Implementation of Voice over Internet Protocol (VoIP) in UUM Campus

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Implementation of Voice over Internet Protocol (VoIP) in UUM Campus

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ABSTRACT

Currently, users (student, staff and lecturer) at UUM still using traditional mechanism to contact with each other using phone call and messenger contact. In this case, the uses need to have an Internet access in order to communicate with the other side. In both cases, using phone call or Internet has some limitations in terms of cost. This paper proposes the Voice over Internet Protocol (VoIP) system that can help users at UUM campus to freely communicate by using this VoIP technique. In the other side, this proposed system also helps to increase the effectiveness of using the Internet bandwidth; since the users can communicate with each other without the need to have an Internet access. Instead, they can contact with each other using the current Local Area Network (LAN) at UUM. Thus, this system can let the users to contact the destination user anywhere anytime in the coverage area of UUM.

Acknowledgments

In the Name of Allah, the Most Gracious and Most Merciful

All Praises to Allah for his guidance and blessing for giving me the strength and perseverance to complete this study. I would foremost like to thank my beloved family, for providing me with the opportunity to pursue my goals and for their love and affection, which has helped me along my study stages and through all my life. I would like to express my deepest gratitude to my supervisors Mr. Adib M.Monzer Habbal and Mr. Khuzairi bin Mohd Zaini for them guidance, instructions, and his advices that have enabled me to complete my project properly. Last my thanks would go to my all friends and classmates who gave me their help and shared with me their knowledge.

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TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

LIST OF APPREVIATION

VoIP	Voice over IP
LAN	Local Area Network
WLA	Wireless Local Area Network
PSTN	Public Switched Telephone Network
IP	Internet Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
RTP	Real Time Transport Protocol
RTCP	Real Time Transport Control Protocol
IP PBXs	IP Private Branch Exchange
ACD	Automatic Call Distribution
IVR	Interactive Voice Response
SIP	Session Initiation Protocol
PBX	Private Branch Exchange
MGCP	Media Gateway Control Protocol
ITU	International Telecommunication Union

CHAPTER ONE

INTRODUCTION

1.0. Background

Nowadays, in light of current developments in the field of communication networks and due to the urgent need to send data in the shortest time and the lowest possible cost, the VoIP technology has been found. According to (Zhang, Hillenbrand, Müller, 2005), VoIP is a technology to transport voice communication over IP network such as Internet. This technology provides the capability of making phone calls over the packet switched networks instead of traditional circuit switched networks. With the development of the Internet, VoIP over a Wireless Local Area Network (WLAN) is expected to become an important Internet application. In addition, this technology is becoming a popular service on the Internet platform and many VoIP protocols have been proposed since it was first developed, (Wang, Liew & Fellow, 2005).

VoIP can greatly reduce the telephone call costs comparing with the traditional Public Switched Telephone Network (PSTN) system, due to the low cost features of the Internet usage, so it is expected that the VoIP may completely replace the circuit switched PSTN system in the future. Therefore, VoIP is attracting more users from the traditional telephone communication area, and more companies try to invest in the development and usage of the VoIP systems (Zhang, Hillenbrand, Müller, 2005). Moreover, VoIP is one of the fastest growing Internet applications at this time. It has two fundamental advantages compared with voice over traditional telephone networks. First, by using advanced voice-compression techniques and bandwidth sharing in packet-switched networks, VoIP can significantly improve bandwidth efficiency. Second, it is easy to great new services which contain voice communication with other media and data applications such as video, white boarding, and file sharing. At the same time, by the needing for mobile access, the (WLAN) spread at the world fast, that because it is convenience, mobility, and high speed access. WLAN now is the most important department to get connection with Internet (Wang, Liew & Fellow, 2005). Thus, we are proposing to build a VoIP system that can provide better way of communicating between users at UUM. At the same time, this system can let the users contact the destination user anywhere anytime in the coverage area UUM with low cost comparing to traditional call or call using mobile.

1.1. Problem Statement

Previously, by using the traditional tools of communicating between the users at UUM such as phone call, messenger and face to face contact, users face many difficulties to communicate with each other. For example, if the student wants to contact the lecturer using the phone call he will face many issues. Firstly, the lecturer may not exist in his/her room when the student wants to contact the lecturer using his/her room number. Secondly, the cost of calling using the phone call as a mean of communication between the student and the lecturer is not neglected and it should be taken into account (Kelly, 2005). Moreover, using messenger is not an effective way of communicating between the users since the users need to have an Internet access in order to communicate with each other. In this case, the cost of using the Internet has some limitations in terms of consuming the bandwidth and the number of Internet's clients and this will lead to slow down the speed of messages 'transmission using messenger (Desantis, 2008).

1.2. Research Questions

Based on the problem statement mentioned before, the following questions are constructed:

i. How can we use UUM network to improve the current communicating services?
ii. How can we evaluate the proposed VoIP system?

1.3. Research Objectives

The main objective of this study to enhance the current communicating techniques at UUM by implementing and significant VoIP system based on VoIP technique. From this main objective, some sub objective can be formulated:

i. To design and implement VoIP system.
ii. To evaluate the proposed VoIP system among the users at UUM.

1.4. Significance Of The Study

The proposed system provides a significant and flexible way of communicating between the users at UUM. Here are some beneficial achievements that the proposed system provides:

i. The system helps the university to build strong relationship between their users (UUM Resident i.e. student, lecturer, and staff).
ii. The system provides faster and more convenient services to UUM students.

iii. The system helps to reduce consuming of bandwidth since the users can communicate with each other without the need of having an Internet access.

iv. In order to make connection between the users, they do not need to be existed in a specific place.

1.5. Scope And Limitation Of The Study

This proposed study concentrates on UUM to be as a domain. In the meanwhile, users at UUM have always been able to contact each other via messenger, phone call or face to face contact. That is still true, but the question is: how effective is the use of these tools? In fact, based on the problems that we just mention previously, we are proposing to build a VoIP system that can be used to improve the current traditional communication tools between the users at UUM. Furthermore, in this study we are going to depend on one of the most popular open source software called Asterisk. According to Asterisk company website (www.asterisk.com) Asterisk software has been defined as, the world's most popular open source communications project, is free, open source software that converts an ordinary computer into a feature-rich voice communications server. Asterisk makes it simple to create and deploy a wide range of telephony applications and services, including (IP Private branch exchange) IP PBXs, VoIP gateways, call center (Automatic Call Distribution) ACDs and (Interactive Voice Response) IVR systems. Thus, in this proposed system the VoIP technique will be used to provide a consistent and high quality communicating services to the users at UUM.

1.6. Organization Of The Report

This report consists of six chapters which covers discussing, implementing and designing VoIP system to be used at UUM campus. Here is an overview of the content of each presented chapter:

Chapter one: this chapter introduces the problem, gives an overview about the study and describes the needs of VoIP systems in the educational sectors. This chapter also discusses the scope of the study, the significance of the study and its objectives.

Chapter two: this chapter covers the literature review which is the previous related works that been done before. Moreover, this chapter represents relevant information for understanding the study more.

Chapter three: this chapter explains the details of the selected methodology that we have used in the project.

Chapter four: this chapter discusses about the design and implementation of the proposed VoIP system and how can we apply this kind of systems in UUM campus.

Chapter five: this chapter demonstrates the evaluation process of the proposed VoIP system.

Chapter six: this chapter presents the conclusion, recommendations and future works to improve this study.

CHAPTER TWO

LITERATURE REVIEW

2.0. Introduction

This chapter discusses some researches at VoIP technology that had been done by researchers. Moreover, this chapter presents definitions of VoIP system and why it is important for users at UUM and how users can communicate with each other using the network techniques that is already exist at UUM campus.

2.1. Definition of VoIP System

VoIP (also known as IP telephony) is the transport of voice traffic by using the Internet Protocol (IP), rather than the (PSTN). Compared to traditional circuit-switching technology, VoIP is an attractive choice for voice transport for many reasons such as low cost, and control with quality of voice (Gokhale & Jijun, 2005). VoIP is becoming as an alternative to PSTN scheme. IP telephone services providers are moving quickly from low-scale toll go around deployments to large-scale competitive carrier deployments.

