BORANG PENGESAHAN STATUS TESIS*

JUDUL: Optimization of Performance of VOIP across WLAN,WiMAX AND Integrated WLAN-WiMAX networks using OPNET MODELER

SESI PENGAJIAN: 2012 / 2013

Saya MUNIRAH BINTI MOHD ARIS

mengaku membenarkan tesis Projek Sarjana Muda ini disimpan di Perpustakaan Fakulti Teknologi Maklumat dan Komunikasi dengan syarat-syarat kegunaan seperti berikut:

Tesis dan projek adalah hakmilik Universiti Teknikal Malaysia Melaka.
Perpustakaan Fakulti Teknologi Maklumat dan Komunikasi dibenarkan membuat salinan untuk tujuan pengajian sahaja.
Perpustakaan Fakulti Teknologi Maklumat dan Komunikasi dibenarkan membuat salinan tesis ini sebagai bahan pertukaran antara institusi pengajian tinggi.
** Sila tandakan (/)

________ SULIT (Mengandungi maklumat yang berdarjah keselamatan atau kepentingan Malaysia seperti yang termaktub di dalam AKTA RAHSIA RASMI 1972)

________ TERHAD (Mengandungi maklumat TERHAD yang telah ditentukan oleh organisasi/badan di mana penyelidikan dijalankan)

________ TIDAK TERHAD

(TANDATANGAN PENULIS)	(TANDATANGAN PENYELIA)
Alamat tetap: No 4, Jalan Balairaya 1,	Dr Zul Azri bin Muhamad Noh
Kg Palembang,	Nama Penyelia
86000, Kluang Johor	
Tarikh: 28th August 2012	Tarikh:

CATATAN: * Tesis dimaksudkan sebagai Laporan Projek Sarjana Muda (PSM).
** Jika tesis ini SULIT atau atau TERHAD, sila lampirkan surat daripada pihak berkuasa.

OPTIMIZATION OF PERFORMACE OF VoIP ACROSS WLAN, WiMAX AND INTEGRATED WLAN-WIMAX NETWORKS USING OPNET MODELER.

MUNIRAH BINTI MOHD ARIS

This report is submitted in partial fulfilment of the requirements for the
Bachelor of Computer Science (Network Development)

FACULTY OF INFORMATION AND COMMUNICATION TECHNOLOGY
UNIVERSITI TEKNIKAL MALAYSIA MELAKA
2013

DECLARATION

Optimization of Performance of VOIP across WLAN,WiMAX AND Integrated WLAN-WiMAX networks using OPNET MODELER

is written by me and is my own effort and that no part has been plagiarized without citations.

STUDENT : ______________________ Date: 28th August 2013

(MUNIRAH BINTI MOHD ARIS)

SUPERVISOR : ______________________ Date:

(DR ZUL AZRI BIN MUHAMAD NOH)

DEDICATION

This research is lovingly dedicated to my respective parents who have been our constant source of inspiration. They have given me the drive and discipline to tackle any task with enthusiasm and determination. Without their love and support this research would not have been made possible. I dedicate my dissertation work to my family and my friends for full encouragement for me. Lastly, I want to dedicate this work to my supervisor for his encouragement and support in many aspects.

Thanks to Allah SWT for always being there for me and give me a guidance as well as good health condition to accomplish this task.

This project is the beginning of my life.

ACKNOWLEDGEMENTS

Alhamdulillah praise to Allah SWT because of with HIS permission, I manage to accomplish this project successfully. I want to take this opportunity to say warmest acknowledgment to my parents and my family for giving me continuous support in everything that I do. I also heartily thankful to my supervisor Dr. Zul Azri for his full support and guidance from the initial to the final level enabled us to develop an understanding of the subject. Not to be forgotten, all my lectures that taught me for whole three years and give me input that useful for my research. Last but not least, I want to offers my regards to all my friends who always support me and encourage me to overcome all the problem occurs in my way to accomplish this project.

ABSTRACT

The research is about analysis of VoIP applications in WLAN or WiMAX or integrated WLAN-WiMAX networks when using different voice codec and different number of workstation. The research is focuses on the data of MOS, jitter and packetthe end to end delay for VoIP application. As VoIP is one of the applications that widely used nowadays, thus the research is necessary to know the expectation of user and to analyze what is the effects of current technology in terms of performance VoIP until it meet user expectation. Other than that, there are other expect that can affect the quality of VoIP like voice codec and number of workstations. Thus this research also will determine the effect of voice codec and number of workstation in term of VoIP performance. The research is carried out using OPNET MODELER and the research take about 6 month to finish it. At the end of this research, the result will show the effect of voice codec and number of workstation in VoIP with dedicated environment.

