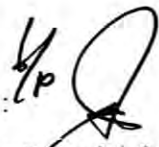


“I hereby declare that I have read this thesis and in my opinion, it is suitable in term of scope and quality for the purpose of awarding a Bachelor Degree in Electronic Engineering (Computer Engineering).”

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Supervisor : Zahariah binti Manap  
Date : 5/5/06

**THE PROTOCOL AND IMPLEMENTATION OF VOICE OVER IP (VOIP)  
IN MALAYSIA: A CASE STUDY.**


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**This Report is Submitted in Partial Fulfillment of Requirements for the Bachelor  
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“I admit that this is done by myself except the discussion and extracts taken from other sources that I explained each in detail.”

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**Dedicated to abah and mama**

## AKNOWLEDGEMENTS

I would like to express my gratitude to all those who gave me the possibility to complete this thesis. I am deeply indebted to my supervisor Pn Zahariah whose help, for giving me permission to commence this thesis in the first instance, stimulating suggestions and encouragement helped me in all the time of research for and writing of this thesis. I would also like to thank Mr. Soo Yew Guan with his support to give millions comment and support to my thesis, without his touch I could not have had this far.

I would like to acknowledge the help of many people during my study especially Pn Siti Seha (kak t) from TM Johor Bahru, Mr. Suren from Huawei Technologies Malaysia for helping to supervise me, providing resources and subjects, and offering direction; Mr. Sharil Johal and Mr Halim Dahlan in providing me idea and guide me to understanding the concept of this title; to all my friends and numerous others for particular help, advice or discussion.

This dissertation could not have been written without Mr. Fajar who not only served as my lecture but also encouraged and challenged me throughout my academic program. Thank you doesn't seem sufficient but it is said with appreciation and respect. I owe a great deal to Abg Sofian thank you for your patience, support, favors and all the other things that make it so worthwhile to know you.

Lastly, and most importantly, I wish to thank my parents, mama and abah. I am grateful also to the examples of my sister and brother in law, Kak Ngah and Abg Don, without their patient and support through my study from the very begin I cannot ever finish my study in University. I cannot end without thanking my family (also from Penang and Malacca), on whose constant encouragement and love I have relied throughout my time at the University. It is to them that I dedicate this work.

## ABSTRACT

Over the past decade, the telecommunications industry has witnessed rapid changes in the way people and organizations communicate. Many of these changes spring from the explosive growth of the Internet and from applications based on the Internet Protocol (IP). The Internet has become a ubiquitous means of communication, and the total amount of packet-based network traffic has quickly surpassed traditional voice (circuit-switched) network traffic. In the wake of these technology advancements, it has become clear to telecommunications carriers, companies, and vendors that voice traffic and services will be one of the next major applications to take full advantage of IP. This expectation is based on the impact of a new set of technologies generally referred to as voice over IP (VoIP) or IP telephony. VoIP is a technology that allows telephone calls to be made over computer networks like the Internet. VoIP converts analog voice signals into digital data packets and supports real-time, two-way transmission of conversations using Internet Protocol (IP). There are three basic protocols which are used to implement a voice over IP solution: H.323, SIP and RTP. H.323 and SIP are mainly concerned with the call establishment and voice encoding, whereas RTP (Real-time Transport Protocol) is used by both protocols for the transport of encoded voice packets over an IP network. This paper is intended to study the technology and its benefits, characteristics, protocols, architectures, differentiates with traditional PSTN, all the hardware and software both the IP technology and its applications, and how Malaysia are approaching IP-enabled technologies, particularly VoIP, and the implement issues surrounding VoIP. In addition, this study case will include the latest technology that support VoIP like VoIP with WIFI, and also satellite.