2.2. Background Of VoIP Technology

Telecommunication becomes one of the most important needs that enable people all over the world to communicate with each other in their daily life using the communications technologies, networks and tools. One of the newest technology which plays an effective role in the telecommunication filed is the VoIP technology.

VoIP technology started in February of 1995 by using Internet Phone. This technology allows one user to call another via their computers, microphone and a set of speakers. Additionally, this product only works if both the caller and the receiver have the same software setup. By 1998, some entrepreneurs started to market PC-to-phone and phone-to-phone VoIP solutions. The phone calls were promoted as “Free” nation-wide long distance calls. Moreover, when the caller would start the call he/she had to listen to advertisements before the call was joined. Another development in 1998 was the hardware’s foray into the market. There were three IP Switch manufactures that presented VoIP switching software as a standard in their routing tools. By the end of 1998 VoIP calls had yet to total 1% of all voice calls. By 2000, VoIP calls accounted for 3% and by 2003 that number had jumped up to 25% (Hallock, 2004).

2.3. Benefits Of VoIP Systems

Nowadays, VoIP system becomes spread at every place in our life since VoIP produces a lot of benefits. These benefits develop the communication tools with low cost. Moreover, using VoIP system at local area network (LAN) helps to start the connection between users at this network and allows communicating directly even when the users are not at same place. In addition, building VoIP system in a way that can work locally helps to use the current equipment of the network and provides the connection between all users without needing to have an Internet access.

VoIP technology increasingly attracts attention and interest in the industry. A number of VoIP application offer all the features available (e.g., caller ID, call waiting) on traditional (private branch exchange) PBX solution and it is easy to disable any non-needing features. A number of companies are moving to VoIP technology, because it permits the use of their current network infrastructure for transferring both voice and regular data traffic. This technology enables a lot of savings by reducing maintenance cost, long distance fees, and other cost associated with a traditional telephone network (Ahson & Ilyas, 2009).

2.4. VoIP Protocols

In general, VoIP technologies employ a suite of protocols including signaling protocols such as (H.323), session initiation protocol (SIP), data control and transfer protocol such as transmission control protocol (TCP), real time transport protocol (RTP), user datagram protocol (UDP), Internet protocol (IP) and Media Gateway Control Protocol (MGCP). VoIP applications such as IP telephony system involves sending voice transmissions as data packets over private or public IP networks as well as reassembling and decoding on the receiving side. However, concerning for the security is always the major barrier that prevents many businesses from employing VoIP technologies (Ahson & Ilyas, 2009).

2.4.1. H.323 Overview

H.323 is International Telecommunication Union (ITU) specification describes the whole architecture and setups of audio and video communications through packetized networks. Since H.323 was released for the first time in 1996, it has been updated with many improvements and the last one was released in 2006, commonly referred to as H.323v6 (Patrick, 2009).

2.4.1.1. Basic Call Flow

The method of implementing the steps varies depending on the service architecture, type of service, and call scenarios. One of the typical call flows with a gatekeeper is presented in figure 2.1 whereas Endpoint A makes a call to Endpoint B through Gatekeeper (Patrick, 2009).

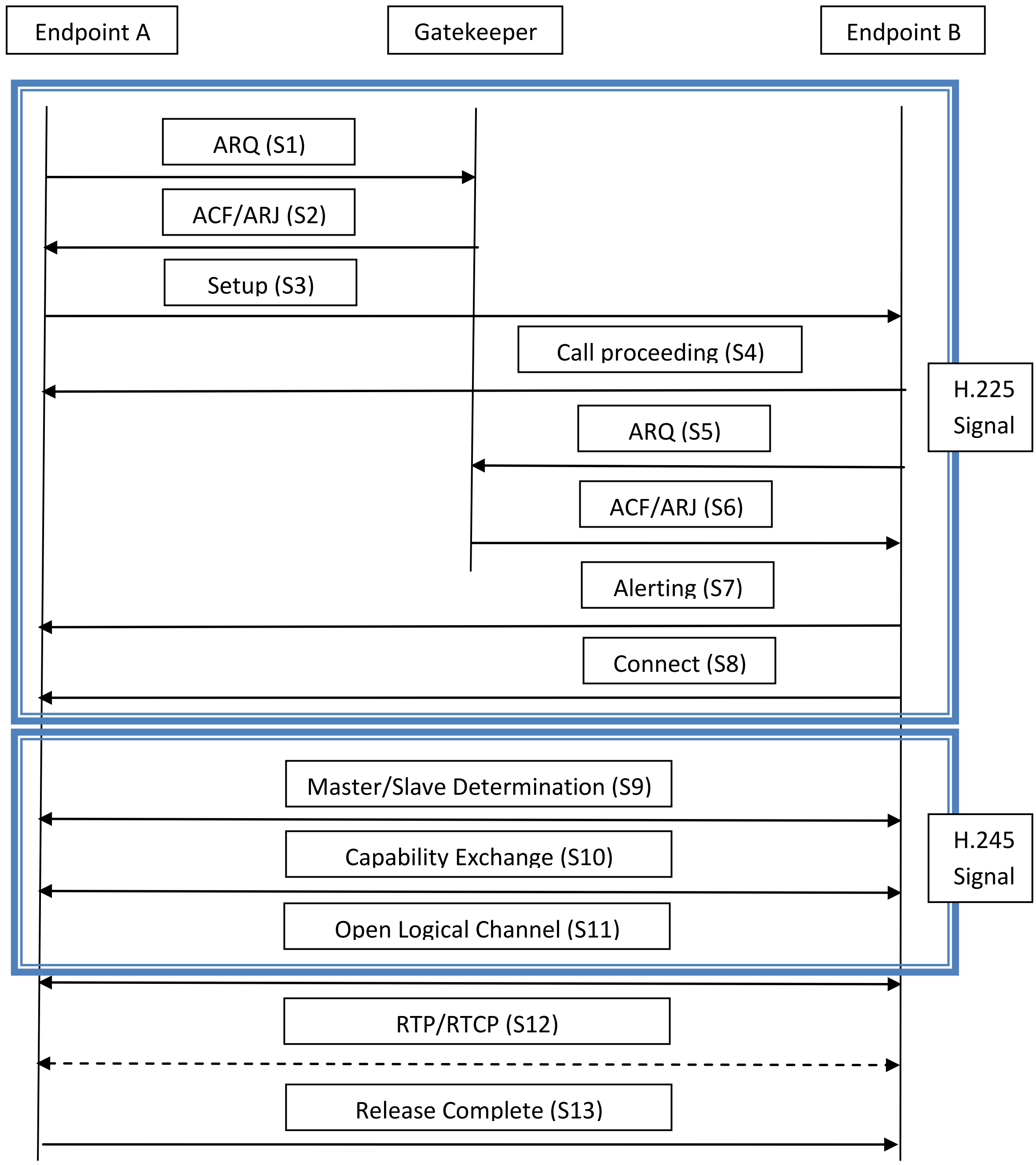

Figure 2.1 : Basic Call Flow

In the scenario shown in figure 2.1, both endpoints are registered to the same gatekeeper and the gatekeeper has chosen direct call signaling. Here is the description of each signal:

- S1- Endpoint A (calling endpoint) initiates the ARQ (admission request) to Gatekeeper.
- S2- Gatekeeper responds ACF (admission confirms) with the call signaling channel transport address of Endpoint B (called endpoint).
- S3- Endpoint A then sends the setup message directly to Endpoint B using that transport address.
- S4- Endpoint B responds call proceeding to notify its processing.
- S5 and S6 if Endpoint B wants to accept the call, it initiates an ARQ/ACF exchange with Gatekeeper. It is possible that an ARJ (admissions reject) is received by Endpoint B, in which case it sends release complete (disconnection) to Endpoint A.
- S7- when Endpoint B is ringing, it responds alerting.
- S8- when the user picks up the call, Endpoint B responds with a connect message, which contains an H.245 control channel transport address for use in H.245 signaling.
- S9- signals between Endpoint A and B to determine the master and slave of the call to avoid conflicts.
- S10- signals to exchange the media capability of each endpoint.
- S11- signals to open logical channels for transferring multimedia data.
- S12- RTP/RTCP channel is opened.
- S13- Endpoint A disconnects the call by sending release complete.

