

Voice over IP (VoIP) Implementation in UTP Campus Network

by

Mohd Saufy Bin Rohmad

Dissertation submitted in partial fulfilment of
the requirements for the
Bachelor of Technology (Hons)
(Information Technology)

JUN 2004

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TK

5102.5

M 697

2004

1) Voice over Internet Protocol
2) IT / IS -- Thesis

CERTIFICATION OF APPROVAL

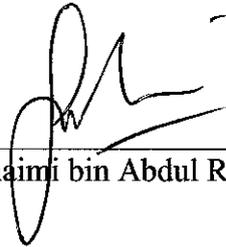
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A project dissertation submitted to the
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Approved by,



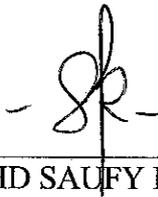
(Suhaimi bin Abdul Rahman)

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CERTIFICATION OF ORIGINALITY

This is to certify that I am responsible for the work submitted in this project, that the original work is my own except as specified in the references and acknowledgements, and that the original work contained herein have not been undertaken or done by unspecified sources or persons.

A handwritten signature in black ink, consisting of a stylized 'S' and 'R' with a vertical line through them, flanked by small horizontal dashes.

MOHD SAUFY BIN ROHMAD

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ABBREVIATIONS AND NOMENCLATURES

FYP – Final Year Project

UTP – University Technology of PETRONAS

VoIP – Voice over Internet Protocol

PSTN-Public Switching Telephone Network

DHCP-Dual Host Configuration Protocol

IVR- Interactive Voice Response

LAN-Local Area Network

MAN-Metropolitan Area Network

WAN-Wide Area Network

TCP-Transmission Control Protocol

E.164- The ordinary telephone number system e.g.: 05-3687412

LDAP- Lightweight Directory Access Protocol, a set of protocol for accessing information directories.

GNU-short for Gnu's not unix, a unix-compatible software system developed by the free software foundation (fsf).

MCU-Multiple Control Unit

MySQL-Open Source Database System

DNS-Domain Name System

CDR-Call Detail Record

CISCO IOS - CISCO Interconnecting Operating System

ABSTRACT

The purpose of this paper is to make the fundamentals research on the Voice over Internet Protocol (VoIP) technology with the implementation of the VoIP system in the UTP network. The problem that been arise before the idea of this paper is the communication problem faced by student and staff when using the PSTN Networks. The difficulties to communicate across PSTN and IP networks are the main problem occurs in our campus networks. The final outcome from this project is the Voice over Internet Protocol (VoIP) system that implemented through out the campus. This study also includes the implementation of intelligent mechanism to authenticate and give user access to the VoIP network. In this paper, the author is using the incremental development model for the implementation and using the common research method such as book review, journal reading and laboratory testing. This implementation may open the new dimension of the communication paradigm in our campus life.

ACKNOWLEDGEMENT

Alhamdulillah, out of a mixture of puzzlement, daring and excitement, I have completed the my Final Year Project (FYP). This project would not have been possible without the vital help of Universiti Teknologi Petronas (UTP) lecturers, students and those who involve directly or indirectly for the conceptual contribution in the research and development of this project. It does not only construct knowledge of or for them, but with them. First of all I would like to express me deepest gratitude to parents, Rohmad Fakeh and Dalila Yusuf, for giving me encouragement, inspiration and valuable comments on the development of my ideas despite lots of unexpected difficulties. Also for my supervisor, Mr. Suhami Abdul Rahman, for the idea and moral support. I also owe a special thanks to him who checked the every word of my writing. I also wish to thank Mr. Shuib Basri, Mrs. Vivian and all other UTP lecturers for your advice, encouragement and help. Without them, it would have been impossible for me to finish this dissertation. This acknowledgement also dedicated to all staff of the UTP especially Mr. Ruslan Idris for their dedication and assistance in maintenance and completing my project. I appreciate very much the companion and help I got from past and present graduate students, laboratory technicians, researchers, and secretaries I had privilege to meet and work with. Finally, and definitely not least I wish to acknowledge all of my friends who have listened to me bitch, moan, rejoice, complain, laugh, cry, and ponder my way through these past years. All of them deserve to sign this dissertation.

CHAPTER 1

INTRODUCTION

1.1 BACKGROUND OF STUDY

The development of the telecommunication world had made the relationship more closely to each other. The emerging of various internet technologies had made the usage of the Internet spread worldwide. The blend integration of the data and voice network begin to burst the internet technology usage.

The invention of fast-growing computer networking ‘language’, TCP/IP protocol stack had improved the way human transfer information, communicate and interact in the virtual world. Discovery of this packet switching based network is originated from the usage of the traditional telephony system that had been use long time ago.

1.1.1 Why Internet Protocol (IP)?

Why then is IP of such the interest to the communication industry? There are four major reasons for this interest, and for the deployment of IP telephony. There are a lot of advantages of using IP telephony rather using other system.

1.1.1a. The Business Case

The first reason is a convincing business case for the deployment of the IP protocol suite and associated equipment to support telephony services. This case can be summarized with three ideas.

i. Integration of Voice and Data. The integration of Voice and data make the communication is multimedia and the multiplication software may be produces. The usage of Web Server also can be expanding and this idea can make the Web Server interact with customer with data, video and voice.

ii. Bandwidth Consolidation. Branch from the first idea, the usage of the integration of data and voice allow for Bandwidth consolidation. It means the data communication channel is fills up more efficiently.

According to Uyless (1999)

“The commonsense idea here is the implementation used IP network is to migrate away from the rigid telephony based timer division (TDM) scheme wherein a telephony user is given bandwidth continuously, even when the user is not talking. Since voice conversations entail a lot of silence (pauses in thinking out an idea, taking turns talking during the conversation, etc.), using the data communications scheme of statistical TDM (STDMA) yields a much more effective use of precious bandwidth.”

iii. Tariff Arbitrage and Beyond. This term mean we are bypassing the public switching network toll to reduce the cost of the transmission. The network that utilizes the internet backbone wick make the cost is decreased. In future chapter we will see how this low charge is implemented. This approach avoids the costly long distance charges incurred in the tariff telephone network in contrast to lower costs of the untariffed Internet.

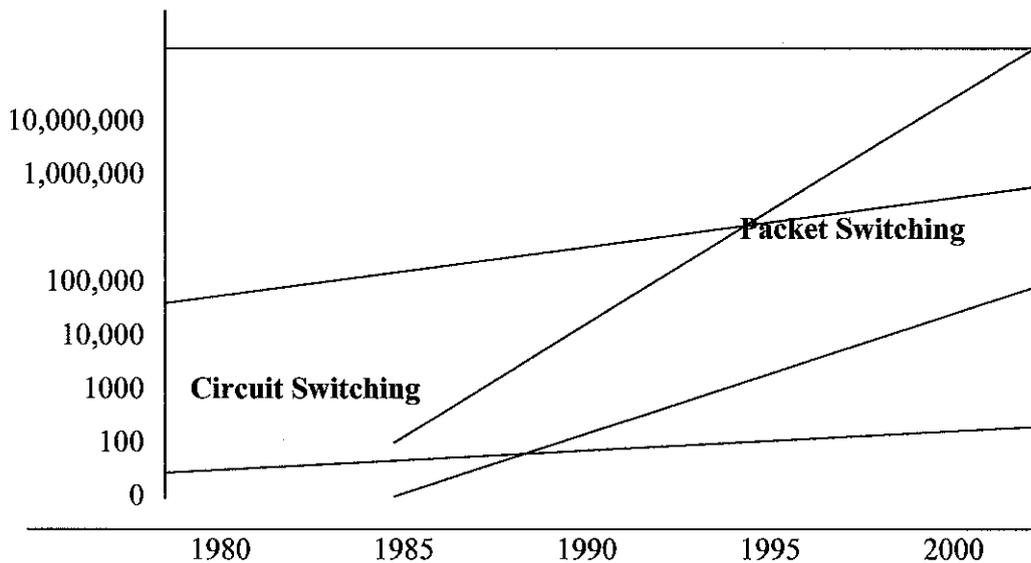


Figure 1.1 Cost Performance: Bit/s per dollar [2]

According to Uyless (1999)

“Some studies that favor packet voice over circuit voice cite 3:1 or 4:1 cost ratio advantage of packet voice over circuit voice. And the ratio is considered conservative by some people. James Crowe, CEO of Level 3, has stated that VoIP calls cost 1/27 of circuit switched calls.

1.1.1b. Universal Presence of IP

The second reason why IP used widely as a medium of future communication is due to the universal presence of IP and associated protocols in user and network. Nowadays, the IP is resides inside the user PC, hand phone and maybe in the near future other things as PDA or maybe our car has its own IP.

1.1.1c. Maturation of Technologies

The third major reasons for the deployment of internet telephony is the maturation of technologies that now make IP telephony feasible. These technologies burst are supported by explode in Digital Signal Processors field.

1.1.2 Architecture of IP and Campus Network

The aim of the author study is to know how the VoIP system can be implemented, so the study about the IP protocol and Campus Network architecture is the most critical part of my project. The fundamentals knowledge about Computer networks, LAN, MAN and WAN is the key of this VoIP implementation.

The Internet is the complex of regional and national network that interconnected with router. A million of routers, LAN and MAN are connected together to perform this huge internet. And IP is an example of connectionless services. But the problem that exists in the connectionless oriented transmission is being solved when using other protocol such as TCP.

1.2 PROBLEM STATEMENT

From the background of study, the problem of the implementation of this VoIP is the integration of IP network with the traditional PSTN Network. The acceptance rates among the society in the moderate and third world country such. Malaysia is very low and we are loss in term of this technology. The business oriented company also always looks for the opportunity to charge high fees for VoIP services. The author see this is the major problem and challenge that we must face to make this technology is available widely.

In the campus daily life, the communication between the Student and lecturer is the major problem that faces by student. With the new academic complex, that located far compared to old academic building, make the time taken to lecture is more. And when the student wants to meet the lecturer, there are frequent cases that the lecturer is not at their place.

