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JUDUL: PERFORMANCE OF VOICE OVER INTERNET PROTOCOL (VOIP) IN WIRELESS LOCAL AREA NETWORK (WLAN)

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PERFORMANCE OF VOICE OVER INTERNET PROTOCOL (VOIP) IN WIRELESS LOCAL AREA NETWORK (WLAN)

ONG JAW FEEI

This report is submitted in partial fulfillment of the requirements for the Bachelor of Computer Science (Computer Networking)

FACULTY OF INFORMATION AND COMMUNICATION TECHNOLOGY
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DECLARATION

I hereby declare that this project report entitled
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is written by me and is my own effort and that no part has been plagiarized without citations.

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ABSTRACT

In this project, the suitability of 802.11b networks to carry real time voice traffic communication will be assessed using the Internet Protocol, where in the network the voice session have to share the link with data, video and voice. Measurements of the quality of voice are performed in the 802.11 wireless networks compared to the wired kind of network. Measurements are focused on the three major Voice over Internet Protocol (VoIP) parameters: loss, delay and jitter. The environment factors such as the distance between nodes, network traffic and other obstacles are also being considered when carrying out this research. Due to the unpredictable nature of the wireless environment, providing Quality of Service (QoS) service is a challenging task. Voice traffic is more susceptible to the three major performance QoS parameters: delay, jitter and packet loss in the wireless domain than data traffic. In order to ensure the QoS of VoIP in wireless LAN, some experimental investigations have been carried out to test on the performance limitations in establishing VoIP in wireless LAN. All these experimental testing have been done under three varying scenarios network which are working under an ideal and real condition. The analysis is based upon the software that are being used in this project, ClearSight analyzer and Ethereal, from packet loss, jitter and delay measurements. According to the results, measures and graphs of packet delay, packet jitter and lost packet rate show the performance limitations when VoIP runs in wireless LAN. VoIP is a multimedia application that requires timely servicing of the voice traffic even when using QoS enforcement. As a result, a dedicated bandwidth with several time constraints is the main conditions for VoIP in wireless LAN environment to avoid the degradation of performance of VoIP service in the received signal.

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ABSTRAK

Dalam projek ini, kesesuaian rangkaian 802.11b akan dikaji untuk menghantar komunikasi trafik suara masa nyata dengan menggunakan *Internet Protocol*, dimana rangkaian pembawa *voice session*, mesti dikongsi untuk membawa data, video dan suara. Suatu teknik sukatan kualiti suara digunakan dalam rangkaian tanpa wayar 802.11 berbanding dengan rangkaian wayar. Sukatan berkenaan tertumpu kepada tiga parameter VoIP yang utama: *loss*, *delay* dan *jitter*. Faktor persekitaran seperti jarak antara dua titik, trafik rangkaian dan halangan-halangan lain juga merupakan faktor yang perlu diambil kira semasa menjalankan kajian ini. Oleh sebab sifat yang tidak stabil persekitaran rangkaian tanpa wayar, untuk memenuhi permintaan kualiti perkhidmatan (QoS) VoIP-WLAN merupakan suatu tugas yang agak mencabar. Prestasi trafik suara boleh dipengaruhi oleh tiga parameter QoS yang utama seperti *delay*, *jitter* dan *packet loss* terutamanya dalam rangkaian tanpa wayar berbanding dengan rangkaian yang menghantar data sahaja. Untuk mengenal pasti kualiti perkhidmatan VoIP dalam LAN tanpa wayar, eksperimen telah dijalankan untuk menguji had-had kepada prestasi penggunaan VoIP. Uji kaji telah dilakukan dengan tiga senario yang berbeza dalam keadaan ideal dan nyata. Analisa dilakukan berdasar kepada perisian yang digunakan dalam projek, iaitu ClearSight analyzer dan Ethereal untuk mengukur *delay*, *packet loss* dan *jitter*. Merujuk keputusan yang didapati, ukuran dan graf-graf *packet delay*, *packet jitter* dan *lost packet* dilakar menunjukkan prestasi VoIP yang terhad pada LAN tanpa wayar. VoIP ialah sejenis aplikasi multimedia yang memerlukan ketepatan masa dalam penghantaran trafik suara walaupun menggunakan penguatkuasaan QoS. Sebagai keputusan, lebar jalur dedikasi dengan kekangan masa merupakan keadaan utama bagi VoIP dalam persekitaran LAN tanpa wayar bagi mengelakkan prestasi perkidmatan VoIP menurun pada isyarat yang diterima.