2.4.2. SIP Overview

SIP is an application-layer control protocol that can launch, modify, and terminate multimedia sessions such as Internet telephony (VoIP) calls. SIP can also request participants to already existing session such as multicast conferences. In addition, Media can be added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility- users that can maintain a single externally visible identifier regardless of their network location (Rosenberg, Schulzrinne, et al. 2002).

2.4.2.1. Basic Call Flow

The call flows between SIP clients and servers are various depending on the service architecture. One of the common call flows is shown in figure 2.2, assuming that servers share the registration information of users, and clients have to send INVITE to a redirect server first when initiating a call (Patrick, 2009).

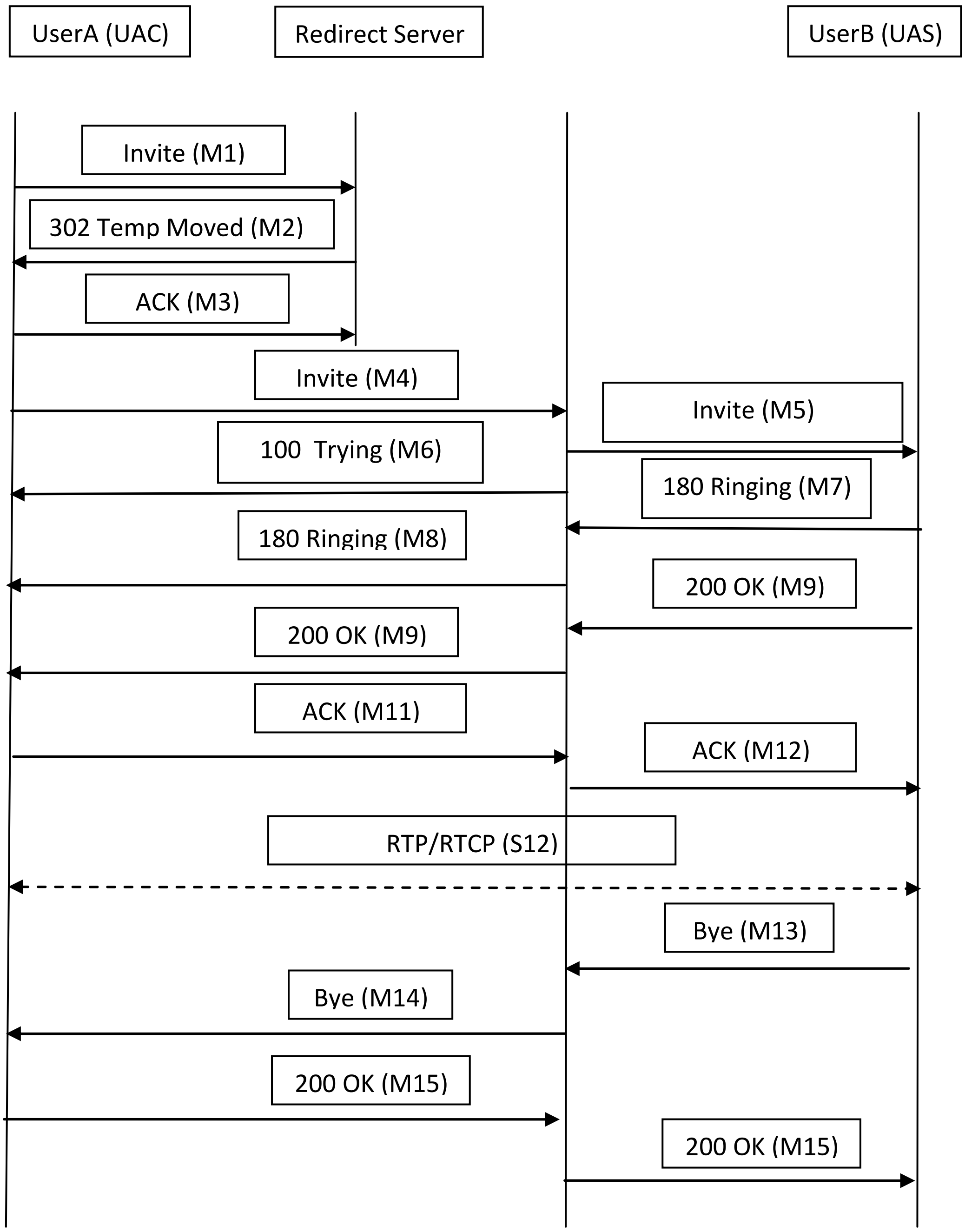

Figure 2.2 : Basic Call Flow

At SIP scenario there are many basic steps to make connection between two users as follows:

- M1 User A (UAC) sends invite message to redirect server first to make a call to User B (UAS).

- M2 the redirect server returns 302 moved temporarily response containing a contact header with User B’s current SIP address.
- M3 Acknowledgment.
- M4 and M5 User A then generates a new invite with SDP and sends to User B via a proxy server.
- M6 notifies that the proxy server received the request and continues to process it.
- M7 and M8 User B sends 180 ringing when the phone is ringing.
- M9 and M10 User B sends 200 OK with SDP when picking up the phone.
- M11 and M12 acknowledgment. After this, the media channel (RTP/RTCP) is opened.
- M13 and M14 User B sends bye when hanging up the phone.
- M15 and M 16 confirmation of disconnecting.

This is typical example of call flow to set up and disconnect a SIP dialog among UAC, UAS, and a proxy server.

2.4.3. RTP& RTCP Overview

RTP is a protocol that supports the user voice. Each RTP packet contains a small sample of the voice conversation. The size of the packet and the size of the voice sample inside the packet will depend on the CODEC used. For these reasons, RTP is important for Simple Multicast Audio Conference, Audio and Video Conference, and Mixers and Translators (when sites want to receive media data in the same format). (RTCP) Real-Time Control Protocol is a data plane protocol that is not always used. This protocol allows the endpoints to communicate directly regarding the quality of the call. RTCP affords the endpoints the ability to adjust the call in real time to increase the quality of the call RTCP also aids significantly in the troubleshooting of a voice stream. Traditional VoIP analyzers sit at specific locations on a circuit and base their derived results from only the packets that they capture. With RTCP enabled, the analyzer can see the end-to-end quality as well as the quality at the point at which the analyzer is inserted, allowing the user to sectionalize problems much more quickly (Schulzrinne, Casner, et al.1996).

2.4.4. TCP Overview

TCP is almost never used for VoIP, for while it does have mechanisms in place to ensure delivery, it is not inherently in any hurry to do so. Unless you have an extremely low-latency interconnection between the two end points, TCP will tend to cause more problems than it solves. The purpose of TCP is to guarantee the delivery of packets. In order to do so, several mechanisms are implemented, such as packet numbering (for reconstructing blocks of data), delivery acknowledgment, and re-requesting lost packets. In the world of VoIP, getting the packets to the endpoint quickly is paramount—but 20 years of cellular telephony has trained us to tolerate a few lost packets. TCP's high processing overhead, state management, and acknowledgment of arrival work well for transmitting large amounts of data, but they simply aren't efficient enough for real-time media communications (Stevens, Wright 1994).

2.4.5. UDP Overview

UDP unlike TCP, it does not offer any sort of delivery guarantee. Packets are placed on the wire as quickly as possible and released into the world to find their way to their final destinations, with no word back as to whether they got there or not. Since UDP itself does not offer any kind of guarantee that the data will arrive, it achieves its efficiency by spending very little effort on what it is transporting (Rosenberg, Weinberger, et al. 2003).

2.5. How Does VoIP Technique Work?

Basically, VoIP means Voice move over the network. When VoIP was first developed, it functioned only with the Internet and nothing but the Internet. Today, VoIP operates over most network types, including those used during the corporate area. Protocol defines as a type of rules that used in the network to send and receive signals. These signals are represented as series of 0s and 1s which are high and low electrical or optical pulses of the digital networking. IP Telephony works by adapting voice communications into data packets. Conveniently, it runs on the current Ethernet LAN technology, which currently supports over 96 percent of all companies' needs for LANs. Moreover, Each LAN in a multi-location enterprise network is linked to the larger WAN. For examples, if someone is located at the headquarters in Pittsburgh, and he/she calls a co-worker located at the office in Los Angeles, his/her call begins as IP Telephony call. Then it goes from head LAN at Pittsburgh through an edge device.