## ABSTRAK

Lebih sedekad yang lalu, industri telekomunikasi telah menyaksikan pecutan perubahan bagaimana pengguna dan industri berkomunikasi. Kebanyakan perubahan ini adalah bersandarkan kepada penumbuhan pesat penggunaan Internet dan aplikasi berdasarkan Protokol Internet (IP). Internet telah menjadi satu kebiasaan dalam erti kata perhubungan manakala jumlah rangkaian paket-paket dengan pesatnya telah berubah melebihi rangkaian suara konvensional. Dengan kemajuan pembangunan teknologi ini memberikan gambaran jelas kepada organisasi-organisasi, pembekal serta syarikat-syarikat telekomunikasi bahawa perkhidmatan panggilan suara akan menjadi aplikasi utama menggunakan kelebihan IP. Jangkaan ini berdasarkan kepada impak teknologi baru iaitu Panggilan Suara melalui Protokol Internet (VoIP) atau dikenali juga Telefon IP. VoIP merupakan teknologi yang membenarkan panggilan suara melalui rangkaian komputer seperti internet. Ia menukarkan isyarat analog suara kepada paket-paket data secara digital dan menyokong masa nyata (real time) dan perhubungan dua hala melalui IP. Terdapat tiga protokol – protokol asas yang digunakan untuk melaksanakan VoIP: H.323, SIP dan STP. H.323 dan SIP lebih menumpukan kepada bagaimana panggilan suara dilaksanakan dan pengkodan suara. Manakala RTP digunakan SIP dan H.323 untuk menghantar suara yang telah dikodkan melalui rangkaian IP. Kajian ini akan mengkhususkan kajian kepada ciri – ciri, seni bina, kelebihan, aplikasi, protokol-protokol, perbezaan dengan talian konvensional (PSTN), perkakasan dan perisian yang digunakan, serta pelaksanaan VoIP di Malaysia. Sebagai tambahan, kajian turut meliputi bagaimana teknologi lain menyokong VoIP seperti WiFi dan satelit.

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## ABBREVIATION

DSL	Digital Subscriber Line
DVB	Digital Video Broadcast
FTP	File Transfer Protocol
GPRS	General Packet Radio Service
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPSAT	Internet Protocol Satellite
ISDN	Internet Service Digital Network
ISP	Internet Service Provider
ITSP	Internet Telephone Service Provider
ITU	International Telecommunication Union
LAN	Local Area Network
MAC	Media Access Control
MEGACO	Media Gateway Control
MGCP	Media Gateway Control Protocol
NIC	Network Interface Card
PBX	Public Branch eXchange
POTS	Plain Old Telephone Service
PSTN	Public Switch Telephone Network
RAS	Registration, Admission & Status
RTP	Real-time Transport Protocol
SAP	Session Announcement Protocol
SIP	Session Initial Protocol

Sigtran	Signaling Transport
SS7	Signaling System 7
TDM	Time Division Multiplexing
TCP	Transfer Control Protocol
TCP/IP	Transfer Control Protocol / Internet Protocol
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
WAN	Wide Area Network
WiFi	Wireless Fidelity



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## CHAPTER I

### INTRODUCTION

#### 1.1 OVERVIEW

To most of us, VoIP may sound like a new advancement in technology, but it has actually been in Malaysia for a decade. As Malaysians started embracing the Internet in 1995, the more sophisticated computer users among us began to venture into the world of VoIP. Most toyed with software to make PC-to-PC voice calls, and while it was considered a very cool thing to do, it was not very effective as a means to communicate.

Compared to now, consumer-level Internet connections were pretty poor back then if we wanted VoIP, we would have to put up with conversations that were disrupted by hissing sounds and lag. However the situation was different for VoIP companies, who could use their own networks for transporting voice data instead of the Internet. And so, several companies in Malaysia started offering VoIP services by selling prepaid cards (or post-paid accounts to businesses) that promised cheaper international calls. This also gave rise to the trend of call shops in Malaysia, especially with the increase of foreign workers and students who needed a more affordable way to call back home.

However, the Internet connection quality has significantly improved in the past few years, and with the advent of broadband in Malaysia, VoIP has become more practical for households. The Internet has become a ubiquitous means of communication, and the total amount of packet-based network traffic has quickly surpassed traditional voice (circuit-switched) network traffic. In the wake of these technology advancements, it has become clear to telecommunications carriers, companies, and vendors that voice traffic and services will be one of the next major applications to take full advantage of IP. In fact, last year itself we saw a few local companies enter the consumer-level VoIP scene by offering VoIP packages catered to homes and small businesses. But as these packages are only limited to PCs connected to the Internet, it makes VoIP calls less convenient than using a cell phone. Fortunately, this situation may yet change with the arrival of mobile VoIP and VoIP prefix number 015 from several Malaysia's provider.