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LIST OF ABBREVIATIONS

ACK	-	Acknowledgement
ADSL	-	Asymmetric Digital Subscriber Line
AP	-	Access Point
CAT5	-	Category 5
CCK	-	Complementary Code Keying
DHCP	-	Dynamic Host Control Protocol
DID	-	Direct Inward Dialing
DoS	-	Denial of Service
EAP	-	Extensible Authentication Protocol
HCF	-	Hybrid Coordination Function
IAPP	-	Inter Access Point Protocol
IDS	-	Intrusion Detection System
IEEE	-	Institute of Electrical and Electronics Engineers
IP	-	Internet Protocol
IPS	-	Intrusion Prevention System
ITU	-	International Telecommunication Union
LAN	-	Local Area Network
MAC	-	Media Access Control
MIC	-	Message Integrity Checks
MOS	-	Mean Opinion Score
MOSc	-	Mean conversation-Opinion Score
MOS_L	-	Mean listening-Opinion Score
MTBF	-	Mean time between failures
MTTF	-	Mean time to failure
MTTR	-	Mean time to repair
NAT	-	Network Address Translator
OFDM	-	Orthogonal Frequency Division Multiplexing
PBX	-	Private Branch eXchange
PC	-	Personal Computer
PCF	-	Point Coordination Function
PPS	-	Packets per Second
PSK	-	Phase Shift Keying
PSTN	-	Public Switched Telephone Network
QAM	-	Quadrature Amplitude Modulation
QoS	-	Quality of Service
RF	-	Radio Frequency
STUN	-	Simple Traversal of UDP through NAT
TCP	-	Transmission Control Protocol
TKIP	-	Temporal Key Encryption
UDP	-	User Datagram Protocol

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UPQ	-	User-Perceived Quality
UTP	-	Unshielded Twisted Pair
VoIP	-	Voice over Internet Protocol
VoWLAN	-	VoIP over wireless LAN
WEP	-	Wired Equivalent Privacy
WLAN	-	Wireless Local Area Network
WMM	-	Wireless MultiMedia
WPA	-	Wi-Fi Protected Access
WTT	-	Wait to Transmit

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CHAPTER I

INTRODUCTION

1.1 Project Background

The world is becoming increasingly Internet Protocol (IP) oriented, with a large number of devices getting networked every day. At the same time, individuals are starting to favor smaller and lighter devices such as laptops since they are portable and not too bulky compared to the normal desktop computers. As their modalities and patterns of use get shaped, there is a trend of doing some researches of the performance of the Voice over IP (VoIP) service whenever the distance of the wireless laptops are getting bigger to a certain distance with coverage of wireless that can provide the good quality of VoIP service. Thus communicating through VoIP can be more reachable between these devices in the same Local Area Network (LAN) environment.

VoIP is one of the fastest growing Internet applications today. It has two fundamental benefits compared with voice over traditional telephone networks.
First, by exploiting advanced voice compression techniques and bandwidth sharing in packet-switched networks, VoIP can dramatically improve bandwidth efficiency. Second, it facilitates the creation of new services that combine voice communication with other media and data applications like video, white boarding and file sharing.

At the same time, driven by huge demands for portable access, the Wireless LAN (WLAN) market is taking off quickly. Due to its convenience, mobility and high speed access, WLAN represents an important future trend for the Internet access.