Edge devices contain products such as the Extreme Networks' Unified Access allowed switch. The edge device is programmed to re-packetize the call and encode the larger VoIP packet with the additional necessary information such as the address for the destination LAN or the mobile end-user. In order for the LAN to participate in the company’s VoIP WAN, each LAN needs at least one edge device such as a router, three level switch, or a gateway. These devices, like all other addressable devices on the LAN, have a MAC address and a NIC to physically connect them to the LAN. But in addition, they each have an interface card that physically connects them to the company’s WAN or some external network. Depending on the company’s network design, size, and mission, these edge devices can have multiple interfaces that connect them to multiple outside networks. The point is that these edge devices manage all the IP Telephony traffic going off-LAN by encapsulating the signals into packets, encoding the packets with the correct addressing information, and forwarding the packets out onto the WAN where they make their way in a packet-switched manner to their respective destination LAN.

VoIP traffic on the WAN uses the IP addressing scheme. When the packets arrive at the destination LAN, the edge device breaks down the VoIP packets and forwards them internally to the server that manages the IP Telephony services on the LAN. From this point, the rest of the process is similar to IP Telephony services. The phone rings; the person being called answers and a virtual circuit is established between the caller and the person receiving the call. Instead of maintaining separate networks for computers and telephones, companies can converge both of these networks into one network using IP Telephony and VoIP.

2.6. VoIP Planning And Implementation

The amount and type of IP Telephony products and services will be exclusive to an organization's existing telephony configuration and future requirements.

Additionally, there are multiple configurations for connecting IP phones to the network as figure 2.3 describes. Converged networks extend IP networks to leverage a common infrastructure for voice, video, data and all other converged communications. Main telephony equipment vendors have stable IP-based solutions to complement their traditional TDM voice switching gear. Network foundations vendors have enhanced their products specifically for converge applications and are working narrowly with application vendors to fully exploit the underlying intelligence in the network. Organizations have a choice of proven solutions and are not locked into a single vendor for a highly functional end-to-end solution. The move to a converged communication infrastructure should be incremental and protect investments in existing legacy infrastructure to ensure a smooth, low risk transition. Voice traffic is converted to IP telephony packets by different devices, depending on the architecture of the solution. In most IP telephony deployments, packet conversion occurs at the IP phone as well. In the case of architectures involving analog phones and phone hubs, IP telephony conversion follows at the phone hub. Most IP phones use industry standard protocols such as SIP or H.323 to connect with other IP devices. However, similar to TDM-based PBX and certain proprietary handsets, not all IP phones will work with just any vendors' IP call manager due to proprietary client/server protocols. For an organization that is considering voice and data convergence, putting IP telephony on the WAN is as important as IP telephony on the LAN. IP telephony on the WAN is where the advantages of toll bypass become evident. The reasons for this are mainly inexpensive. Cost savings can be direct when long distance phone calls are diverted from PSTN and sent over an existing IP-based WAN. Implementing IP telephony is infrastructure requirements to be an evolutionary step with proper reflection for legacy TDM PBX systems and adequate design. (Gillispie, 2005).

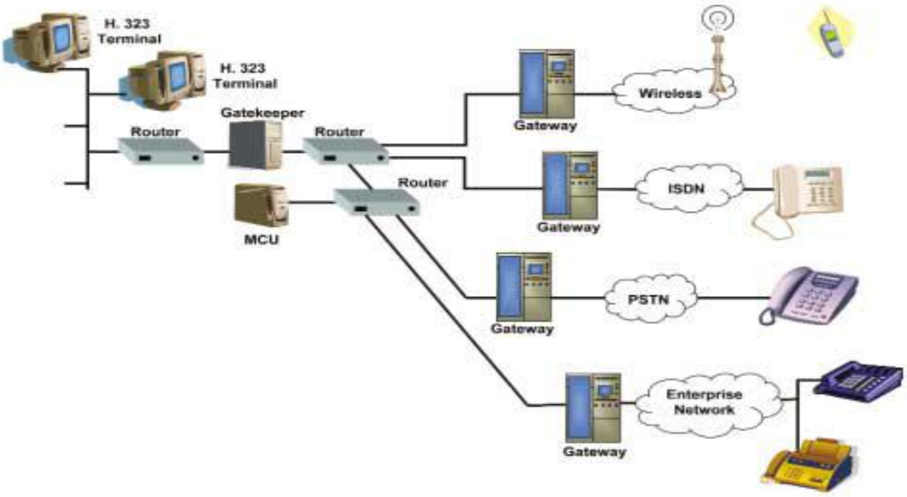

Figure 2.3: Connectivity Options (Gillispie, 2005)

2.7. Related Works

There are many researches had been done before using VoIP technique. This technique had been used in order to improve the telecommunication mechanisms in terms of efficiency and the cost. One of the researches that had been already done is the study released by Hitachi in April 2000. It proposes a VoIP-supporting version of its main force CX8000 series private branch exchange (PBX) and the NT-1000 VoIP gateway, which can be communicated with PBXs other than those in the CX8000 series. This company tries to build VoIP networks on a large scale. It rebuilt Time Division Multiplexing (TDM) based corporate network to make it an IP-based one. In order to create a supply chain in Hitachi Company and share resources among research laboratories, the company needs to use IP network which can support the need for high speed transmission of IP data and reduce the communication cost. Thus, Hitachi provides optimal and efficient corporate network solutions (Kojima, Ôshita, Kaneko, & Maruyama, 2000).

One of the most popular VoIP systems is Skype. According to (), Skype is a peer-to-peer VoIP client developed in 2003 by the organization that created Kazaa. Skype claims that it can work almost seamlessly across NATs and firewalls and has good voice quality than other VoIP users. It encrypts calls end-to-end, and stores user information in a decentralized style. Skype also supports instant messaging and conferencing. Skype allows its users to make voice calls and send text messages to other users of Skype clients. In essence, it is very similar to the MSN and Yahoo IM applications, as it has capabilities for voice-call, immediate messaging, audio conferencing, and buddy lists

From the side of bandwidth and security are addressed, more users will have access to Skype, and the service will continue to add services, increase quality, and connect with other applications. Skype has shown that people are not only interested in free calls but also are willing to pay for services that are inexpensive compared to the alternatives. In this way, Skype might anticipate the evolution of voice and video communication by taking advantage of the power of the Web. To the extent that wireless Internet access approaches completely, Skype becomes a real competitor to cell phones, and this could have a deep effect on the communication landscape. In the meantime, vendors are introducing new Skype strategies, some of which work as landline and Wi-Fi phones. One day people might make phone calls or have video conferences without having to know (or care) how that communication happens. (Baset & Schulzrinne 2004).

2.8. Chapter Summary

The analysis of literature review had broadened the VoIP issues. The information and findings collected form this chapter is used as a guidance to implement the VoIP system.

CHAPTER THREE

RESEARCH METHODOLOGY

3.0. Introduction

This chapter discusses the adopted methodology which applied to achieve the objectives of the study in designing and implementing the proposed VoIP system for UUM campus.

3.1. Research Methodology

Research methodology plays a very important role to proceed and carry out with the whole research study. Moreover, it is very important to choose the suitable methodology for the study in order to achieve the objectives of the study. In general, a lot of studies that had been done before using the research methodology to achieve many purposes such as gathering data and information, development and evaluation (Refsdal, 2008 & Schmuller, 2002). Besides that, research methodology step makes us fully awareness about the requirements of our study and the problem statement of the research. Thus, the methodology of this research is based on the five general research steps that are proposed by Vaishnavi & Kuechler. These steps includes the awareness of problems, suggestions, development, evaluation, and the conclusion of the research as they are illustrated in the Figure 3.1.