The current system problem includes:

- a. The disconnected network between student PC that used IP and staff telephone that used ordinary E.164 phone number.
- b. The local IP user is disconnected to the outside telephone system and can't reach outside UTP telephone end points.
- c. There are no middle man servers that translate the name of the endpoint to the IP number. So, every user needs to remember other user IP number.
- d. The unsecured connection between 2 end points and there are no intelligent server that can track the called that maid by the user.

1.3 OBJECTIVE AND SCOPE OF STUDY

The Author is proposed to implement the one of the component of next generation networks that is VoIP. Below is the Objective of this project.

1. Research on VoIP Technologies.
2. Implementation of CISCO Voice Module.
3. Integration with the Wireless Networks.
4. Billing and Management system Integration.

The Scope of this study is to explore the entire concept and behind the VoIP technology and Implement It in our campus environment. The Research made is based on the background technical knowledge that required in order understanding the actual concept. The technical study includes:

2. Video Coders and Analog Digital Conversion.
3. Traditional Telephony network and PBX.
4. Performance Issue in VoIP.
5. VoIP Gateway and Gatekeeper.
6. VoIP protocol type and interconnection.
7. Fundamentals reading on CISCO IOS.
8. Openh323 Open Source Software VoIP implementation.
9. Reading on Software Configuration Guide for CISCO Internetworking Devices.
9. Reading on CISCO IOS Voice Interface Module.
10. Integration of Openh323 with the CISCO system.
11. GNU Gatekeeper study and implementation.

CHAPTER 2

LITERATURE REVIEW AND THEORY

2.1 THEORY OF VOIP SYSTEM

If we can notice that our home regular internet dial-up services is actually connected from our phone line. How the circuit switching line is converted into packet that transmit inside it is the core of the understanding of the VoIP system. This simple analogy shows that the VoIP system is the communication lines that exist from the conversion of the analog and digital signals.

For this voice communication, the basic principle involved digitally sampling speech signals obtained from a handset or other audio device connected to the client, as shown on the below figure [3].

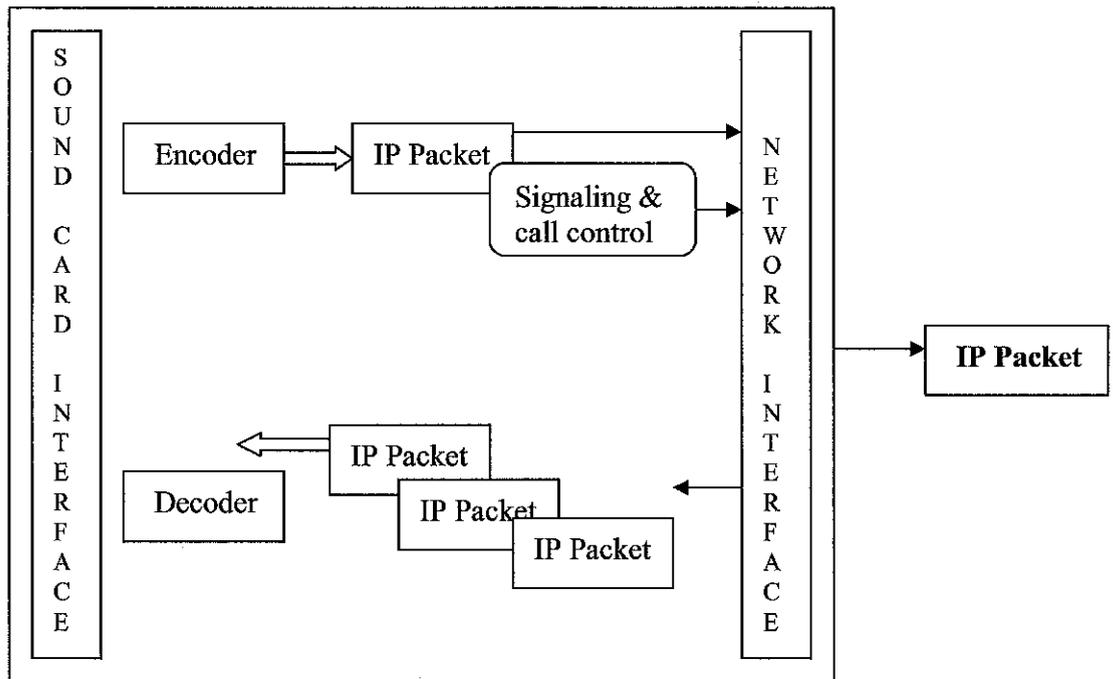


Figure 1.2 How VoIP client works [3]

According to Toby J. Velte (2001),

There are three basic types of VoIPs. They are designed around the user's specific needs and suit a specific market.

1. Simple toll Bypass

The Most Basic, straightforward use for VoIP is using it to make telephone calls without having to use the public switched telephone system (PSTN). This is ideal if you just want to use IP to transport calls between branch offices within the corporate network. This design requires minimal change to existing PBX, cabling and handset infrastructures, it is relatively easy to develop, and has no PSTN integration issues to worry about.

2. Total IP Telephony

This design relegates your existing voice systems to the dumpster. No longer will desktop have conventional telephone handset-instead., they're traded in on IP telephones that plug into Ethernet ports. You'll use LAN servers to provide the majority of the features your PBX now provides. This is the Holy Grail of VoIP and not a journey to begin on a whim.

3. IP-enabled PBXs

This solution isn't as gutsy as total IP telephony, but you still get a mélange of functionality. You don't have to change the existing cabling or handset, but you will upgrade the PBXs so that your organization core system can speak with IP telephony protocol. PBX users will be able to communicate with other IP telephony users, but the limitation is that your PBX will have to rely on IP telephony gateway to communicate with the conventional, public telephone system.

The types of VoIP implementation shows that we need to define our need and technologies capabilities before designing our system. From the theory the author can say that there are simple VoIP and more complex one that varies in term of implementation complexity and technical depth ness.

2.2 DATA AND VOICE NETWORK ISSUES

Imagine when we accessing only the simple webpage also have some connection delay and problem, how about we want the real time communication from east to the west part of the world. This dilemma is the main thing that plays around all telecommunication engineers around the globe.

As the idea is look great, but we need to look to the issues that related in order to provide full scale VoIP solution. There are a lot of factors that will influence the performance of your networks. The fine tuning the networks to support VoIP incorporates a series of protocols and features that improves quality of service (QoS) [4a].

According to Paul Drew from MetaSwitch and Christ Gallon from Fujitsu (March, 2003),

There are a few issues need to be address,

- What **services** to be offered? Are the fully equivalent PSTN services, a more 'cheap second line' or a simple user-to-user voice services.
- The type of **end user terminals** supported – POTS phone, PC client, IP Phones or PBXs
- **Quality of Service** requirement for voice to ensure that the agreed quality is provided.
- The **security** risks need to be identified and define **protection technique** to protect call agent.
- How much **bandwidth** available, that affect the choice of **codec**, **packetization period** and **compression** to use.
- The **signaling protocol** must support the services required.

2.3 GENERIC VOIP IMPLEMENTATION

The author has discussed the issues surrounding VoIP. Let us now look at some VoIP configuration and topologies. Several configuration options are available to support VoIP operation. [5]. These 5 configuration below is the general architecture used in any VoIP configuration and implementation.

2.3.1 Telephone connections with N: 1 gateway

This is the first type of implementation of VoIP that used the conventional telephone as well as the telephone networks and convert the signals through the gateway. On the transmission side, the voice is encode, compressed and encapsulate into data packets (IP datagram). And at the receiver gateway, the process is reversed. The gateway convert back the digital signal into DS0 signals. It called N:1 gateway because it multiplex the n number of telephone signals into IP datagrams onto one link to the internet of Intranet.

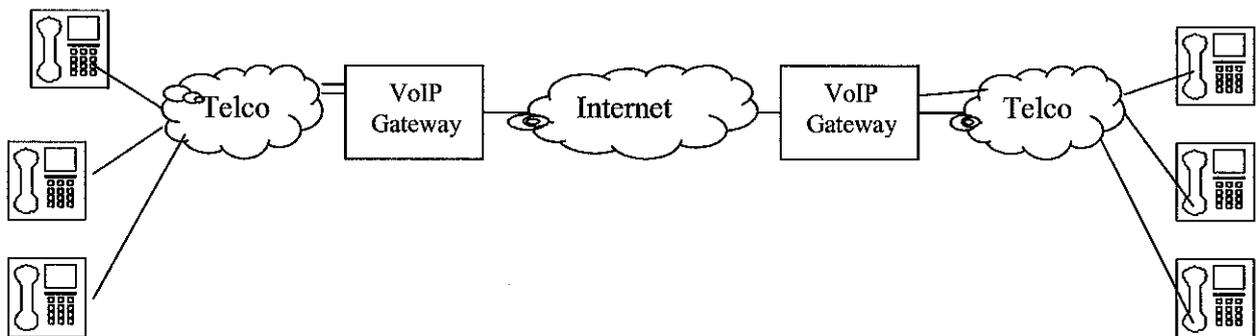


Figure 2.1 Telephone connections with N: 1 gateway

2.3.2 PC connections with router

This configuration option shows the use of personal computers (PC) and the deployment of router. This is the simple and typical VoIP networks that be implemented. This is simple and straight to the point approach to VoIP. All the encoding, compression and datagram is being done inside the sender PC. The router only transmits the datagram and route it to the destination. On the receiver end PC, the decoder and decompression will take place. The encoder protocol and standard is determined by the PC itself. But there are the noise problems that may occur in the encoder PC, so the voice quality is not efficient enough compared to the encoder performed by gateway.



Figure 2.2 PC connections with router

2.3.3 Telephone to PC connection

This is another option eliminate the usage of open microphone that cause the noise to the normal telephone. Here, the voice quality is enhanced but the PC still do the encoder (Analog to Digital) and decoder (Digital to Analog) same as the previous one.