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As a result of the convergence of these two trends, we believe VoIP over WLAN is poised to become an important Internet application. Before that can happen, however, two technical problems need to be solved. The first is that the system capacity for voice can be quite low in WLAN. The second is that VoIP traffic and data traffic from traditional applications such as Web and e-mail can interfere with each other and bring down VoIP performance.

As we know, when deploying VoIP in this environment, one of the challenges is meeting the performance requirements to provide the best condition which meet with the Quality of Service (QoS) in VoIP. When roaming around in a wireless network environment, at some point the mobile device will have to switch access point that is being associated with due to the media in wireless network is not constant. These media may be very different and each of it will face an additional problem such as the delay, jitter and losses as a network problem that can affect the voice quality of the VoIP service that is being used at that time.

Voice quality is an end-to-end matter with various network equipment, protocols and policies. The performance of voice service is sensitive to delay, and the voice quality can tolerate some dropped packets within a certain limit of packet loss rate. It is supposed that delays are made small enough to make two ways conversation possible. Beside that, it is important to know how long it takes for a packet to travel through a network.

The quality of a voice call is not only depends on network conditions, but also on the perceptual characteristic of the end users and how these network impairments affect the user's perception. Other factors are the user's expectation, since the mood and human memory can increase or decrease the perceived quality.

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1.2 Problem Statements

The coverage of the wireless LAN is limited. As a result, we need to do a research on the quality of the VoIP service as the distance of the device (laptops) are getting bigger. To maintain the quality of service (QoS) in VoIP, we need to find out which length of distance is the most suitable distance to get the best service of VoIP. Besides, the performance of the VoIP service must be monitored to ensure that every packets that being sent by the sender can be received by the receiver without any loss of information. There may be a problem if the distance between two devices getting bigger, the quality of the VoIP can be degraded to a certain extend based on the wireless LAN environment. This research aimed to come out with a graph to observe every packets of information that is sent by VoIP service whenever there are any changes of distance between these two laptops in a Wireless LAN (WLAN) environment.

1.3 Objectives

We conduct measurements of critical parameters that affect the quality of voice over wireless networks, namely loss, delay and jitter. Delay has the important characteristics that once introduced it cannot be hidden or minimized, hence it is important to know where exactly it is introduced into the system. Loss on the other hand can be compensated for typically by adding redundancy to the transmitted signal allowing some of the received signal to be reconstructed at the receiver.

This research explores the causes of the degradation of those parameters, and finds an optimal condition of the environment for software, hardware to be deployed in order to improve voice performance of VoIP over Wireless LAN (WLAN). Therefore, this research will be done due to the purpose:

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i. To maintain the coverage of wireless network in order to provide a good quality of VoIP service in a LAN. The coverage of the wireless network is limited and not too constant, so we need to set up a good wireless LAN environment to ensure the quality of service in VoIP is up to standard.

ii. To analyze the quality of VoIP when the distance between the devices are getting bigger as the delay, jitter and packet loss will cause the performance of VoIP degrades. When we located two devices (laptops) in a certain distance, a testing will be made to check if the quality of the VoIP is satisfied. The length of distance can be varied in order to get the best quality of service for VoIP service.

iii. To map the coverage of the wireless network for a VoIP service to communicate between devices. We can get a brief coverage of the Wireless LAN (WLAN) to enable the two laptops to communicate in a wireless environment although there may be a loss of data when transmitted through VoIP service whenever the distance has being changed.

1.4 Scopes

The scopes of this research topic are included the following:

i. The performance of VoIP
   - This performance of VoIP will be measured in the term of loss rate, delay of data and jitter using the specialized quality of service (QoS) Measurement Tools.

