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Alamat tetap : 55 Felda Ketengah 02
23300 Dungun, Terengganu.

Tarikh : 17/11/06

(TANDATANGAN PENYELIA)

MOHD FAIRUZ ISKANDAR B. OTHMAN
Pensyarah
Fakulti Teknologi Maklumat dan Komunikasi
Kolej Universiti Teknikal Kebangsaan Malaysia
Karung Berkunci 1200
Ayer Keroh, 75450 Melaka

Nama Penyelia

Tarikh : 17/11/06

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PERFORMANCE ANALYSIS OF VOIP APPLICATION

IRZA HAFIZA BIN YAAKUB

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

FACULTY OF INFORMATION AND COMMUNICATION TECHNOLOGY
KOLEJ UNIVERSITI TEKNIKAL KEBANGSAAN MALAYSIA
2006

DECLARATION

I hereby declare that this project report entitled

PERFORMANCE ANALYSIS OF VOIP APPLICATION

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

STUDENT : ______________________ Date : 17/11/06.

(IRZA HAFIZA BIN YAAKUB)

SUPERVISOR : ______________________ Date : 17/11/06

(MOHD FAIRUZ ISKANDAR BIN OTHMAN)

DEDICATION

To my beloved Mom, Zainun Binti Abas,
to my beloved sisters, brothers and all my friends who have
encouraged, guided and inspired me throughout my journey of studies.
Also thanks to God.

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ABSTRACT

Although there are many suitability of Voice Over Internet Protocol (VoIP) and the advantages of one over the other. VoIP was originally developed to provide voice communication between computer users in different locations. But, today People using VoIP can call any telephone anywhere in the world and can receive calls on telephone sets connected to the Internet or Local Area Network (LAN). These situations bring an idea to make a Performance Analysis of VoIP Application over internet. This thesis intends to analyze the performance of VoIP Yahoo Messenger and Skype Messenger. It explains about the voice quality performance, Jitter and packets loss. The measuring score for voice quality that use is Mean Opinion Score (MOS). A few different types of internet connection and different tools will be used to compare the best VoIP application to used. The analysis result could illustrate the benefits of the technology to be used and implement on business environment or education environment in the future

ABSTRAK

Walaupun terdapat pelbagai jenis kesesuaian dalam *Voice Over Internet Protocol (VoIP)* dan juga kelebihan berbanding dengan yang lain, VoIP asalnya di bangunkan bagi menyediakan komunikasi suara di antara 2 komputer pengguna di tempat yang berlainan. Tetapi, pada hari ini pengguna menggunakan VoIP boleh membuat panggilan ke seluruh tempat dan boleh menerima panggilan di talian telefon yang telah di lengkapi reka bentuk di dalam talian internet mahupun di kawasan setempat. Berdasarkan situasi ini lah, cadangan bagi membuat analisis terhadap prestasi dalam VoIP aplikasi iaitu Yahoo Messenger dan juga Skype Messenger. Ia menerangkan berkenaan prestasi atau daya kualiti suara, Jitter dan kehilangan paket. Jenis pengukuran yang di gunakan untuk kualiti suara adalah *Mean Opinion Score (MOS)*. Beberapa jenis hubungan rangakaian yang berlainan dan juga alat pengukuran di gunakan bagi membezakan setiap VoIP aplikasi yang di gunakan untuk analisis. Daripada keputusan analisis, ia memberikan kebaikan dalam teknologi VoIP untuk di gunakan dan di perluaskan ke dalam persekitaran perniagaan atau pun persekitaran pengajian di masa hadapan.

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INTRODUCTION

1.1 Project Background

No standardized scheme exists for monitoring Voice over Internet Protocol (VoIP) quality. Many efforts are underway to measure and monitor the quality, however many have been solo ones. The goal of this project has been to determine what parameters are important and how it should be measured. As a side product should have measured the quality of VoIP in Instant Messaging (IM) application and compared it with different type of application between Yahoo Messager v 7.0 and Skype v 2.5. This project also will be compared on the different type of connection will be used when running this VoIP application.

This project has focused on the method factors of quality, namely packets loss jitter and parameters Mean Opinion Score (MOS). These parameters are used for analysis and to measure of the sound quality at the receiving end of a communication circuit and also to measure performance of VoIP application over the internet.

This research of analyzing the performance on Instant Messaging over the internet and done to analyzing on the internet broadband connection and also dial-up access. The outcome of this analysis will be conclude what is the best VoIP application should used and also the type of internet connection are available today to support VoIP application over the internet. The solution how to increase the performance of voice conversation will be discussed at the end of project analysis.