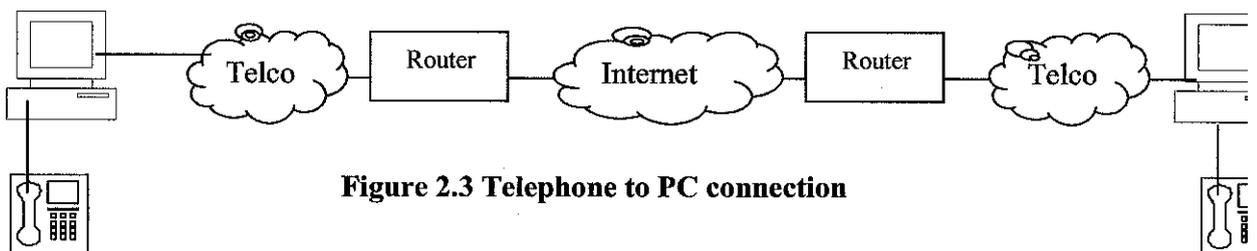


Figure 2.3 Telephone to PC connection

2.3.4 Connection with 1:1 gateway

This is the simple and low-cost VoIP options. This option can be develop in our home and personal Computers. We only need the voice modem and telephone as the hardware, and gateway software as the software installed. The 1:1 means that only one telephone is connected to the gateway. The gateway accepts the speech signal and performs the encoder operation. At the receiver, the reverse operation takes place.

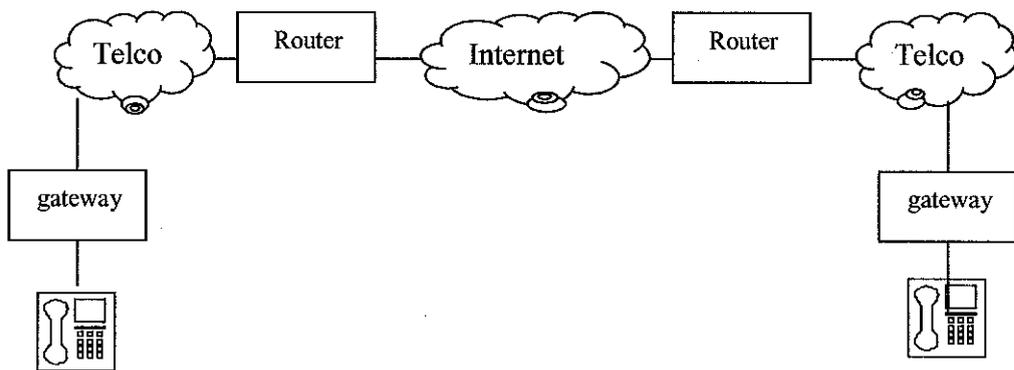


Figure 2.4 Connection with 1:1 gateway

2.3.5 PC-to-phone calls

This is the last configuration options for VoIP. This is the considerable the most gaining attention in the industry. In fact this is the variation of the configuration usec Telephone to PC connection and Connection with 1:1 gateway. First, this configuration does not require a getaway at each end of connection. Second, the users are attach to the local area networks at one side, and the local calls on the LAN are managed by the gateway. Inside the gateway (or another PC I the LAN), the management function is being performed. The PC inside the LAN networks can access the PSTN telephone by using the gateway and meet the telco (telecommunication networks) requirement.

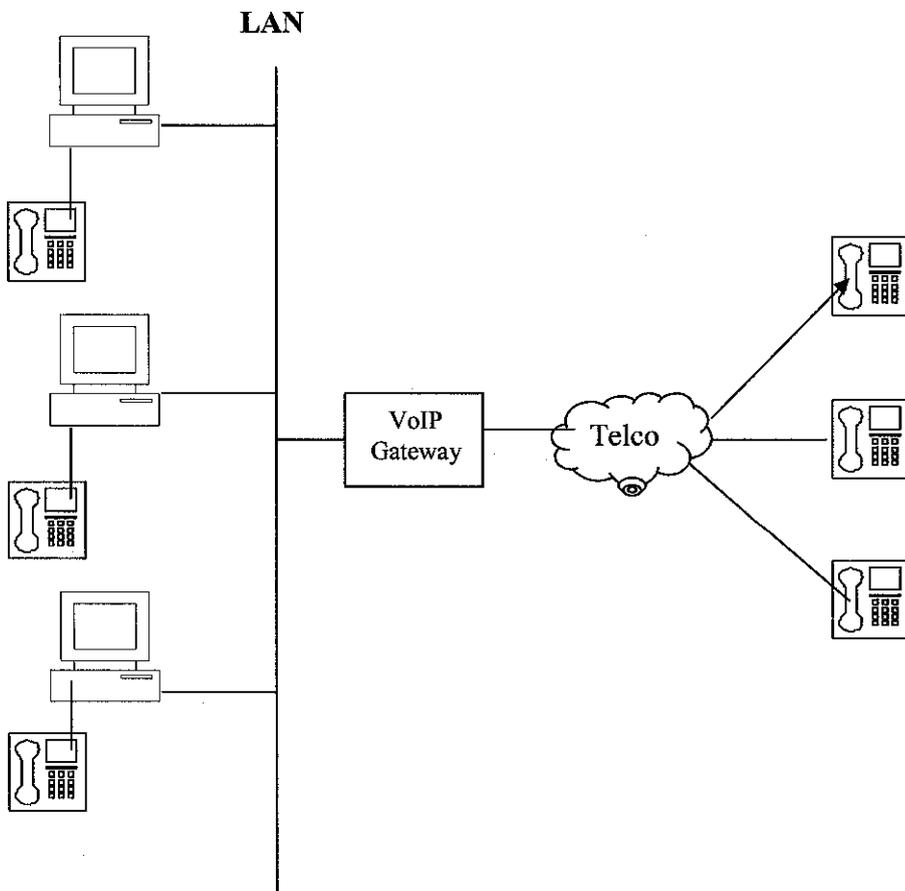


Figure 2.5 PC-to-phone calls

2.4 THE VoIP PROTOCOL STANDARD

The VoIP system also incorporating other tools and services such as gatekeeper and network servers that used widely in the complete VoIP implementation. We are not discussing detail on the internal protocol for the VoIP inside the TCP/IP protocol stack.

For the information on the depth protocol and IEEE standard compression that used inside the worldwide VoIP system, all system developer and vendor are consistently used the H.323 protocol stack and all various protocols inside it. [6]

Application	Terminal Control and Management			
Voice Codec	RTCP	H.225	H.225	H.245
RTP		RAS	Q.931	
UDP			TCP	
IP				
Data Link Layer				
Physical Layer				

Figure 2.6 The H.323 Protocol Stack

The H.323 protocol stack is the integrated protocol that contains a few other child protocols. The protocol within the stack is independent parts and stand alone, but can used and be used by other protocol.[1]

The function of each these IEEE protocol generally as describe as below:

- RAS manages registration, admission and status
- RTP/RTCP as the media for transport, RTP carries the actual media and RTCP carries status and control information.
- TCP is transport the signaling.
- Q.931 manages call setup and termination.
- H.245 negotiates the channel usage and capabilities
- H.235 handles security and authentication.

CHAPTER 3

METHODOLOGY

This chapter is the most important chapter that describes the methodology that involved in my implementation. There are main methodology, research methodology and implementation methodology. Although these three are overlap each other, the author try to differentiate it to make sure we are understand the actual work that involved until the final implementation of the system.

The main methodology is playing the role as the parent model of this system development. Take from our well know scholars in the area of Software Engineering, the author choose the model that very suit with my requirement and constrain.

3.1 MAIN METHODOLOGY

The main methodology is development benchmarking and framework that the author used generally in implementing this system. There are a lot of constraints, obstacle and academic requirement that need to be fit in selecting my main model.

Finally, the author decides to use the Incremental Development model approach in this study. The Incremental development is means of reducing rework in the development process and giving customers some opportunities to delay decisions on their detailed requirement until they had some experience with the system.[7] In the author case the incremental model is the best due to the lack of the knowledge about

the system and the ongoing research study that done concurrently with the development. The uncertainty about the final product that want to deliver also influenced this selection.

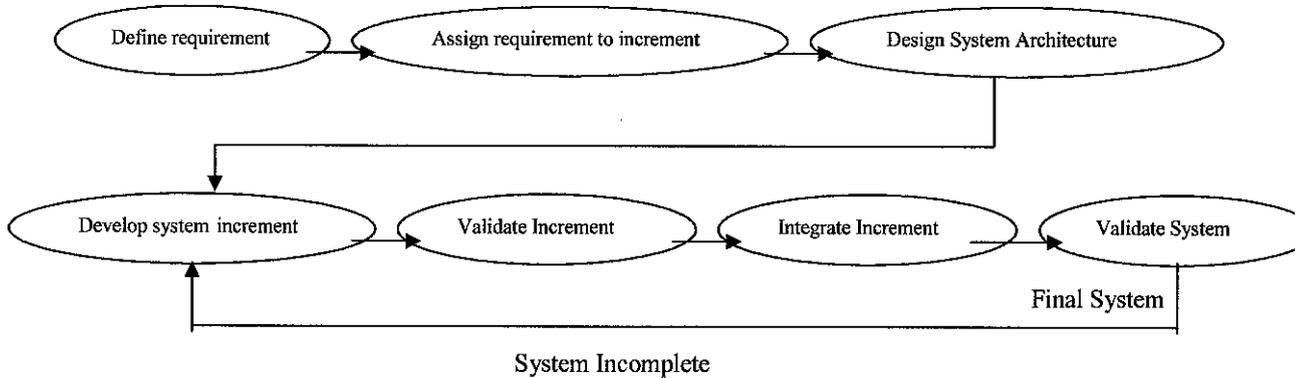


Figure 3.1 The Incremental Model

This is the overall diagram show the general process that involved in this type of model. The backward arrow occurs when there are incomplete problem during the implementation phase.

The author try to suit in all constrains in completing this study inside this model. The time, budget and capabilities or skills constrains be the three biggest obstacles that the author need to handle smartly in completing this project. Below the author will split the explanation into the research and implementation methodology that based on this main model that the author used.

3.2 RESEARCH METHODOLOGY

All the research the author done is in the incremental form. As the name implies, the authors knowledge about general networking, TCP/IP, general VoIP concept until the designing issues in VoIP is done one by one and the author get a lot from the entire materials read. This project implementation required a lot of reading, the understanding of the concept of network behind it and the access to online materials is very crucial.