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ii. The distance of the laptops

- The length of distance between the two laptops will be changed from time to time to test on the quality of the VoIP service in a Wireless LAN (WLAN) environment.

iii. The coverage of wireless

- The coverage distance of the wireless network may be different due to some factors such as walls, windows and interference sources and with typical open-air operating distances of 100 to 500 meter or 300 to 1500 feet (Trulove, 2002).

iv. The brand of the laptops

- In this research, the laptops that is being used has been specified to ensure that the communication in VoIP service between these two devices is constant and not affected by any hardware or software issue due to the differences for the brand of laptops.

1.5 Project Significance

This research will be used as a guideline for those users who wish to use VoIP service in a Wireless LAN (WLAN) environment to enable the quality of the VoIP service is always being fulfilled within the coverage of distance. In addition, for those Network Expert, this research can be studied more in depth to solve the problem that deal with the quality of service (QoS) in VoIP service as the length of distance between these two wireless devices are getting bigger and the ways to improve it without any loss of data or delay during the transmission of voice packets.

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1.6 Expected Output

In this research, we need to find out that the reason of degradation of those parameters during the conversation in VoIP service such as the traffic and distance of each node to access point. In the normal condition, we can expect that the closer the devices to the access point the stronger the signal will be received. When the distance of the devices are getting bigger, the quality of the VoIP service will be degraded in the term of loss packet, delay of voice packets and jitter.

1.7 Conclusion

As the Internet growing fast recently, VoIP just like PCs, web access, e-mail will grow out of the enterprise market and get into the residential market. It is because this technology can be considered as a cheaper, simpler, and more convenient to use and it is expected to replace the legacy network infrastructure. However the appeal of the efficiencies of VoIP and Wireless Network (802.11) cannot be ignored and the end result may be very well being a mix of technology.

The following chapter is the Literature Review and Methodology in order in completing this research. This chapter can be viewed as an important element in the development process as it can used as a guideline to the researcher. The method that is being used to do this research will include the facts that we have found in any journals or articles in researching in this topic and the requirements in doing this research will be stated out as well.

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CHAPTER II

LITERATURE REVIEW AND PROJECT METHODOLOGY

2.1 Introduction

In this chapter, we will discuss and study about the quality of service (QoS) of the Voice over IP (VoIP) service, the distance of coverage in Wireless Local Area Network (WLAN) and the method we used in executing this research. To complete a full thesis topic, we need to do more research and study more on that topics or terms in order to understand the research topic more clearly. Those researches that we have gathered such as journals or articles can be used as a guideline to ensure that the research can be executed successfully. We need to study more in depth in the area of scope for this research topic, its main objective, find out the system requirement of this research topic and a comparison can be made if there is any similar research topic which has been done before this.

To fulfill the requirements in executing this research, there are a few tasks which need to be done in order to complete this chapter. Firstly, we need to search for particular information through Internet, reference books and journals that published through article in Internet. Besides this, this research also can be done by comparing with other existing research so that we can do some enhancements on it to make it better and valuable for future market research. All these information can be used as a source and guideline in completing this research.

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The techniques which are being used in executing this research play an important role as it can be used as a research element whether the techniques are something that have never been learned before or those that are currently used in the market. For the methodology part, every step that is being used in executing this project will be stated out clearly from the beginning of planning until producing the end product and getting the desired result. The method that used may be varied from other research thesis but with the same purpose or vice versa. The project requirements are those equipments or tools that needed in executing this research such as the hardware, software and network requirements. These requirements enable a project can be planned properly before carrying out and developing it.

There is a session named project schedule and milestone which allow students to plan their project according to the time frame and organize their time table wisely so that they managed to complete the project in the duration of time given. As a result, each student must plan their project according to the schedule and follow it in order to complete the project.

As a conclusion, this chapter allows students to search for the particular information about the existing research and try to figure out its weaknesses or enhance it to a more valuable research in the market. Therefore, this chapter can be used as a reference to study the current research and then come out with our own new idea that is unique and totally different from those that are currently can be found easily in our life or environment.

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