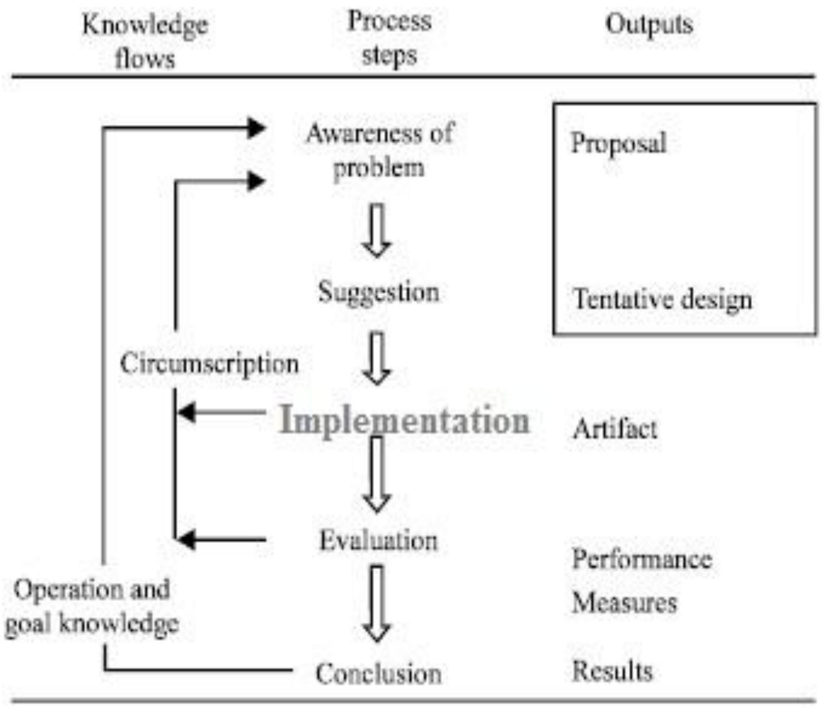

Figure 3.1 The research methodology (Vaishnavi & Kuechler, 2004)

3.1.1. The Awareness Of Problems

In this step, some important information had been gathered about the current problems and limitations of using the current traditional communicating mechanisms at UUM campus. This can be achieved by interviewing the students, lectures, and staff to see what are the disadvantages of using the current communicating techniques in order to overcome it. Moreover, we can use some beneficial instruments such as questionnaire. In addition, we can collect some information from some related work that had been done before by other researchers. Thus, this step aims to identify the problems and come out with a significant and more organized solution.

3.1.2. Suggestions

The second step of the methodology of this research is to suggest designing and implementing a VoIP application in order to enhance the current communications tools used by UUM campus. In this phase, some flow charts and diagrams are used to design and identify the proposed system.

3.1.2.1. Hardware Requirements Of The System

Since the proposed system is a VoIP application, so to run the system on the UUM campus we need LAN network (wired / wireless) and for advanced communication operation it is important to have SIP server device to manage the calls. In addition to run the system we need PCs. On the other hand, the users should combine speakers and microphone to their devices.

3.1.2.2. Software Requirements Of The System

As we mentioned in the hardware requirements of the system, we need to install the Linux operating system for installing the server system. After that, we will install the Asterisk open source system. Besides that, the user should have softphone program which supports VoIP applications.

3.1.3. Implementation

A system implementation process is a process to implement a software product or to enhance an existing one. Moreover, the implementation stage of the research methodology is the most significant stage in our study since it represents the answer of the problem statement also it represents the main objective of this study (Hoffer, J. et al., 2002 & Liu, 2002). Generally, system implementation process consists of the three distinct phases are Analysis, Construction, and Testing. These phases follow each other smoothly; each phase of implementation proceeds in strict order, without any interfering or repetition steps.

3.1.3.1. Analysis

The purpose of the requirement capture analysis is to aim the implementation toward the right system. Its goal is to produce a document called requirement specification. Moreover, the requirement specification is the official statement of what is required of the system to be developed. It must state what to be done rather than how it is done. It must be in a form which can be taken as the starting point for the system implementation. A specification language is often used in this stage to translate the user requirements in a form that is beneficial to use in the next stage.

3.1.3.2. Construction

System design is realized as a set of programs or stages. All stages coordinated with each other in order to make up the VoIP System. This phase ensures that all modules that had been configured for the VoIP system meet the system specifications. Primarily, this stage consists of design and implementation. The representation of the VoIP system consist of several steps starting from doing the configuration for the server ending with establishing the connection between the users and server using softphone. (Ivar et al., 1993).

3.1.3.3. Testing

In this phase, each individual module that is implemented for the VoIP system has to be tested as a complete system to ensure that the system meets the UUM requirements. From another side, the connection operation between the server and the user need to be tested to ensure that the operation has completed successfully. To build a testable system, the need of building the system in such a way that it depends on testing each stage isolatable before we test the system as a hole component.

3.1.4. Evaluation

The next step of the implementation is the evaluation. In this step, the system has been evaluated as a one component to make sure that the results meet the research objectives. In more details, it includes verifying whether the users of the system are satisfied while using this system, what are the enhancements that the system provides compared to the previous VoIP system that had been used? Is it effective and easy to use by the users?

3.1.5. Conclusion

This step includes the results that have to be achieved after completing the study. In addition, a future work can be included in this section.

3.2. Summary

As a summary, this chapter has discussed the project methodology used to develop the VoIP system for UUM campus. This chapter also has described in depth the general research steps that are proposed by Vaishnavi and Kuechler**.**

CHAPTER FOUR

SYSTEM DESIGN AND IMPLEMENTATION

4.0. Introduction

This chapter describes the operation of VoIP system implementation. In addition, in this chapter we will explain our work step by step starting from preparing the system until making a simple call. Thus, this chapter includes discussing design stage and implementation stage. Design stage focuses on network design, types of services and call flow, while the implementation explains configuration operations, Dial plan, registering new user, choosing a soft phone program.

4.1. Design Stage

This stage consists of three sub stages including network design, type of service and call flow. In network design, we depend on the network structure of UUM. Then, at the type of service we will mention about VoIP services in general and we explain the kinds of calls between users using VoIP communication.

4.1.1. Network Design

The UUM has had a major impact on society by achieving significant growth, extending its services, and building more facilities to meet the natural demands of student community and staff. According to Computer Pusat at UUM, the structure of UUM network consists of switches, Pcs, several kinds of servers such as portal, and routers. In addition to the pervious of UUM network equipment that we just mention we need to add new server called Asterisk (SIP) as it is shown in figure 4.1. For more details about UUM network structure see appendix A.