The access freely to the printer also important to make sure the author could print all the white papers, vendor's manual papers and presentation that related to the research area. The method that the author used as the source of research includes:

3.2.1 Reference Book and Vendors Manual Study

- Software Development Book
 - *Software Engineering*, 6th Edition, by Ian Sommerville publish by Addison-Wesley
- General Networking Book Study
 - *TCP/IP using Graphic, An easy way*
- General VoIP system Book
 - *Voice & Data Internetworking*, by Gil Held, publish by Mc-Graw Hill ,1998
 - *Voice Over IP*, by Uyles Black, publish by Prentice Hall series in advanced Communication Technologies, 1999
 - *Voice over IP: system and solution*, by Richard Swale, publish by Institution of Electrical Engineers, London, 2001
- CISCO System Beginners Book
 - *CISCO A Beginner's Guide*, by Togy J Velte, publish by Osborne, 2002

- CISCO general network Configuration Manual
 - *Software Configuration Guide for 2600, 3600 and 3700 Series Routers*, from CISCO Systems.

- CISCO VoIP Manual
 - *Chapter 3,4,5 Configuring Voice over IP*, by CISCO Press
 - *CISCO Voice Gateway Router Interoperability with Cisco Call Manager*, by Cisco System.
 - *VoIP Configuration using 1750 Router*, publish by Cisco System.
 - *Cisco – Configuring Connection PLAR for VoIP Gateways*, publish by Cisco System, Dec 2003.
 - *Configuring Microsoft Netmeeting with Cisco IOS Gateways*, publish by Cisco System.
 - *Cisco Basic Two Zone Cisco Gateway to Gatekeeper Configuration*, publish by Cisco System.
 - *Cisco – VoIP with IVR*, Cisco System.
 - *Synopsis of Basic VoIP Concepts*, publish by Cisco System.

- CISCO Voice and Data Reference Book
 - *Cisco Packetized Voice & Data Integration*, by Robert Caputo, publish by Mc-Grew Hill,2000

- Openh323 Online Documentation
 - www.openh323.org

- GNU Gatekeeper Installation and Documentation
 - www.gnugk.org

- Various Linux Websites
 - www.yolinux.com

3.2.2 Research Journal and White Papers

- *Next Generation VoIP network Architecture*, by Paul Drew publish by multiservice switching forum, March 2003.
- *Converged Network Architecture Overview*, publish by Flextel.
- *Voice over IP – White Paper Revision 1.0*, publishes by Telco System A Batm Company, Jan 2002.
- *A Student-Lecturer Communication System through VoIP Technology*, by Shahidul Izzani, UTP, Nov 2003.]
- *The Successsful Deployment of VoIP*, by Dr.Jim Metzler, Ashton, Metzler & associates.
- *VoIP Howto*, by Roberto Arcomano, August 2002.
- *Do-It-Yourself VoIP*, by Gary Audin, July 2001.
- *Cisco vs. Shoreline: The Impact of VoIP Architecture on Management, Functionality and TCO*, by Kevin tolly, publish by Tolly Group.

3.3 IMPLEMENTATION METHODOLOGY

After collects the relevant information and enough theory from the research that had been made, the author required to transfer it to the real implementation. This is very important because there are no meanings if we only know but do not do it. All the implementation works that the author done in completing this project also prepared within the incremental process follow the main methodology philosophy that the author mention above.

In implementing this system there are step by step things that the author design and try it independently. There also specific problem that the author faced in implementing this system. But, the problem is not more different from constrains state above. Below is the method that being used in the implementation effort of this project.

3.3.1 Step-by-Step Development

This is the basic method that I apply I this project. The stuff that I implement includes:

- Identifying the Voice Port and its function.
 - The Foreign Exchange Station (FXS) port is configured with a standard RJ-11 connection port .The FXS port is used to connect the router to standard telephony devices and endpoint stations, such as basic telephone equipment, key sets, or FAX machines. The FXS port can supply ring voltage, dial tone and other basic signaling to an end station.[8]
 - The FXO port is also configured with a RJ-11 connection port. However, rather than supplying the signaling and voltage needed for basic telephony equipment, FXO ports are used to connect the IP network to off-premises equipment such as a PSTNs (Public Switched Telephone Networks) Central Office (CO) or to a PBX tie line interface. You can set several different parameters that are compatible with tie line features on a PBX.[8].
- Identify Voice Module and Type
 - VIC-2FXS VIC-2FXO
- Implement the POTS Dial Peer from the Call Legs
 - **POTS** (Plain Old Telephone System) - dial-peer represents an access port that is wired to a phone set or some similar telephony device locally attached to the router. This connection will interpret or “play out” all of the dialed digits from the sending entity and interpret them to see if they are destined for the dial-peers’ particular port.[8]
- Implement the VoIP Dial Peer from the Call Legs
 - A **VoIP dial-peer** represents a connection that will be routed to another voice enabled router on the network. In this case there is no

need to have the port interprets the dialed digits; this will be handled by the receiving entity at the other end of the VoIP connection. Therefore the VoIP dial-peer will simply pass all of the digits to the receiving entity.[8]

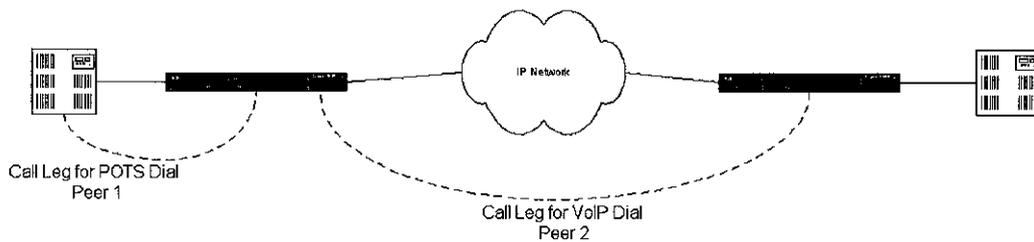


Figure 3.2 The Call Leg for VoIP and POTS

- Implement Dial-Plan for PC to Phone and Phone to PC
 - A dial plan is a template from which your company can implement the VoIP routing structure. Each routed area of the business will be assigned a set of phone numbers, along with an area code, and other shortcuts such as quick dialing features that allow calling parties to reach the calling area without dialing the entire number. Before a VoIP network can be implemented, all of the voice parameters, phone numbers, and dialing conveniences, need to be identified and planned out ahead of time. This will exponentially decrease the time needed for implementation and troubleshooting of the new VoIP network[8]

Router/Dial Peer Tag Number	Number Expansion Extension	Destination Pattern	Type of Peer	Voice Port	Session Target	CODEC	QoS Method
Router A							
1	1001	555....	POTS	1/0/0			
2	1002	555....	POTS	1/0/1			
3		556....	VoIP	Serial 0/0	IPV4:10.0.0.2	G.729	Best Effort
4	1003	555....	VoIP	Fast Ethernet	IPV4:10.10.10.10	G.729	Best Effort
5	1004	555....	VoIP	Fast	IPV4:10.10.10.11	G.729	Best

				Ethernet			Effort
6	1005	555....	VoIP	Fast Ethernet	IPV4:10.10.10.12	G.729	Best Effort
7	1006	555....	VoIP/Wi-Fi	Fast Ethernet	IPV4:10.10.10.13	G.729	Best Effort
Router B							
1	2001	556....	POTS	1/0/0			
2	2002	556....	POTS	1/0/1			
3		555....	VoIP	Serial 0/0	IPV4:10.0.0.1	G.729	Best Effort
4	2003	556....	VoIP	Fast Ethernet	IPV4:10.20.20.18	G.729	Best Effort
5	2004	556....	VoIP	Fast Ethernet	IPV4:10.20.20.19	G.729	Best Effort
6	2004	556....	VoIP	Fast Ethernet	IPV4:10.20.20.22	G.729	Best Effort

Table 3.1 The Dial-Plan for IP-POTS Phone

- Fine tune all the ports
 - I adjusted the ports to fine tune the its which will minimize some of the issues involved with delay and echo. In most cases, the default parameters for FXO/FXS ports will be sufficient, but special values can be set for the following parameters:[8]
 - Input gain
 - Output attenuation
 - Echo-cancel coverage
 - Nonlinear processing
 - Initial digit timeouts
 - Interdigit timeouts
 - Timing other than timeouts
- Connect to the PABX
 - This work involved the usage if local UTP PABX and connect it using FXO ports.
 - This work includes configuring the ports, dial plan and test it.

3.3.2 Try and Error Approach

I used this approach in development almost in all cases, but specially when there are different between the system requirement inside the manual from CISCO with the actual Lab equipments.

- Implement the Gatekeeper for Dedicated Zone
 - I designed and up the gatekeeper for the VoIP zone firstly using CISCO 3725, although that Router can't be the gatekeeper because there are no gatekeeper function in its IOS. So, I decide to use Open Source gatekeeper called "*Openh323 Gatekeeper and MCU*" and I also tested "*Dual Gatekeeper for Netmeeting*" as the brain of my VoIP networks.

- Connect to the Wireless Access Point
 - I also up the wireless environment with the 2 CISCO Aironet 350. So the Wireless client can access the network resource and can be called using the VoIP networks.

CHAPTER 4

RESULT AND DISCUSSION

4.1 THE VOIP SYSTEM: LABORATORY TESTING AND INFRASTRUCTURE

This is the initial parts that initiate the other parts of execution. In the Data Communication Lab on the second building, all the laboratory testing, research and development had been done. In this Laboratory also I discuss, brainstorm new idea and the problem that arises along the development phase with the lab technician.

There are 4 network islands that we setup in order to make the networking world inside the network alive. There is KL Island, Penang Island, Ipoh Island and JB Island. This is four islands that we setup separately with the different type of service each of them. This infrastructure is very important not only for my project but for all research student that make the development inside this Lab.

There is specific hardware that is dedicated to each island inside the Laboratory:

1. KL ISLAND

- One CISCO 3725 Router with 2 Serial, 2 Fast Ethernet, 1 VIC-FXO, 1 VIC-FXS module.
- One CISCO 3550 Switch 10 ports that consist of console ports and it is the layer 3 switch that also can work as Router.
- 4 Client PC that using Windows XP Professional connected to it.

- One Windows 2000 Server that connected to it. – as the VoIP and server.
- One Wireless Aironet 350 Bridge that connected to the Switch.
- The entire PC, server and Wireless Bridge is configured through the Router and get the IP from DHCP server.
- The Serial 0/0 is connected to the Ipoh Router.
- The FXS ports are configure as the POTS telephony.
- The FXO port is connected to the PSTN wall and access the PSTN user from outside.