## **1.2 TITLE AND OBJECTIVES.**

The title of this project is 'The Protocol and implementation of voice over IP (VoIP) in Malaysia: A case study. There is more than one major of VoIP providers in Malaysia using the same protocols but some of them are different in implementing of this technology. The main purpose of this research is to specialist study in implementing VoIP in Malaysia. Thus, it will include:

- To describe the protocol used to implementing VoIP
- To review and list the software and hardware.
- Identify the application and technology
- To define and differentiate with PSTN technology
- Identify the importance and benefit of VoIP to user in Malaysia.

## **1.3 SCOPES OF WORKS**

### **1.3.1 Protocol**

#### **1.3.1.1 H.323**

International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendation H.323 describes the architecture to support multimedia communications over networks without quality of service (QoS) guarantees. Originally intended for LANs, H.323 has been adapted for IP.

#### **1.3.1.2 MEGACO**

The MEGACO protocol is used in environments in which a media gateway consists of distributed subcomponents and communication is required between the gateway subcomponents

#### **1.3.1.3 MGCP**

Defines a call control model that controls VoIP gateways from an external call control element or call agent

#### **1.3.1.4 SIP**

Describes a multicast mechanism for advertising the session characteristics of a multimedia session, including audio and video

### **1.3.1.5 Real-time Transport Protocol (RTP)**

Used by H.323 and SIP protocols to transport of encoded voice packets over an IP network. Also defines a format for different audio and video encodings to promote interoperability among different computer platforms, operating system and application software products.

## **1.3.2 Service**

### **1.3.2.1 IP Telephony**

The two-way transmission of voice over a packet-switched IP network, which is part of the TCP/IP protocol suite.

### **1.3.2.2 Internet Telephony**

Is another term for IP telephony and VoIP. Internet telephony referred to voice over the public Internet, while VoIP referred to voice over private IP networks.

### **1.3.2.3 LAN Telephony**

An IP telephony system that is controlled within a local area network (LAN). More appropriate for medium to large enterprises that wish to use their existing IP network to save money on voice calls.

### 1.3.3 Applications

#### 1.3.3.1 PC to PC

VoIP applications that end to end user are PC as tools to communicate each other. Apparently need VoIP software (Softphone) that sometimes provided free by VoIP provider.

#### 1.3.3.2 PC to Telephone

This type of application need regular phone hardware or special IP phone that can uses to received or calling to VoIP number

#### 1.3.3.3 Telephone to Telephone

VoIP applications that uses regular or IP phone that can be made like normal call using PSTN.

### 1.3.4 Transmission

VoIP use various ways in order to transmitted over. Transmission line should support as voice packet are big enough than data even it had converted. There are several transmission types' uses to implement VoIP such as:

- **Over PSTN**
- **Broadband**
- **Satellite**
- **WiFi**

### 1.3.5 Differentiates between VoIP and PSTN

To understand the VoIP operation and to show its benefit, the differences of this technology must be compared to PSTN. These will include:

- Architecture
- Switching
- Transfer rate
- Noise (if any)
- Bandwidth

I'll cover these scopes of work in chapter IV for more fine points.

## 1.4 THESIS STRUCTURE

Chapter I (on this part) discusses about introduction and overview about background VoIP in Malaysia. It also includes objective and scope of works for this thesis.

Chapter II discusses on project literature including VoIP and PSTN background, how VoIP and PSTN related and PSTN protocol.

Chapter III discusses on project methodology used to achieve the study of this title including the flow chart of the methodology.

Chapter IV is the most important, discusses on the result and findings of the thesis. It includes VoIP protocol and standard, how VoIP is implemented in Malaysia, what transmission is used, comparison to PSTN and also its benefit.

Chapter V is the last part but not least for this thesis which states the discussion and conclusion.

## **CHAPTER II**

### **PROJECT LITERATURE**

#### **2.1 INTRODUCTION**

##### **2.1.1 The Telephony Network**

From a user's perspective, a telephone network is a big cloud that connects the originating points for transmitted information with terminating points for information. A network is typically made up of nodes interconnecting transmission links to form transmission paths for the transmitted information to follow. The transmission links in typical network often use different transmission modalities (e.g. wire, wireless, fiber).

The nodes may contain electronic devices, such as switches, which establish a temporary or permanent path to be taken by the communicated information traveling from one link onto another link, and/or those nodes may contain conversion devices that change the modality of the transmitted signal to allow it pass between links using different modalities (e.g. connected a wired link to a wireless link, an analog link to a