ABSTRAK

Kajian ini analisis aplikasi VoIP dalam WLAN atau WiMAX atau pun gabungan WLAN-WiMAX apabila menggunakan codec suara yang berbezadan jumlah pengguna yang berbeza. Penyelidikan ini memberi tumpuan kepada MOS,getaran dan juga kadar kelewatan data untuk diterima. Oleh sebab VoIP adalah salah satu aplikasi yang digunakan secara meluas pada masa kini, dengan itu kajian ini adalah perlu untuk mengetahui jangkaan pengguna dan menganalisis apakah teknologi tanpa wayar semasa dari segi prestasi VoIP sehingga ia memenuhi jangkaan pengguna. Selain daripada itu, terdapat factor lain yang boleh menjejaskan kualiti suara dalam aplikasi VoIP seperti codec suara dan jumlah pengguna. Oleh itu, kajian ini juga akan menentukan kesan codec suara dan jumlah pengguna dari segi prestasi VoIP. Kajian ini dijalankan dengan menggunakan OPNET MODELER dan penyelidikan ini mengambil masa 6 bulan untuk menyelesaikannya. Pada akhir kajian ini, keputusan akan menunjukkan kesan codec suara dan jumlah pengguna dalam VoIP berdasarkan teknologi tanpa wayar yang digunakan.

TABLE OF CONTENT

LIST OF TABLES

LIST OF FIGURES

LIST OF ABBREVIATIONS

VoIP	**-**	**Voice Over Internet Protocol**
WLAN	–	Wireless Local Area Network
WiMAX	-	Worldwide Interoperability for Microwave Access
PCM	-	Pulse Code Modulation
RP	-	Research Problem
RO	-	Research Objective
RQ	-	Research Questions

CHAPTER I

INTRODUCTION

1.1 Project Background

Wireless is one of the technologies that widely used in computer networking. The term wireless show that the network is establish without using wired but the connection possibly occur using access point. It become very popular nowadays because of the characteristic of wireless communication that is user mobility and device portability in which user can communicate anytime, anywhere with anyone and the device can connected with the network without using wired connection. As the demand for wireless communication is increasing, so many technologies that is launch for the purpose of wireless communication like Wireless Local Area Network (WLAN) and Worldwide Interoperability for Microwave Access (WiMAX).

As stated in a book From GSM to LTE (Sauter, 200)the usage of WLAN increases rapidly when the hardware become more affordable and WLAN become the standard technology to interconnect computers with the internet without having wires. Wireless LAN is local area network that using wireless so that user has mobility to move around within the certain area coverage in order to stay connected to the network. It is always been chosen for home network as it is easy to install.

WLAN are using technology of breaking the data into packet and send it across the network and using shared medium in which the channel access is shared. There are a few standard of WLAN, extended with 802. 11 and each of the standards have their own characteristic. For example are 802.11 a, 802.11b, 802.11g and 802.11 n. Other than that, there are a few types of WLAN which are peer-to-peer, ad-hoc network and WiFi Direct network.

Worldwide Interoperability for Microwave Access also known as WiMax is one of other wireless and mobile communication technology which extended with 802.16 standards that design to provide connectivity in wider area coverage. As WiMAX is introduced after WLAN, thus there is some improvement in the features in term of bandwidth and range of WiMAX that can support connectivity of portable mobile broadband. WiMAX are using IP based, thus providing performance with coverage and quality of services of cellular networks. The system of WiMAX are based on digital communication system, which can provide broadband wireless access that up to 50km for fixed stations and abot 5-15 km for mobile station. The advantages of using WiMAX it is fast and cheap broadband access to market that lack of infrastructure.