2. IPOH ISLAND

- One CISCO 3725 Router with 2 Serial, 2 Fast Ethernet, 1 VIC-FXO, 1 VIC-FXS module.
- One CISCO Catalyst 2795 Switch 24 ports that consist of console ports.
- 4 Client PC that using Windows XP Professional connected to it.
- One Windows 2000 Server that connected to it. – As the VoIP and Multimedia server.
- The entire PC, server and Wireless Bridge is configured through the Router and get the IP from DHCP server from the router.
- The Serial 0/0 is connected to the KL Router.
- The FXS ports are configure as the POTS telephony.

3. JB ISLAND

- One CISCO 3725 Router with 2 Serial, 2 Fast Ethernet, 2 ATM Modules and 1 ISDN BRI module.
- One CISCO Catalyst 2795 Switch 24 ports that consist of console ports.
- 3 Client PC that using Windows XP Professional connected to it.
- One Windows 2000 Server that connected to it. – As the Video Encoding and streaming server.
- This island is connected to the KL Island using the Fast Ethernet 0/1.

4. PENANG ISLAND

- One CISCO 1751 Router with 1 Serial, 1 Fast Ethernet and 1 ISDN BRI module.
- One Intel Switch with 24 ports that consist of console ports.
- 3 Client PC that using Windows XP Professional connected to it. – Also be the cluster server for the Oracle 9i and as well as the client.
- There also one Linux Client that also configured as the DHCP server.
- One Windows 2000 Advanced Server that connected to it. – As the main Server for Oracle 9i Development.
- This island is connected to the KL Island using the Wireless connectivity using 350 Aironet Bridge.
- On this island there are research student that develop the Mobile learning on it.
- The entire PC, server and Wireless Bridge is configured through the Router and get the IP from DHCP server from the router.

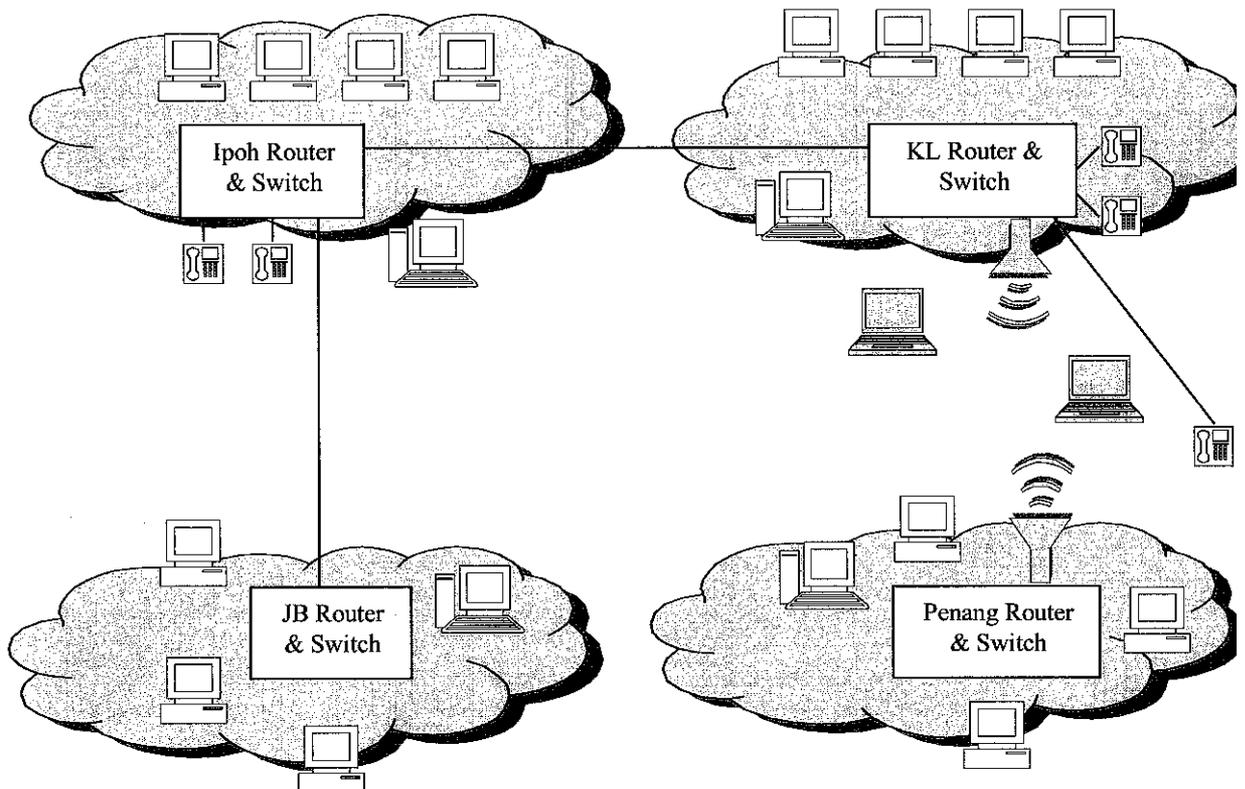


Figure 4.1 The Data Communication Lab Network Infrastructure

The Router KL and Ipoh configuration is attach In the **Appendix 4-1** and **Appendix 4-2**.

4.2 THE VOIP SYSTEM: PC TO PHONE, PHONE TO PC AND PABX

This is the code component of this project. The development works to get the Pc to Phone called, the Phone to PC called is the main and most challenging part. Firstly we need to differentiate between the laboratory testing and actual PABX that being used.

Here, I used the actual UTP internal PABX that located at the data center. To implement this integration actually need a lot of procedure, official letters and support letters from supervisor and Network Engineer UTP. This tedious procedure make this thing very hard to be integrated and I need to fid the most suitable time to make sure my development and testing not disturbing the office hour.

For the actual PABX connectivity, we are only need minimum one router to integrate it all. That main router is the router that connected to various networking protocol that we used I this project.

There are one study [9] tells that the large scale deployment of VoIP in the future is to take place completely independently of existing networks and services. Recognizing this problem, the Telecommunication and IP Harmonization over Networks (TIPHON) projects has been created with aim of addressing service level interworking between traditional Switched-circuit networks (SCNs) and the emerging next generation networks based on VoIP technology. This is just the discussion about the area that I working on.

The PABX connectivity is the medium to connect and integrate between the packet switching and the circuit switching networks. The Router is link to the PBX using the FXO port in the Voice Module. The designing steps and implementation work is generally like the POTS telephony system, but the Dial-Plan is more complex and the expansion number is link with the actual UTP telephone number.

Router/Dial Peer Tag Number	Number Expansion Extension	Destination Pattern	Type of Peer	Voice Port	Session Target	CODEC	QoS Method
Router A							
1	7412	372....	PSTN	1/1/0			
2	7413	372....	PSTN	1/1/0			
3	7414	372....	PSTN	1/1/0			
4	7415	372....	PSTN	1/1/0			
3		556....	VoIP	Serial 0/0	IPV4:10.0.0.2	G.729	Best Effort
4	1003	555....	VoIP	Fast Ethernet	IPV4:10.10.10.10	G.729	Best Effort
5	1004	555....	VoIP	Fast Ethernet	IPV4:10.10.10.11	G.729	Best Effort
6	1005	555....	VoIP	Fast Ethernet	IPV4:10.10.10.12	G.729	Best Effort
7	1006	555....	VoIP/Wi-Fi	Fast Ethernet	IPV4:10.10.10.13	G.729	Best Effort

Table 4.1 The Dial-Plan for IP-PABX Phone using One Router

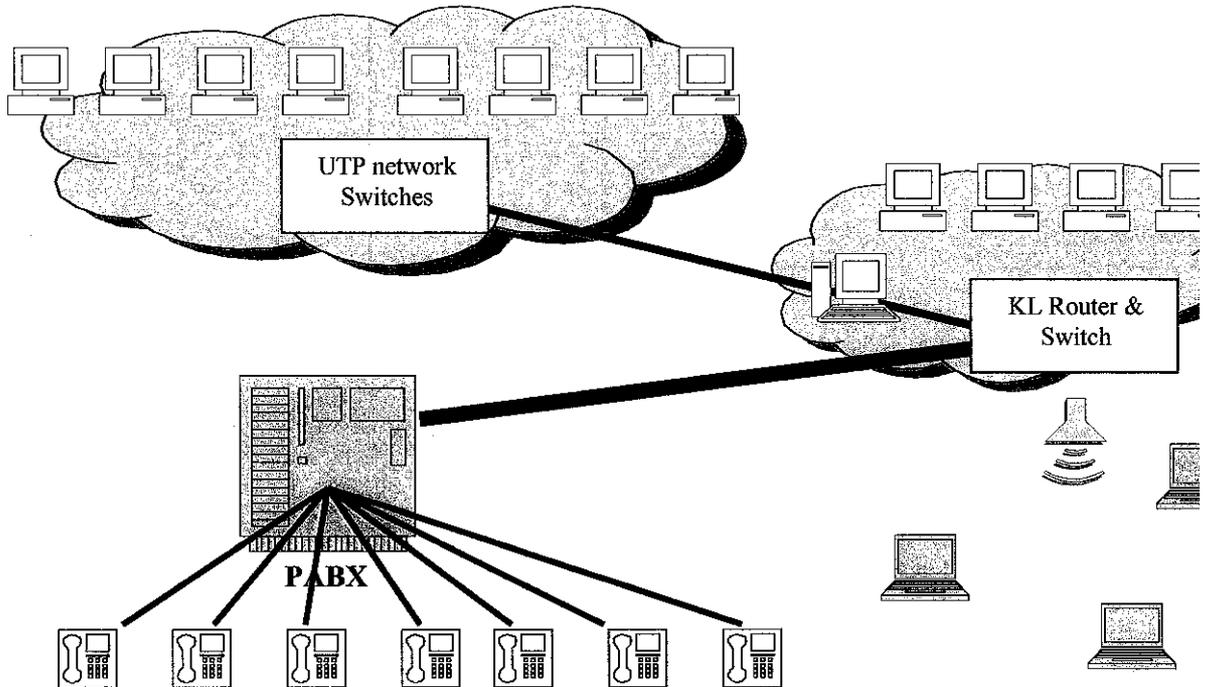


Figure 4.2 The Network Diagram for PC to PABX, PABX to PC and PDA to PABX

All of this implementation is include the following stage by stage design and implementation:

1. Fast Ethernet LAN setup.
2. Serial Interface configuration
3. DHCP setup.
4. Wireless LAN setup and integration.
5. 2 zone Router-to-router configuration.
6. PC to Phone call.
7. Phone to PC call.
8. QoS in the VoIP network.
9. FXO gateway to PSTN network
10. PLAR connection mode.
11. Dial-Plan Development.
12. PBX Connectivity and Integration using FXO.
13. E.164 Number translation.
14. Billing system Integration and Gatekeeper Installation.
15. Fine tune the Voice Port.

