TESIS^ APPROVAL STATUS FORM

JUDUL: VOICE MESSAGING FOR HOME LAN

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Tesis dimaksudkan sebagai Laporan Projek Sarjana Muda (PSM)

VOICE MESSAGING FOR HOME LAN

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This report is submitted in partial fulfillment of the requirements for the
Bachelor of Information and Communication Technology
(Computer Network)

FACULTY OF INFORMATION AND COMMUNICATION TECHNOLOGY
KOLEJ UNIVERSITI TEKNIKAL KEBANGSAAN MALAYSIA

ADMISSION

I admitted that this project title name of

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ABSTRACT

The purpose of this project is to develop a Voice Messaging System for Home LAN to reduce telephone usage and to proof that voice can be send through IP using the VOIP concept. The methodology in developing this system is referred to the waterfall method that one of the SDLC. This method is use because it has a suitable phase that match with the PSM timeline and the system requirement to be developed. This application will be developed on Microsoft Visual C++. This project is important to be developed because it helps in researching an Internet protocol (IP), which is apart of TCP/IP architecture and an effective way to bring data voice in network. Although, from this project it can help to examine how the voice of data can reach to the destination within minimize time such as normal telephone usage. This application also will be develop by using higher level of TCP protocol like UDP where it providing in real time application.

ABSTRAK

Tujuan projek ini dibangunkan adalah untuk aplikasi Voice Messaging For Home LAN yang mana untuk membuktikan bahawa data berbentuk suara boleh dihantar melalui IP dengan menggunakan konsep VOIP. Bagi membangunkan projek ini satu metodologi pembangunan sistem telah diguna pakai. Metodologi yang digunakan adalah model air terjun dimana ia adalah sebahagian daripada elemen Kitaran Pembangunan Sistem (SDLC). Kaedah ini amat sesuai digunakan kerana ia mengadungi fasa-fasa yang boleh disesuai dengan jangkamasa pembangunan projek PSM dan keperluan sistem. Sistem applikasi ini akan dibina dengan menggunakan persekitaran bahasa pengaturcaraan Visual C++. Projek ini amat penting dibangunkan kerana ia membantu di dalam mengkaji penggunaan IP yang merupakan sebahagian daripada senibina TCP/IP dalam keberkesanannya membawa data berbentuk suara. Seterusnya ia juga dapat mengkaji mengenai tentang bagaimana data berbentuk suara itu sampai ke destinasi dalam jangkamasa waktu yang paling minima seperti penggunaan telefon biasa.Aplikasi yang dibangunkan ini juga menggunakan protokol TCP pada tahap tinggi seperti UDP yang menyediakan servis aplikasi masa nyata.

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ACRONYM

KUTKM	Kolej Universiti Teknikal Kebangsaan Malaysia
PSM	Projek Sarjana Muda
SNMP	Simple Network Management Protocol
LAN	Local Area Network
WAN	Wide Area Network
VOIP	Voice Over Internet Protocol
PC	Personal Computer
ICT	Information Communication Technology
IP	Internet Protocol
TCP	Transmission Communication Protocol
NMS	Network Management System
IT	Information Technology
CRC	Critical Request Check
PPP	Peer to Peer Protocol
ATM	Asynchronous Transfer Mode
WAN	Wide Area Network
MIB	Management Information Base
HTTP	Hyper Text Transfer Protocol
SSH	Secure Shell
HDLC	High-level Data Link Control
UTP	Unshielded Twisted Pair
HMP	Host Monitoring Protocol
ICMP	Internet Control Message Protocol
SDLC	System Development Life Cycle
FTP	File Transfer Protocol

UDP	User Datagram Protocol
NIC	Network Interface Card
MAC	Media Access Control
NIC	Network Card Interface
OS	Operating System
RAM	Random Access Memory
MB	Mega Byte
KB	Kilo Byte
PSM1	Projek Sarjana Muda 1
PSM2	Projek Sarjana Muda 2

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CHAPTER I

INTRODUCTION

1.1 Project Introduction

Voice Messaging for Home LAN (Local Area Network) is an extensive subject, but at the core it comes down to trying to transport speech signals in an acceptable way from sender to destination over an IP network. An Internet Protocol (IP) network is a computer network that uses the IP protocol to transmit information. The definition of acceptable depends on the particular situation dealing with. If, for example speech signals are being transported as part of a real-time communication between two persons, it will mean that the real-time aspects of this conversation must be respected. The overall delay between sending and receiving should be low to avoid irritably long gaps of silence. If, however speech signals are being transmitted as part of a one-way process for example on-line radio show or a lecture, the delay constraints are less strict since the interactive aspect is no longer present.

1.1.1 Project Formulation

Conventional way to communicate with each other using IP-networks is through the use of textual chat facilities. The purpose of this project proposal is to take this one step further by using voice communication instead of these textual facilities. The goal of this proposal is to perform research and development in order to let persons that are in the same virtual environment talk to each other, as they would do in reality. Their positions and orientations can be used to vary the intensity of the words, persons close to each other will hear each other clearly persons which are moving away from each other will understand each other less and less as their distance increases. It should be clear from this description that the real-time aspects of voice message would be very important. A virtual environment in which persons can communicate with each other and so the overall delay between talking at one end and hearing what is said at the other end should be as small as possible.

