.059935	8.91E-06	0.146458
19.6	6.66E-05	3.059931	8.69E-06	0.147222
21	1.23E-05	3.059755	9.32E-06	0.14757
22.4	5.49E-06	3.059769	9.59E-06	0.147767
23.8	0.000142	3.059815	9.57E-06	0.148299
25.2	4.47E-05	3.059925	1E-05	0.149872
26.6	0.000183	3.058159	1.39E-05	0.153253
28	0.000275	3.057658	3.09E-05	0.159305
29.4	0.00074	3.059715	7.48E-05	0.171297
30.8	0.000959	2.93131	0.000218	0.194686
32.2	0.001095	2.704378	0.000562	0.222422
33.6	0.001157	2.467694	0.001161	0.252225
35	0.001566	3.036361	0.001901	0.287769
36.4	0.001572	1.88006	0.003193	0.329104
37.8	0.001567	1.545012	0.005032	0.36927
39.2	0.00153	2.967208	0.007232	0.407444
40.6	0.002053	3.040552	0.009151	0.455251
42	0.002138	1.095245	0.012381	0.504477
43.4	0.002388	1.078346	0.016846	0.560713
44.8	0.002562	3.015888	0.021392	0.613943
46.2	0.002822	1.269955	0.026344	0.680165
47.6	0.002955	1.089519	0.034619	0.754533
49	0.002799	2.951622	0.044126	0.816842
50.4	0.002973	3.002945	0.051547	0.872073
51.8	0.003167	1.072073	0.061677	0.956919
53.2	0.003279	1.06307	0.074292	1.0286
54.6	0.003261	2.942939	0.088397	1.08961
56	0.003534	2.993008	0.099776	1.169061
57.4	0.003732	1.041353	0.116135	1.266398
58.8	0.003813	2.94197	0.134228	1.340629
60.2	0.003877	1.044641	0.152671	1.39353
61.6	0.004235	3.016731	0.171314	1.508528

Appendix A3: Gantt Chart