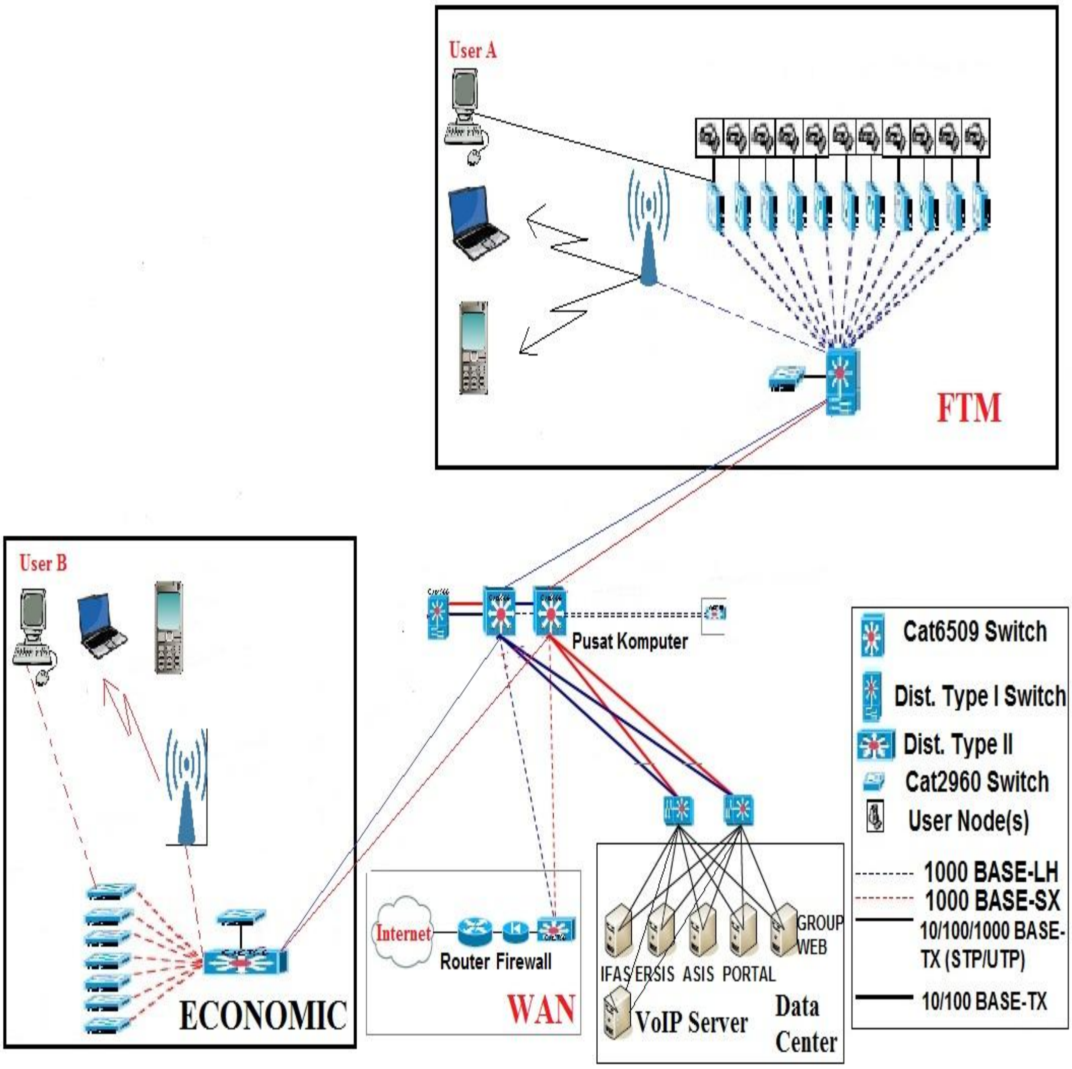

Figure 4.1: Proposed UUM Network Architecture For VoIP Implementation

4.1.2. Type Of Service

In general VoIP system provide several kinds of service such as voice mail, text message, FAX, video call, and voice call (PC to PC, PC to phone and vice versa, and phone to phone). In this proposed system, we concentrate on talking about voice call as following:

i. **PC to PC**

In this case, it is a software-based and a free service which allows users using their computers to place calls to each other for free.

ii. **PC to Phone And vice versa**

This service is a software-and-hardware-based service. It allows user to use his/her PC to call a landline or mobile phone.

iii. **Phone to Phone**

It is a hardware-based service. For phone to phone VoIP, the caller and the call recipient can use their phones to call each other using UUM network. VoIP converts audio sounds into data packets and transfers these packets over the network. The packets would then travel from one open router to another following the built-in packet address. Once the packets reached the call recipient, they would be converted back to audio sounds.

4.1.3. Call Flow

This section introduces how a call session between two users at UUM campus is established through a VoIP server. The reason for adding this section is to give deeper understanding of call flow between user A at FTM and user B at economic. Figure 4.2 is illustrates a typical example of voice message exchange between two users assuming that they are in the same domain and single VoIP server facilities the session establishment.

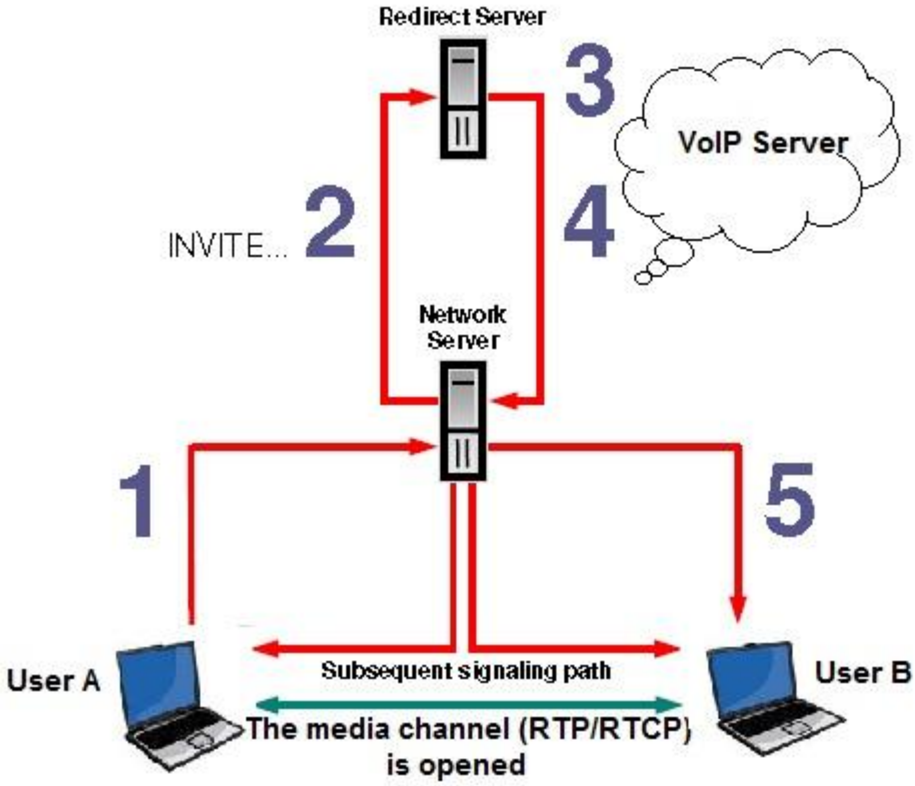

Figure 4.2: Call flow between two users at UUM campus

First, User A at FTM send invitation message to VoIP server to communicate with the User B at economic. Then, VoIP server redirect the request to User B. after that, the User B accept the request and send the acknowledgment to server. Now, the softphone ringing on User B at the same time can hear the ringing tone, when User B answer the media channel (RTP/RTCP) is established.

4.2. Implementation Stage

Based on the design stage we are going to achieve the implementation operation to the Asterisk system during the server and user configuration steps.

4.2.1. Server Configuration

After adding VoIP server, we should do many steps to make VoIP server ready to provide VoIP services as we will explain later. We should install the compiler of Asterisk system which called GCC. Then, GCC compiler will be installed with its dependency on our system. On the other hand, Asterisk also needs installing other new packages in order to work properly which are ncurses compiler for CLI (Call Level Interface) functionality, newt, and libxml2 (Van Meggelen, Madsen, et al. 2007).

After we finished installing required compilers, we can start downloading Asterisk by visiting Asterisk website //www.asterisk.org//. Then, we can get the link of newest version /asterisk-1.8.4/ to download the source code packages in our server. Controlling with configuration Asterisk system is flexible as well as choosing what module has to be built during building time, this can be done by writing the command `/# make menu select /` in order to access the configuration menu screen as you can see in figure 4.3. For more explanation about Asterisk installation see appendix B