As WLAN and WiMAX both wireless technology, but the fact that, WiMAX is consider as more robust compared to WLAN. The main difference in WLAN and WiMAX is shown below:

Table 1.1 The comparison of WLAN and Wimax

Technology	WLAN	WiMAX
Max Speed	11 Mbps	16Mbps
Coverage	300 feet	Several miles
Advantage	Speed, price	speed, range
Disadvantage	Short range	Interference issues

The conflict between WLAN and WiMAX is the limitation for WiMAX to develop. Thus in order to extend the reach of WiMAX technology, the satisfaction of WLAN user should be considered. The best way to make sure the performance can be improve, the both technology is combined so that it can sure the satisfaction in both user's technology. (Dong, 2009) It is practical to perform the large coverage of networks. The network of integrated can be developed with the WLAN hot spots and WiMAX backbone. Basically, special type of node which call WLAN-WiMAX router is used with dual-function which act as WLAN AP and WiMAX subscriber station which is communicate with base station. (KUNDU, May 2010).In the integrated networks, WLAN gets signals from WiMAX base station but the use can connected using WLAN or WiMAX signal, depends on the signal strength. (Karthika A L, 2013)

Resulting in the growing of wireless and mobile communication technology, so many real-time applications are also growing. For example is the Voice over IP services in which services that involving sending the voice across the network. This service helps people to sending voice data in packet which also known as phone calls through Internet by using IP. The process of transfer voices using VoIP is started with converting the analog signal into digital signal. Before the signal is converted into digital there are a few process take place to convert it. Then, after the conversion, the digital will be compressed to make sure it can be sent through the internet. The quality of the voice that sent through internet is depending on the voice codec that is used using the compression process. VoIP become popular among other voice data applications because the capability to transport of voice data in form of digitized data and with the use of network connection, it is easier to handle security and services quality. Other than that, VoIP is free from any payment like traditional telephone services as the payment is only for broadband connection.

As stated in research paper Voice over Wireless LAN and analysis of MiniSIP as an 802.11 Phone (Khurram Jahangir Khan, 2004), In order to send voice via IP networks, H.323 protocol is implemented to provide specification for real time, interactive videoconferencing, data sharing and audio applications.

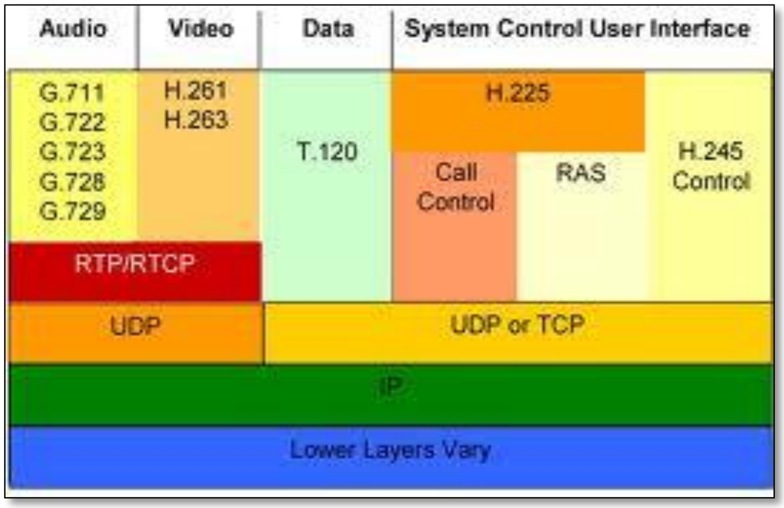

Figure 1.1 H.323 structure

As H.323 is the standard for sending voice (audio) and video using IP on a LAN, it also provides a few VoIP codec for VoIP application. Some of them are G.711, G.723, G.722, G.728 and G.729. In this research, the voice codec users are G.711, G.723 and also G.729 with the use Pulse Code Modulation (PCM). VoIP codec is been develop to implement speech Voice coders to support VoIP applications for Customer Premises Equipment (Fujitsu) .In VoIP applications, codec is used to compress regular audio. It has a characteristic of loss in which the received voice is not perfectly identical to the source, but still the sound remain the same.

There are different effect of each voice codec in the quality and performance of VoIP. Generally, G.711 is base standard for more devices. It is basically the data format of how to transfer a voice over internet in most common form. The advantages of G.711 are it will not reduce the size when transferred over internet. The size will always as the receiver sent when it arrived to the receiver. It also has higher quality voice for, end, sender and receiver. But the main problem in G.711 is G.711 use major bandwidth in order to maintain the size and quality. If the receiver or sender does not have enough bandwidth, the voice can be choppy. In addition, not all VoIP providers support G711 codec.