4.3 THE GNU GATEKEEPER

Communications need to be managed. The VoIP networks is also need to be well managed and all the operation that occur inside it need to be log for the particular purposes. That's why the author used GNU Gatekeeper to do this function.

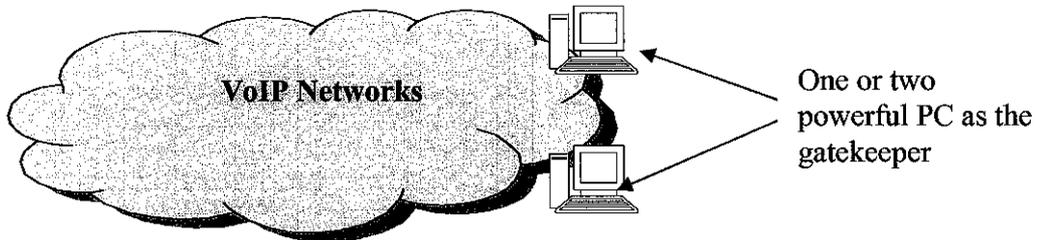


Figure 4.3 The Gatekeeper Position in the VoIP networks

The Gatekeeper is a required feature of the VoIP gateway system.[10] According to Recommendation H.323, a gatekeeper shall provide the following services[12]:

- Address Translation
- Admissions Control
- Bandwidth Control
- Zone Management
- Call Control Signaling
- Call Authorization
- Bandwidth Management
- Call Management

This is the complete features of this Openh323Gatekeeper. In this prototype development, the authors don't use this entire feature but select the most appropriate for the project constraint.

- RIP (Request In Progress) message is now understood by the gatekeeper.
- Fixed interoperability problem with some Cisco IOSes, because of copying nonStandardData field from RAS requests to RAS replies.
- Fixed vulnerability to an invalid destCallSignalAddress in Q.931 Setup messages.
- Fixed a critical bug with queueing proxied packets.
- A new ParseEmailAliases config option that allows parsing of email/DNS-like aliases for a destination address.

- Fixed LDAP support for Windows and Unix.
- Changed **addpasswd** utility invocation syntax.
- Fixed critical bug with internal handling of signalling addresses.
- Rewrite mechanism should work now also in direct signalling mode.
- New ScreenSourceAddress configuration option.
- MCUs are treated like gateways and allow to register with their prefixes.
- Improved GNU Gatekeeper Service Windows utility from Franz J Ehrenguber.
- Direct IP-IP (from an unregistered endpoint to an unregistered endpoints) calls are now possible.
- Permanent endpoints are correctly reloaded now.
- Generic SQL engine with MySQL and PostgreSQL support. New authentication modules introduced - [SQLPasswordAuth] and [SQLAliasAuth].
- Ability to read some configuration settings from a SQL database. The new config section [SQLConfig] introduced.
- New config variable **CheckSetupUnregisteredOnly** for [RadAliasAuth] module.
- Performance of the socket code improved (especially when LARGE_FDSET is enabled).
- Acct-Session-Id is now 16 characters long to guarantee uniqueness.
- New direct SQL accounting module ([SQLAcct]).
- Flexible FileAcct CDR file rotation.

The author used this Gatekeeper as the brain and the central of the VoIP networks. As mention in above gatekeeper function, the gatekeeper will route and control signaling between all endpoints that make the calls. The Gatekeper also will route the IP calls from PC to the gateway that register to its networks. The below is the simplify Diagram of the operation of the gatekeeper.

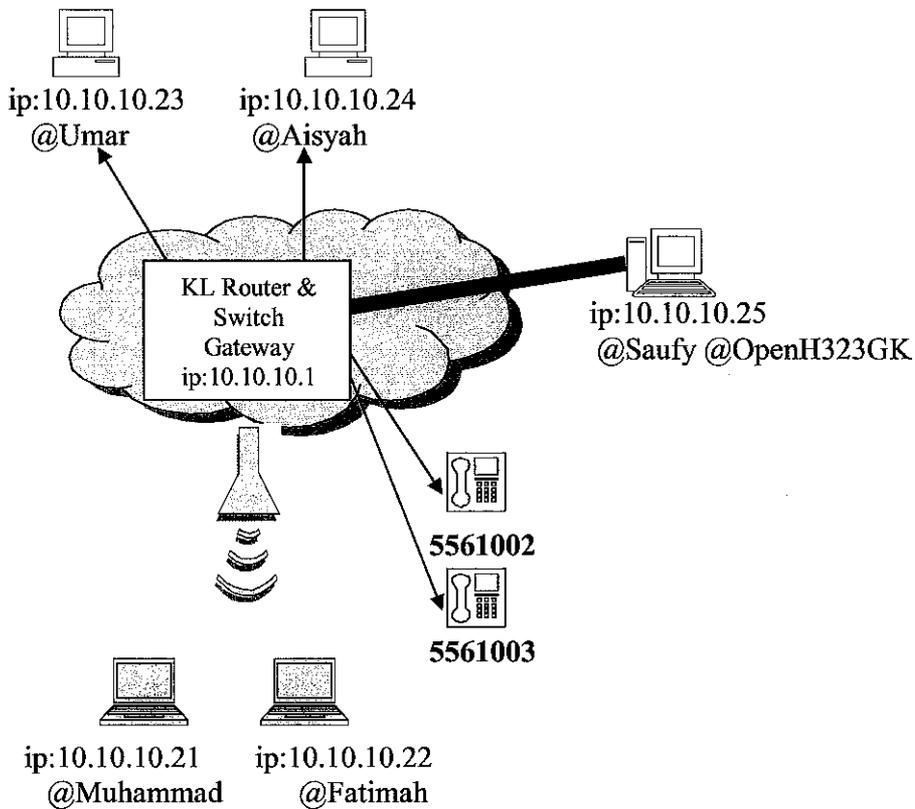


Figure 4.4 Diagram of the operation of the gatekeeper

The figure above show the diagram of the VoIP network with the Gatekeeper. The step by step operation that occur in this networks is:

1. The End point client registers with the Gatekeeper with the IP and alias name. eg: 10.10.10.21 alias Muhammad.
2. The Gatekepeer will store information all the user that registers to it.
3. The IP number of the user will be translate to the alias name. eg:10.10.10.22 to Fatimah. So, user just needs to enter Fatimah to call 10.10.10.22.
4. The Router Gateway 10.10.10.1 register with the Gatekeeper and the Gatekeeper store the E.164 endpoint (5561002 and 5561003) to the Gatekeeper.
5. The prefix of the gateway also store to the Gatekeeper database. Users just need to dial 1002 to call 5561002 and 1003 to call 5561003. It's because the prefix of the Gateway is store in the configuration file.
6. The Gatekeeper will monitor and can control all the endpoint, the call that been made and the bandwidth of the networks. Now, the Gatekeeper plays role as the PABX of the network.

The Complete services and configuration tools is being attach in **Appendix 4-3** and the Complete GNU Gatekeeper Command in attach in **Appendix 4-4**

4.4 THE QOS RESULT AND OTHER STUDIES

According to Toby J Velte (2001),

“QoS is a collection of run-time process that actively manage bandwidth to provide committed levels of network service to application and/or users. QoS implements a framework for service policy and action that extends end-to-end for serviced connections, even across autonomous system.”

According to Toby J Velte (2001),

“Packet are prioritized at the client and given enhanced treatment at the router”

Telephony is real-time and people do not want to use telephone that can't support the real-time communication. That's why the QoS is very important to make sure the voice packet is being prioritized in my VoIP implementation. There are a lot of configuration parameters and function can be used in order to give the 1st class services to the Voice traffic but we are not used to one particular line at all time and we also not used only voice in our IP networks.

There're people with video streaming, emailing, chatting and internet surfing that also used our limited bandwidth. Here, the smart and intelligent configuration of QoS is needed. But in this implementation, the most critical part that required the best QoS is in the Wireless link

I make the fine tune configuration on the Wireless link and adjust the individual ports that involved. In actual fact, QoS is very subjective and hard to measure but it can be split out to these parameters:[11]

- Bandwidth – the perceived width of the pipe
- Delay – the perceived length of the pipe
- Jitter – the perceived variation of the length
- Packet Loss – the perceived leak on the pipe

According to Shahidul (2003)

“There are study made by himself from the questionnaire from the UTP student shows that 80 % from the 16 responded says that had a problem in communication with lecturer. And the 95% statistic shows that this system is good for the future communication system”

This is other study reported by the author about the student feedback from the questionnaire. The statistic shows that this new communication line is very useful to the student as well as the employees.

CHAPTER 5

CONCLUSION AND RECOMMENDATION

5.1 SYSTEM REVIEW

This project is intended to implement the next generation networks in UTP campus environment that integrates the traditional PSTN telephone services with the IP networks.

This is the new implementation inside our UTP campus but in the modern country, this technology is commonly used. The implementation of this project is first start on the fundamentals theory about computer networks, TCP/IP and CISCO product. This system is design to be the completed deployed system that can be used through out our campus. From the lecturer and staff side that used PSTN telephone, it can call the student or other staff PC that link with the UTP computer networks. As I mentioned in previous chapter, this system is involved of usage of PABX and CISCO router as the important 'middle man'.