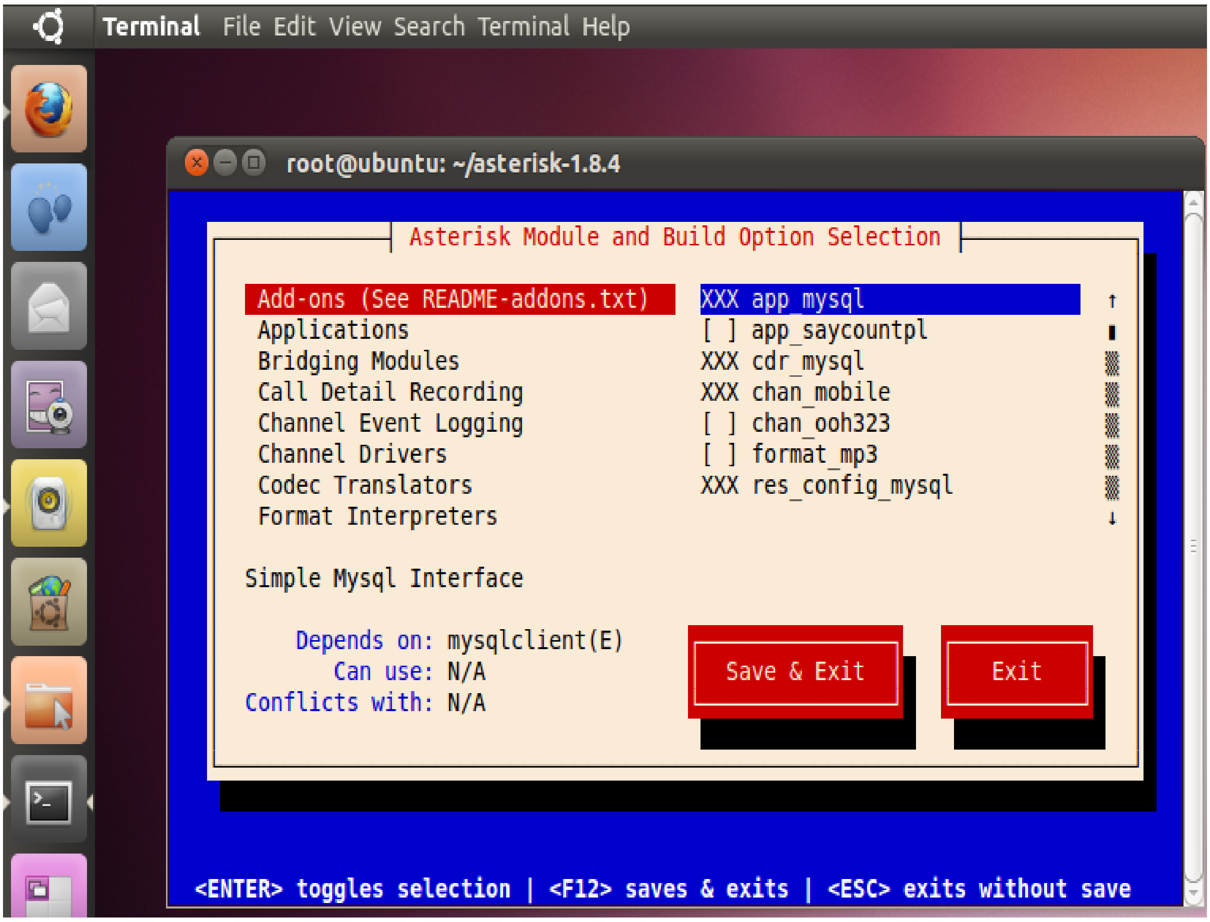

Figure 4.3: Menu select screen for the Asterisk software

At this time, our system is ready to configure operation and the command line for doing that is as the following:

- `#./configure`

Once the configuring has been done successfully, the screen will be shown as figure 4.4 to inform admin that the operation has been done successfully.

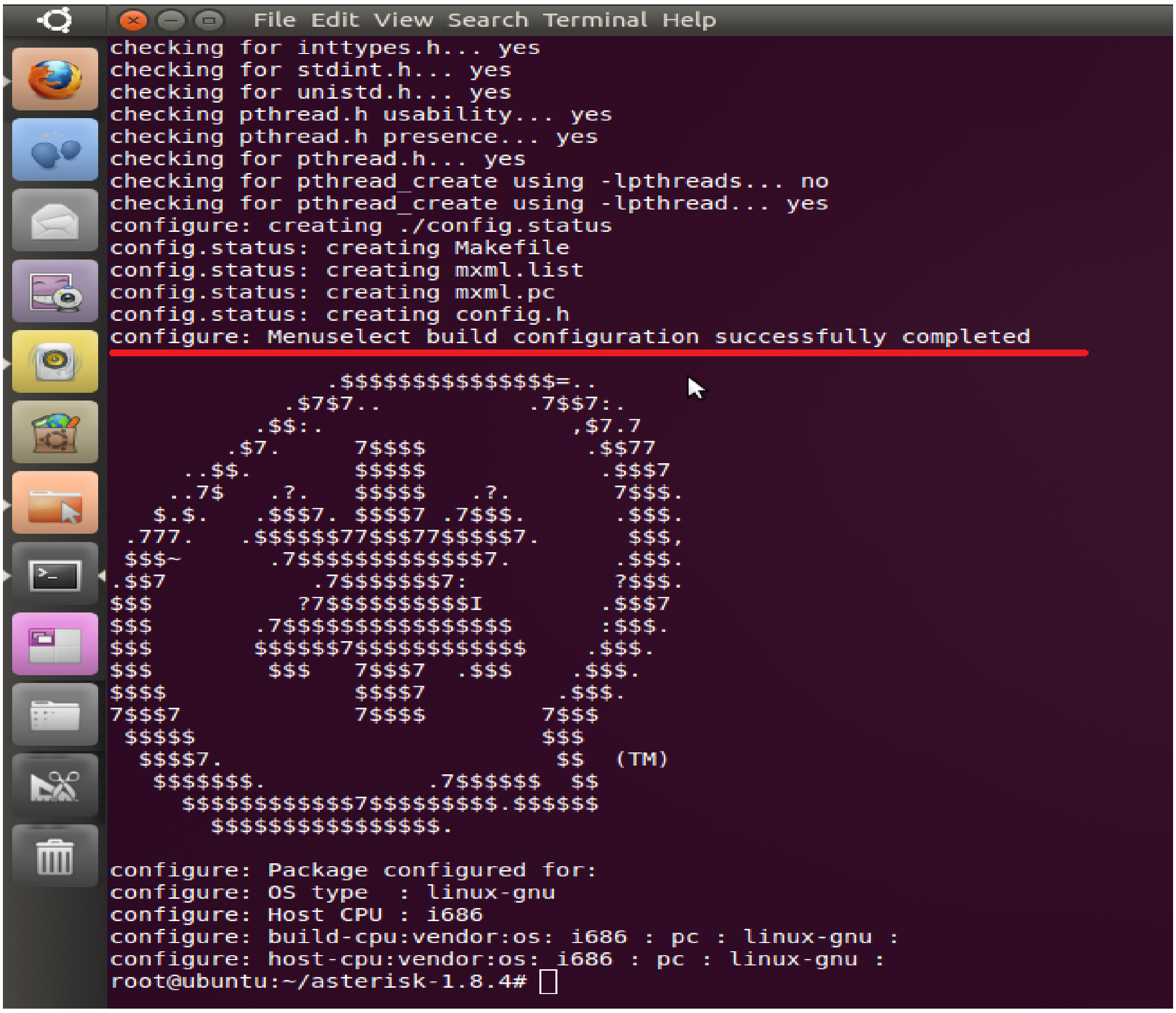

Figure 4.4: Build Menu select

Now Asterisk should compile all packages by using the command:

- `#make`

Executing this command needs some time to finish and when it finishes, we can use the command as the following:

- `# make install`

Figure 4.5 shows how Asterisk has been built successfully. At this point, Asterisk is ready to execute the command /make install/. Then, and by using Asterisk software we can create sample by writing the following command:

- `# make samples`

Then use command:

- `# make config`

This will install the start-ups crypts' and configure the system to execute Asterisk automatically at start up.