1.2 Problem statements

In order to use the Internet for voice communication besides data transmission, a voice-over-IP communication system needs to be developed that reduces the communications, system and maintenance costs at the end user's. VoIP application is a technology that allows voice communication using Internet Protocol (IP).

i. The Limitations

The scalability of Voice conversation is limited. That meant only one conversation able to running at the one time. The capability of the network connection also are limited because when another program is running during voice conversation is running, the quality of VoIP application is worst.

ii. Normal users have no knowledge about quality of application

Many of us today use the internet, but in reality don't have any idea what the instant messaging is best to used. They talk about the problem connection of the internet, but don't understand what was going on.

iii. There are many Instant Messaging providers that provide their services

Many of VoIP application can be download and use as free, but the user didn't know which one is better for communicate without any disruption.

iv. Connection medium are distinguished by their bandwidths limitations

Many of internet connection are provided in the market, but user don't know which connection can support for VoIP conversation.

1.3 Objective

There are several objectives that will be achieved throughout this project. The objective of this research and analyzing are:

i. To compare and analyzing performance on different VoIP application by testing these application in streamyx broadband and dial-up Modem access.

ii. To choose what is the best VoIP application available today's betweenYahoo Messenger and Skype Messenger.

iii. To measures the rate quality of voice conversation. The parameter are analyzed is packet loss, jitter and Mean Opinion Score (MOS) in different type of VoIP application.

1.4 Scopes

This project is not to develop any tools for testing, analyzing the performance, measurement and capturing the data of the voice conversation over the internet. What should be the scopes of this research are like follow:

i. Researches are focus to measures the voice quality and performance of VoIP application on Internet broadband connection and dial up Modem connection for simulating VoIP traffic between both computers somewhere on the Internet.

ii. Measurement of Packet loss, Jitter, Mean Opinion Score of every VoIP application technology overview used by different Instant Messaging provider today will be covered one by one during this research.

iii. Using various tool on the internet to measures quality and performance of VoIP application. These tools will be tested one by one so that different outcome will be collected.

iv. The implementation will be done in this research is by installing the correct tools for each client PC to measure data over internet from server.

v. Internet broadband connection and dial up modem connection will be used in this research. The data of this different type of medium will be capturing with appropriate tools.

1.5 Project Significance

Finally, the result of this research of analysis of quality and performance of VoIP application is to study, analyzing, capturing, comparing, testing and measuring every aspect of Instant Messaging technology that involved over the internet. From this research hope brings the findings that can be documented carefully to be a reference to users and industrial communities. This research should end up with solutions, suggestions, and recommendations to support every statement that are included in this project.

1.6 Conclusion

In the current market, there are many types of VoIP application which can be used by users for many purposes depend on their needs. All user requirements regarding VoIP performance application should be collected and should produce best application that can give the users more satisfied using this application. So for the next activities, literature review and project methodology will take place. In literature section, search, collect, analyze and draw a conclusion from all debates and issues in relevant body of literature are made. The sources can get from books, journals, technical reports, web pages and others which related to this project. Project Methodology is a way to use all available technique, tools and approaches used to achieve predetermined objectives.

CHAPTER II

LITERATURE REVIEW & PROJECT METHODOLOGY

2.1 Introduction

This chapter will focus on literature review and project methodology. Literature review are based on searching, collecting, analyzing and drawing conclusion from all debates and issues raised in relevant body of literature such as books, journals, technical reports, web pages and e-books. A methodology is a multi-step approach to the analysis, design, and delivery of an Information System.

2.2 Fact and Finding

In this part, many researches have been done to collect information related to this project. There are plenty of researches could be found in Internet, e-Books, journals and etc. This chapter presents an overview of several basic models that can use in analyzing

VoIP application. This overview provides some guidance with respect to when and why particular parameters should be selected.

2.2.1 Overview

According to *Tanembaum (1999)* More than 30 years ago, Internet didn't exist. Interactive communications were only made by telephone at PSTN line. The data exchange or transfer was expensive and the concept of video interactions was not even known. It has been few years that the communications world changed a lot. The usage of computers increased tremendously and new technologies to communicate like cell phones and finally, the advent of the internet made communication so easy and comfortable. People began to communicate with one another with new services like e-mail, chat, etc. Today, a real revolution is seen in the communications world, people use PCs and internet for professional needs and free time to communicate one another, to exchange data, like images, sounds, documents, etc. and also to talk to each other using applications like Netmeeting or Internet phone.

Voice-over-IP, or more simply VoIP is a term used in telephony for a set of facilities for managing the delivery of voice information using the Internet Protocol (IP). Generally, telephone service is used for voice communication and the Internet, for data transfer. The Internet does not directly support voice transmission due to bandwidth and real time limitationss. People, who have to use both the telephone and Internet for voice and data communications respectively, have to pay for both the services separately to their respective telephone and Internet service providers, which in turn increases the communications costs.

Moreover, the cost of voice communications through telephone increases with distance between the caller and the called party and the time of the call. Most of the businesses that have branch offices around the country or world have to invest a lot on voice and data communications. The internet users with a fixed or flat rate per month can save money with the help of an internet telephony technology like VoIP application that allows them to communicate through voice and transfer data too, using the Internet Protocol (IP). Companies with branch offices can save more money by using only Internet for long distance calling, data transfer, video conference and etc.