1.1.2 Uses of Voice Messaging for Home LAN

The first kind of use is the telephone alternative. This means that one person would use some kind of voice message system to make a voice call to another person with full duplex methodology. First of all, if a PC that can be connected to some kind of network is available, it can be used to make a call to somebody else who is also connected to that network. This PC would then be equipped with speakers and a microphone and some voice message application would be used to make the call.

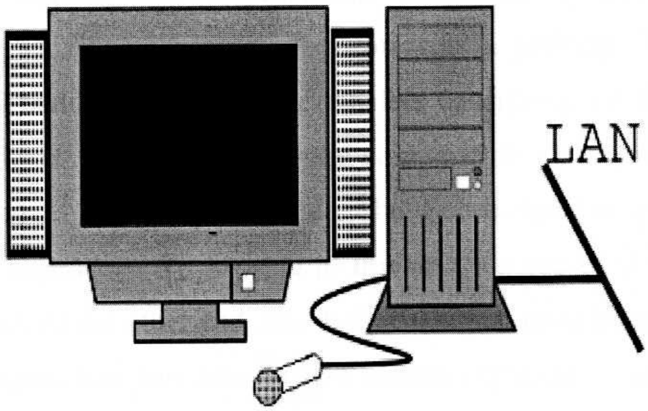

Figure 1.1: PC to LAN configuration

1.2 Project Objective

- To develop voice communication between end-to-end user in LAN radius.
- The Voice Messaging for Home LAN will generate without using IP Telephony.
- This Voice Messaging for Home LAN tool also can decrease the noisy and conversations in network environments.
- Also, IP networks do not rely on a separate signaling network, which is vulnerable to outages.
- Speed up the sending important data in voice and clear and guarantee the personal secret to person on LAN.

1.3 Project Scope

The project scope for the development of the Voice Messaging for Home LAN is simple where it uses only for intranet radius with full duplex methodology. The usable bandwidth of a shared Ethernet LAN is typically 8 bits 8 KHz duplex. Frequently, members of a workgroup share a LAN segment. An Ethernet LAN segment therefore has a capacity of approximately 1 to 2 voice channels.

Two people holding four simultaneous conversations can load a LAN by approximately 10%. This number, while low, presents a problem. Voice connections rely on a guaranteed bandwidth. Large file transfers consuming all the bandwidth of a LAN can severely impact voice traffic. Several components are required to make Voice Messaging for Home LAN in intranet possible. The speech signal is split into tiny pieces that are transmitted separately. To be able to transmit a piece of the speech signal, it must first be digitized. At the other end, this digitized signal must be reconstructed into a continuous speech signal that can then be sent to some speakers. There will be several signals may have to be mixed together if several persons are talking at the same time either at the sender or at the receiver. To reduce the amount of required bandwidth to transmit the signal, the digitized speech signal should be compressed. At the other end it must be decompressed before it can be processed. Finally there must also be a component, which handles the transmission and reception of packets containing speech data.

1.4 Project Priority

Voice Messaging for Home LAN is about transmitting a voice signal across an IP network (the Internet for example). The context of this voice signal determines constraints for this transmission. For instance, if this voice signal is a part of a conversation between two people, care must be taken to preserve its real-time characteristics. The delay between one person talking and the other person hearing what was said should be as low as possible to avoid irritable gaps in the communication. Other applications of Voice Messaging for Home LAN like an on-line lecture do not have this delay constraint.

The classical use for voice messaging is as a replacement for a telephone call. Using voice messaging like this can reduce costs in various ways, but the quality of the conversation is usually lower than that of a normal telephone call. Using Voice Messaging for Home LAN environments is relatively new. Such applications would allow users to chat with each other, like on IRC, but instead of typing messages to each other they could simply talk with other users. Other applications use similar techniques as VoIP, for example the transmission of a video signal.

1.5 Conclusion

This project is about voice messaging in networked LAN environments. It contains information about VoIP (Voice over Internet Protocol) in general and its application in LAN environments. It will also describe the application that is developed to test aspects base on VoIP in LAN environments. It just take the concept of VoIP in how voice work in network with using the correct protocol.

CHAPTER II

LITERATURE REVIEW

2.1 Introduction

Nowadays network software is usually very structured. This section is about the way this software is organized. It also contains about the layered design model, which is a good example of this structured design and about the TCP/IP reference model, in which as the name suggests IP plays a very important role.

2.1.1 Layered Design

To facilitate the design of network software, usually the approach of a layered design is used. In this approach, each layer provides a certain functionality, which can be used by the layer directly above. There are several advantages to this approach.
First of all, the software is much easier to design. Trying to implement the desired functionality all at once will be very difficult and will probably result in many flaws in the program. Furthermore, these flaws will be difficult to track. By dividing the software in layers, the important thing should be to worry is about implementing some functionality for each layer. By using a structured approach, it will be able to tackle it more efficiently.