5.2 RECOMMENDATION / FUTURE ENHANCEMENT

Even this project is successfully completed; there is a lot of space for improvement and future enhancement that could be done on it. The communication world is developing, the technology is change, and the threat to the technology is also change. As this study is also the research study about the VoIP, the research can be continued by someone after me. Maybe for the future, another study need to look at another perspective of implementing this system, maybe from the security aspect, from the performance analysis view and from the usability aspect. There are a few things that can be enhanced in the future:

1. The more advanced communication server is needed.
2. Integrate the Architecture for Voice, Video and Integrated Data (AVVID) in the implementation.
3. Full integration with the Wireless Hotspot.
4. The extra internal UTPGSM network that can be implement as the complement to be the Next Generation Networks.
5. Interactive Voice Response (IVR) with VoiceXML through VoIP.
6. Research on the possible attack on VoIP networks and prevention steps.
7. Integrate with Advanced protocol of VoIP – Session Initial Protocol (SIP).

5.2.1 The more complete and Interactive billing system

This is the first point that needs to be stress because the smart telecommunication needs to have smart brain. This is very important because the server is the center of the network operation. As the Novell server in UTP that track down the entire student ID login and file server, this VoIP server also track down all the endpoints, the gateway usage and also the bandwidth usage of all network resources. More intelligent the server, more reliable the system.

5.2.2 Integrate the Architecture for Voice, Video and Integrated Data (AVVID) in the implementation.

The Architecture for Voice, Video and Integrated Data (AVVID) is the advanced VoIP system that support by CISCO and other hardware. This is the on of the advanced solution for the integrated next generation networks. This system integrates the VoIP with the CISCO call manager and AS5300 CISCO Gateway. There are Video Conferencing, Multi User Chatting and Discussion and a lot more features that provide under it. There are also advanced management tool inside the AVVID implementation.

5.2.4 Full integration with the Wireless Hotspot.

Although this implementation includes the one Wireless 350 Aironet Bridge, this integration could be enhanced in the future. Maybe we need the login for the hotspot so the user is triggered about the wireless signal he or she receives. For the future, we also can setup a few repeaters to repeat and strengthen the signal. If the repeater is setup, the distance of wireless devices is longer and the signal is stronger.

5.2.4 The extra internal UTPGSM network that can be implement as the complement to be the Next Generation Networks.

This is just a suggestion to implement one of the future networks dreams by the telecommunication researcher.

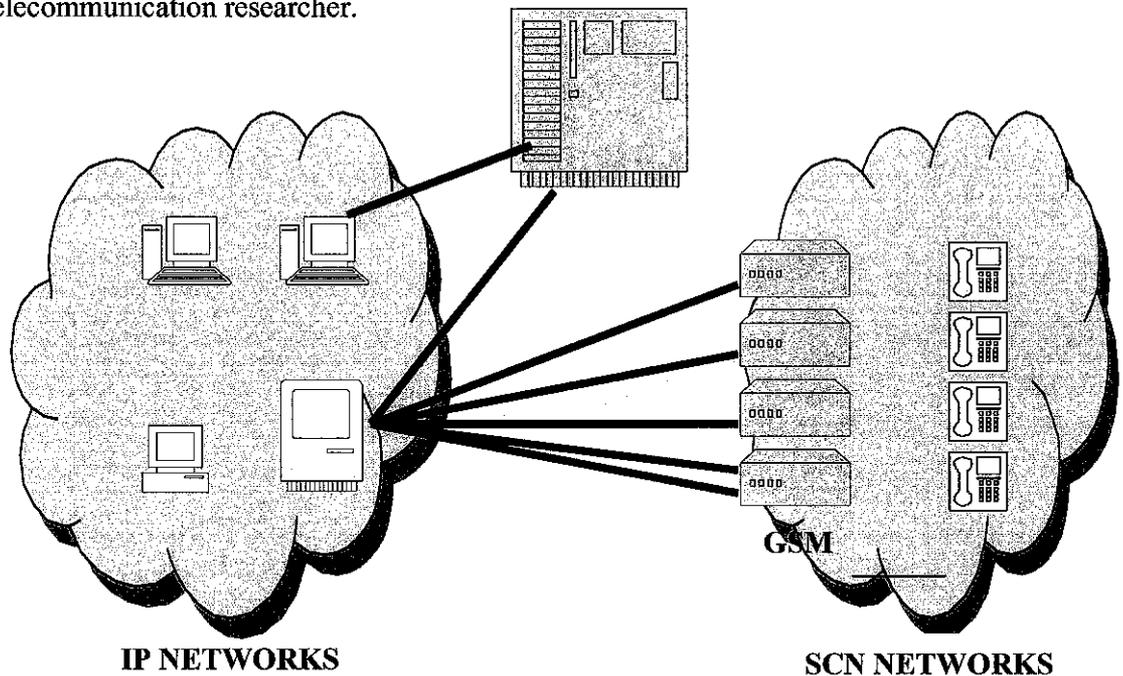


Figure 5.1 The GSM Network Integration

This is the model of the TIPHON [9], is the integration idea of all the communication medium and channel. This principle goal is to enable users connected to IP Based networks to communicate between themselves and also with the users in SCN (Switching Circuit Networks), especially those served by PTSN, ISDN or GSM Networks. From this philosophy that I hope someone could make the prototype of it or at least make the research on it.

5.2.5 Interactive Voice Response (IVR) with VoiceXML through VoIP.

This is also the good idea to enhance the usage of the VoIP network. With the IVR, the user can access the database from the server he or she called to, and the database is converted to the Voice using the Voice XML programming. This is the area that student need to venture in the future due to the emerging of the voice technology.

5.2.6 Research on the possible attack on VoIP networks and prevention steps.

No live without security, so as well with the human interaction, the computer and communication world also very particular with the security. With the world that not safe right now, there is a lot of possible attack to your networks .Even the multi-billion company network with high-end system also can be hacked. That's why here people need to think the security feature that need to be embedded in the VoIP communication line to prevent some wiretapping, packet that being view and other possible attack to our networks.

5.2.7 Integrate with Advanced protocol of VoIP – Session Initial Protocol (SIP).

There also new protocol that come forward right nowadays, called Session Initial Protocol (SIP). It is the new protocol that a slightly different from H.323 that being used as a standard to the VoIP system. There are a few open source SIP server system for free in the net that we can configure it, run it and deploy it by ourselves. It called VOVIDA. The SIP is the VoIP package with the several of servers inside it, including recording server, IVR server, redirect server and Accounting server that offer more powerful and better services compared to H.323 protocol stack. CISCO also provides the infrastructure to deploy SIP system but it may require extra hardware with the expensive cost.

5.3 PROBLEM FACES

If you live without problem, it is actually the main problem. Like me, there are a few problems that I faced In order to complete this study. These problems I encountered along the way in the research phased and also during the development phase. The problem is including:

1. The Variety of product with the different capabilities
2. The Limited of Time

5.3.1 The Variety of product with the different capabilities

CISCO is the leading computer networks and Telecommunication Company. But I faced the problem to identify the different type of module in the CISCO router. There are 1751 Manual configuration for VoIP, 3700 series configuration for VoIP, VoIP configuration with 5500 Gateway Router and also VoIP gatekeeper with 2600. The variety of VoIP Configuration manual using different type of Gateway Router makes me confuse the actual capabilities of the router that I used in the Lab (CISCO 3725). And there is no specific VoIP configuration using Gateway Router 3725. The numbering scheme also different according to the router. That's why even though I follow step by step to the configuration manual; the output that expected didn't appear.

5.3.2 The Limited of Time

With the 16 credit hour including this final year project, I need to distribute my time, energy to another 4 coursework. There is also another project from other subject as well. So, I cannot give all my 'bandwidth' to this Lab experiment and research. The smart distribution is required in order to me finishing the entire task intelligently.

5.4 CONCLUSION

Generally, the implementation of this VoIP does help to provide a solid communication channel to all UTP citizens. It will burst the relationship between the student and lecturer, as well as improved the academic performance. There are a lot of thing that I learn, I feel and I touch though out the project. The valuable hands-on experience in the laboratory the most priceless skills I acquire. I hope I will make the further research on this area and I hope I could pursue my postgraduate study on the deeper topic on the channel coding field in telecommunications.

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Appendices

Appendix 4-1 The KL Router Configuration

```
KL#show running-config
Building configuration...

Current configuration : 3347 bytes
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname KL
!
logging queue-limit 100
enable secret 5 $1$0H./$tBBj9Tywb1q5wj40m52OH1
enable password 7 09797A39
!
ip subnet-zero
!
!
no ip domain lookup
no ip dhcp conflict logging
ip dhcp excluded-address 10.10.10.0 10.10.10.10
ip dhcp excluded-address 10.20.20.0 10.20.20.10
ip dhcp excluded-address 10.10.10.0 10.10.10.20
ip dhcp ping timeout 200
!
ip dhcp pool DataCom
    network 10.10.10.0 255.255.255.0
    domain-name utp.test
    default-router 10.10.10.1
!
ip dhcp pool DataComm2
    network 10.20.20.0 255.255.255.0
    domain-name utp.com
    default-router 10.20.20.1
!
ip dhcp pool Datacomm
    lease 100
!
ip accounting-list 0.0.0.1 255.255.255.0
appletalk routing
ipx routing 000d.28d4.0d60
mpls ldp logging neighbor-changes
!
!
!
voice service voip
```

```

h323
h245 tunnel disable
h245 caps mode restricted
!
!
!
!
!
no voice hpi capture buffer
no voice hpi capture destination
!
!
mta receive maximum-recipients 0
!
!
!
!
interface Port-channel1
no ip address
hold-queue 300 in
!
interface FastEthernet0/0
ip address 10.10.10.1 255.255.255.0
no ip mroute-cache
speed 100
full-duplex
no mop enabled
h323-gateway voip interface
h323-gateway voip h323-id gw1
!
interface Serial0/0
ip address 10.0.0.1 255.255.255.0
no ip mroute-cache
no fair-queue
!
interface FastEthernet0/1
ip address 160.0.57.235 255.255.255.0
no ip mroute-cache
speed auto
full-duplex
!
interface Serial0/1
no ip address
no ip mroute-cache
shutdown
clockrate 2000000
fair-queue 64 256 4
ip rsvp bandwidth 100 32
!
interface Dialer0
no ip address
!
router ospf 1
log-adjacency-changes

```

```

!
ip http server
ip classless
ip route 10.10.10.0 255.255.255.0 172.16.0.0
ip route 10.10.10.0 255.255.255.0 160.0.57.0
ip route 10.10.10.0 255.255.255.0 160.0.59.254
!
!
!
dialer-list 1 protocol ip permit
dialer-list 1 protocol ipx permit
!
!
snmp-server community public RO
snmp-server contact location kg_bali
snmp-server system-shutdown
snmp-server enable traps tty
!
!

!
tftp-server flash:
tftp-server flash:c3725-js-mz.122-15.T2.bin
!
call rsvp-sync
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/1/0
!
voice-port 1/1/1
!
mgcp
!
mgcp profile default
!
!
!
dial-peer cor custom
!
!
!
dial-peer voice 3 voip
destination-pattern 555....
session target ipv4:10.0.0.2
!
dial-peer voice 2 pots
destination-pattern 5561002
port 1/0/1
!
dial-peer voice 1 pots
destination-pattern 5561003
port 1/0/0
!