Other video codec is G729. G729 allow the voice phone to reduce in size before transferred to the internet. In that case, only small amount of bandwidth needed to make a call and many people can make a call at the same time. The other end that receives the voice will convert back the size to the normal so that receiver can take the call. G.729 operates at 8 times lower data rate compared to G711 and provide with the same quality.

The third codec used in this research is G723.1. G723.1 is almost the same as G729 in which used less bandwidth but it is not being a choice for user because it will not maintain the quality of sound after the reduction of the size and make the different clearly heard by the receiver.

Table 1.2 Bandwidth consumption

Number of Simultaneous Calls	**G711codec**	**G729 codec**
1	128kbs	16kbs
10	1280kbs	160kbs
100	12800kbs	1600kbs
200	25600kbs	32000kbs
500	64000kbs	8000kbs
1000	128999kbs	16000kbs

Table 1.1 above shows the bandwidth consumption for G711 and also G729. It's clearly seen that G711 use the higher bandwidth compared to G729. Even though so many technologies that available today, but still user cannot justify which technology can fulfill their expectation in their performance of services. As user a demand the best quality of VoIP, the study of voice codec also help in clarify then better voice codec so that the user are satisfy the services.

1.2 Problem Statement

It seems that the growing of technologies in wireless and mobile communication give so many advantages on user. But on other side, there are few issues that come out that need to be considered to be investigated. The first is about the demand of VoIP services that is one of the very popular services nowadays when we compared it with the wireless technologies that also rapidly improving in their performance. Secondly is about the quality of the VoIP services itself. Lastly is about so many technology are introduced thus resulting user to confuse which services should be used to meet their expected in the performance.

Table 1.3 Problem Statement Table

RO	PROBLEM STATEMENT
RP	The demand towards VoIP services keep on increasing
RP2	As VoIP is popular among users due to the advantages, but the quality of voice cannot be guaranteed.
RP3	User are confusing which wireless and mobile technologies are better for VoIP services

Table 1.4**:** Research Question

RP	**RQ**	**RESEARCH QUESTION**
RP1	**RQ1**	How can the latest mobile communication technology fulfill the demand of VoIP services?
RP2	**RQ2**	How voice codec and number of workstation involve affect the quality of voice used in VoIP?
RP3	**RQ3**	How we can differentiate the performance of WLAN, WiMAX and WLAN-WiMAX networks in carry out VoIP services?

1.3 Objectives

Based on the research problem and research, there are a few objectives that I try to achieved in this research.

Table 1.5 Research Objectives

RP	RQ	RO	RESEARCH OBJECTIVES
RP1	**RQ1**	**RO1**	To simulate the VoIP application over different wireless mobile communication technology
RP2	**RQ2**	**RO2**	To test the quality of voice of VoIP services by using OPNET Modeler
RP3	**RQ3**	**RO3**	To analyze the performance of VoIP over WLAN , WiMAX and integrated WLAN-WiMAX networks in term of their quality of services (QoS)

1.4 Scope

These researches are focuses on the performance of VoIP services using different voice codec, number of workstations and wireless technology. There are 4 metrics that will be considered in determining the performance of the VoIP services which are jitter, packet end to end delay and MOS. The VoIP service will be carried with different environment which are WLAN, WiMAX and WLAN-WiMAX networks to test the performance. As the quality of the voice is one of the issues, thus different type of voice codec also will be tested in this research. Other than that, the research also will focus on effect of workstation to the performance of VOIP. All the experiment will be carried out using wireless simulator that supports all the three technologies which are OPNET MODELER.

1.5 Project Significant

The research is basically more on analyzing the different in performance of wireless technology that can support VoIP service. Other than that, this project will determine the voice codec that can guarantee a better voice quality. Other that, the research will determine the suitable number of workstation used in a time to guarantee the performance of VOIP application. This research will compare the performance of VoIP over WLAN, WiMAX and WLAN-WiMAX .

1.6 Expected Output

At the end of this research, we expect that we manage to get all the data that can fulfill the objectives and the problem statement. The result will show the best wireless technologies for VoIP service and indirectly can help user to choose which technology that they want to use. In addition, we expect that we can determine the best voice codec that can be used for better quality of video used in VoIP as well as the suitable number of workstation used.

1.7 Conclusion

This chapter is basically introducing the wireless technologies, the advantages and disadvantages, their specification and also the way the technologies can support the VoIP service. Other than that, at the end of this chapter, we also know what VoIP service is. Next, at this chapter we can find the problem faced by user that related with the technologies