```

```
dial-peer voice 26 voip
destination-pattern 1026
session target ipv4:10.10.10.26
codec g711ulaw
!
dial-peer voice 27 voip
destination-pattern 1027
session target ipv4:10.10.10.27
codec g711ulaw
!
dial-peer voice 36 voip
destination-pattern 1036
session target ipv4:10.10.10.36
codec g711ulaw
!
dial-peer voice 1029 voip
destination-pattern 1029
session target ipv4:10.10.10.29
codec g711ulaw
!
dial-peer voice 28 voip
destination-pattern 1028
session target ipv4:10.10.10.28
codec g711ulaw
!
dial-peer voice 29 voip
destination-pattern 1029
session target ipv4:10.10.10.29
codec g711ulaw
!
gateway
timer receive-rtcp 5
emulate cisco h323 bandwidth
!
!
telephony-service
voicemail 123
!
banner motd ^C
welcome
^C
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
password 7 071A355C
login
!
end
```

KL#

Appendices

Appendix 4-2 The IPOH Router Configuration

```
IPOH#sh
IPOH#show
*Mar  4 17:24:38.805: %SYS-5-CONFIG_I: Configured from console by
consolerun
IPOH#show running-config
Building configuration...

Current configuration : 3349 bytes
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname IPOH
!
logging queue-limit 100
enable secret 5 $1$0H./$tBBj9Tywb1q5wj40m52OH1
enable password 7 09797A39
!
ip subnet-zero
!
!
no ip domain lookup
no ip dhcp conflict logging
ip dhcp excluded-address 10.10.10.0 10.10.10.10
ip dhcp excluded-address 10.20.20.0 10.20.20.10
ip dhcp excluded-address 10.10.10.0 10.10.10.20
ip dhcp ping timeout 200
!
ip dhcp pool DataCom
    network 10.10.10.0 255.255.255.0
    domain-name utp.test
    default-router 10.10.10.1
!
ip dhcp pool DataComm2
    network 10.20.20.0 255.255.255.0
    domain-name utp.com
    default-router 10.20.20.1
!
ip dhcp pool Datacomm
    lease 100
!
ip accounting-list 0.0.0.1 255.255.255.0
appletalk routing
ipx routing 000d.28d4.0d60
mpls ldp logging neighbor-changes
```

```

!
!
!
voice service voip
  h323
    h245 tunnel disable
    h245 caps mode restricted
!
!
!
!
!
!
!
no voice hpi capture buffer
no voice hpi capture destination
!
!
mta receive maximum-recipients 0
!
!
!
!
interface Port-channel1
  no ip address
  hold-queue 300 in
!
interface FastEthernet0/0
  ip address 10.10.10.1 255.255.255.0
  no ip mroute-cache
  speed 100
  full-duplex
  no mop enabled
  h323-gateway voip interface
  h323-gateway voip h323-id gw1
!
interface Serial0/0
  ip address 10.0.0.1 255.255.255.0
  no ip mroute-cache
  no fair-queue
!
interface FastEthernet0/1
  ip address 160.0.57.235 255.255.255.0
  no ip mroute-cache
  speed auto
  full-duplex
!
interface Serial0/1
  no ip address
  no ip mroute-cache
  shutdown
  clockrate 2000000
  fair-queue 64 256 4
  ip rsvp bandwidth 100 32
!
interface Dialer0
  no ip address

```

```

!
router ospf 1
  log-adjacency-changes
!
ip http server
ip classless
ip route 10.10.10.0 255.255.255.0 172.16.0.0
ip route 10.10.10.0 255.255.255.0 160.0.57.0
ip route 10.10.10.0 255.255.255.0 160.0.59.254
!
!
!
dialer-list 1 protocol ip permit
dialer-list 1 protocol ipx permit
!
!
snmp-server community public RO

snmp-server contact location kg_bali
snmp-server system-shutdown
snmp-server enable traps tty
!
!
!
tftp-server flash:
tftp-server flash:c3725-js-mz.122-15.T2.bin
!
call rsvp-sync
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/1/0
!

voice-port 1/1/1
!
mgcp
!
mgcp profile default
!
!
!
dial-peer cor custom
!
!
!
dial-peer voice 3 voip
  destination-pattern 555....
  session target ipv4:10.0.0.2
!
dial-peer voice 2 pots
  destination-pattern 5561002
  port 1/0/1
!
!

```

```

dial-peer voice 1 pots
 destination-pattern 5561003
 port 1/0/0
!
dial-peer voice 26 voip
 destination-pattern 1026
 session target ipv4:10.10.10.26
 codec g711ulaw
!
dial-peer voice 27 voip
 destination-pattern 1027
 session target ipv4:10.10.10.27
 codec g711ulaw
!
dial-peer voice 36 voip
 destination-pattern 1036
 session target ipv4:10.10.10.36
 codec g711ulaw
!
dial-peer voice 1029 voip
 destination-pattern 1029
 session target ipv4:10.10.10.29
 codec g711ulaw
!
dial-peer voice 28 voip
 destination-pattern 1028
 session target ipv4:10.10.10.28
 codec g711ulaw
!
dial-peer voice 29 voip
 destination-pattern 1029
 session target ipv4:10.10.10.29
 codec g711ulaw
!
gateway
 timer receive-rtcp 5
 emulate cisco h323 bandwidth
!
!
telephony-service
 voicemail 123
!
banner motd ^C
welcome
^C
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 password 7 071A355C
 login
!
end

IPOH#

```

Appendices

Appendix 4-3 GNU Gatekeeper Monitoring Output

```
[root@gatekeeper root]# telnet 160.0.57.224 7000
Trying 160.0.57.224...
Connected to 160.0.57.224.
Escape character is '^]'.
Version:
Gatekeeper(GNU) Version(2.0.7)
Ext(pthread=1,acct=0,radius=1,mysql=0,ldap=0) Build(May 28 2004,
16:41:06) Sys(Linux i686 2.4.20-8)

GkStatus: Version(1.0) Ext()
Toolkit: Version(1.0) Ext(basic)
Startup: Fri, 04 Jun 2004 10:42:34 +0800   Running: 0 days 00:07:34
;
rc
AllCached
Number of Endpoints: 0
;
??
AllRegistrations
RCF|160.0.57.224:1720|saufy:h323_ID|terminal|2385_endp
Fri, 04 Jun 2004 10:43:51 +0800 (permanent) C(0/0/0) <1>
Number of Endpoints: 1
;
r
AllRegistrations
RCF|160.0.57.224:1720|saufy:h323_ID|terminal|2385_endp
Number of Endpoints: 1
;
s
-- Endpoint Statistics --
Total Endpoints: 1 Terminals: 1 Gateways: 0 NATed: 0
Cached Endpoints: 0 Terminals: 0 Gateways: 0
-- Call Statistics --
Current Calls: 0 Active: 0 From Neighbor: 0 From Parent: 0
Total Calls: 0 Successful: 0 From Neighbor: 0 From Parent: 0
Startup: Fri, 04 Jun 2004 10:42:34 +0800   Running: 0 days 00:07:40
;
Find Saufy
SoftPBX: endpoint Saufy not found!
Find saufy
RCF|160.0.57.224:1720|saufy:h323_ID|terminal|2385_endp
;
??
AllRegistrations
RCF|160.0.57.224:1720|saufy:h323_ID|terminal|2385_endp
Fri, 04 Jun 2004 10:43:51 +0800 (permanent) C(0/0/0) <1>
Number of Endpoints: 1
;
s
-- Endpoint Statistics --
```

```
Total Endpoints: 1 Terminals: 1 Gateways: 0 NATed: 0
Cached Endpoints: 0 Terminals: 0 Gateways: 0
-- Call Statistics --
Current Calls: 0 Active: 0 From Neighbor: 0 From Parent: 0
Total Calls: 0 Successful: 0 From Neighbor: 0 From Parent: 0
Startup: Fri, 04 Jun 2004 10:42:34 +0800 Running: 0 days 00:08:10
;
r
AllRegistrations
RCF|160.0.57.224:1720|saufy:h323_ID|terminal|2385_endp
Number of Endpoints: 1
;
```

Appendices

Appendix 4-4 Complete GNU Gatekeeper Help Command

```
[root@gatekeeper root]# telnet 160.0.57.224 7000
Trying 160.0.57.224...
Connected to 160.0.57.224.
Escape character is '^]'.
Version:
Gatekeeper(GNU) Version(2.0.7)
Ext(pthreads=1,acct=0,radius=1,mysql=0,ldap=0) Bu
ild(May 28 2004, 16:41:06) Sys(Linux i686 2.4.20-8)

GkStatus: Version(1.0) Ext()
Toolkit: Version(1.0) Ext(basic)
Startup: Fri, 04 Jun 2004 10:42:34 +0800   Running: 0 days 00:11:31
;
h
Commands:
gk
clearcalls
cv
makecall
yell
disconnectip
disconnectcall
disconnectalias
disconnectendpoint
disconnectsession
who
!
quit
!!
debug
findverbose
RouteToAlias
exit
shutdown
?
printallregistrations
printallregistrationsverbose
printallcached
statistics
reload
help
find
unregisterallendpoints
unregisteralias
unregisterip
transfercall
version
```

rv
c
printcurrentcalls
printcurrentcallsverbose
f
RouteReject
h
fv
rc
??
q
r
s
v
rta
;