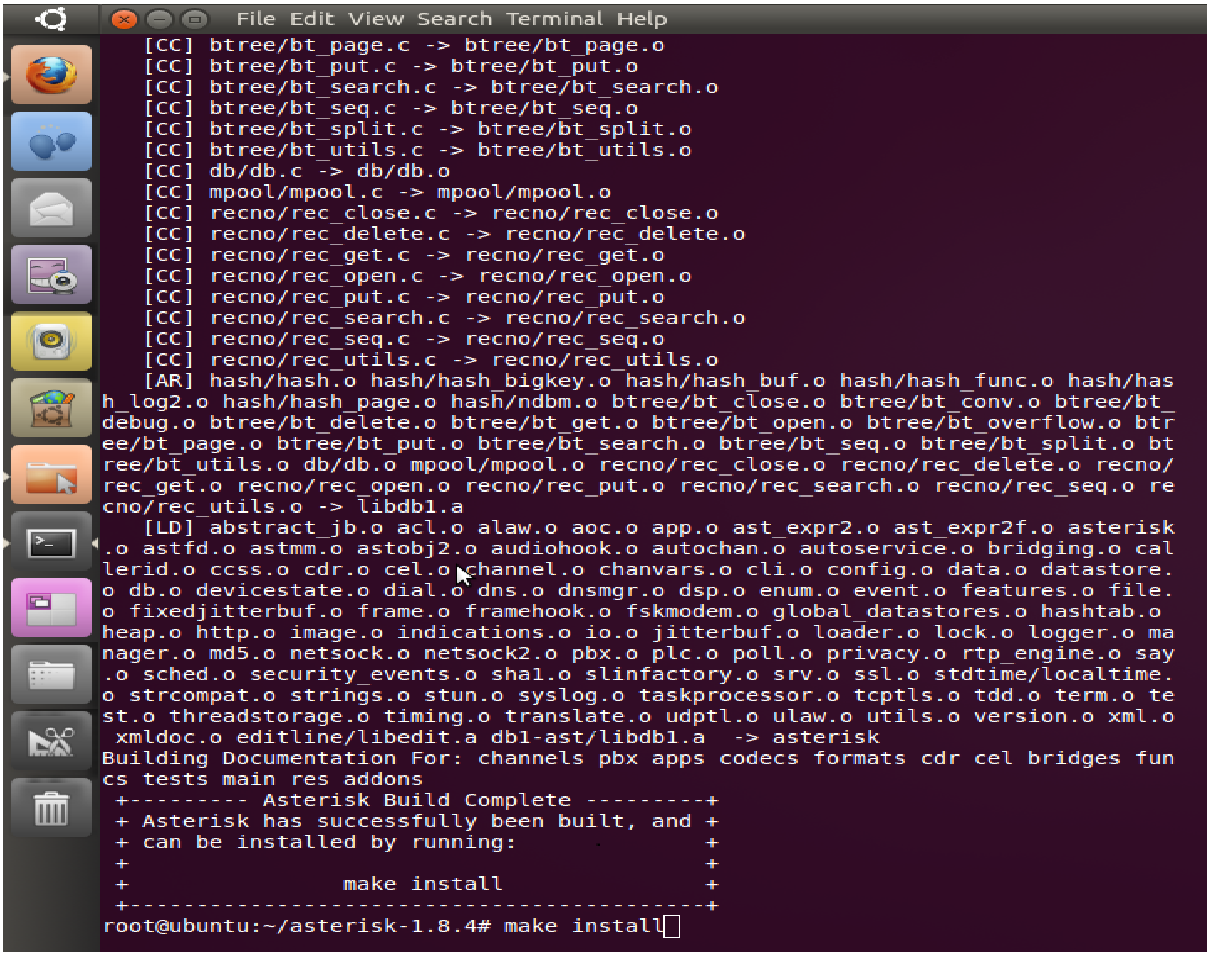

Figure 4.5: Finishing the installation

4.2.1.1. Define SIP user

Asterisk doesn't need any special hardware requirements. The users can use any soft phone program and using any operating system (Linux, Windows). Then, he/she can register to server directly as we will see later. After that, the server routes his/her call to the destination using Internet or local network if both of the communicating users are exist in the same place and using the same local network. There are many configuration files which can be used to complete the configuration operation and the following configuration files are:

- extensions.conf

The dialing plans that we have created will be extremely primitive, but they will improve the system in terms of speed.

- sip.conf

This is where we'll configure the SIP protocol. In the following sections, several configuration files will be edited. Reload these files after any changes in order to see the effects of these changes. Make a backup copy of the sample extensions.conf (try the bash command //# mv extensions .conf extensions. conf.sample//), and then create a blank extensions.conf file (usingthe bash command // #touch extensions.conf //), then insert the following scripts:

- `[globals]`
- `[general]`
- `autofallthrough=yes`
- `[default]`
- `[incoming_calls]`
- `[internal]`
- `[phones]`
- `include => internal`

This file will be used to build a test dial plan so we can ensure that all of our devices are working properly. Also, be sure to run the dial plan reload command from the

Asterisk CLI to update the latest changes that had been done before. Then, verify these changes by running the CLI command dial plan as shown in figure 4.5:

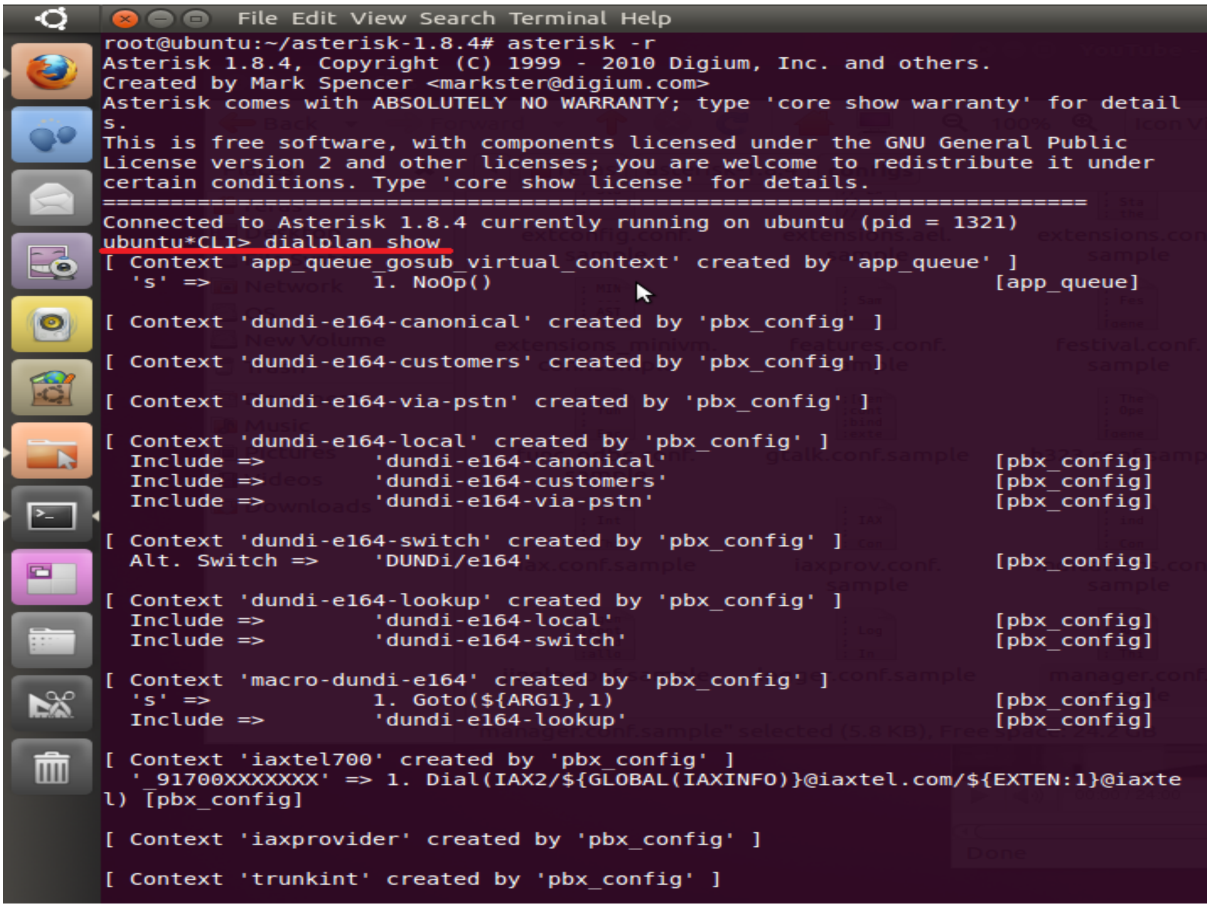

Figure 4.6: Dial Plan Show

It is the simplest way, and from a working foundation, it is much easier to take a basic configuration. Just as had been done before with the extensions.conf file; run the following commands in your bash shell:

- `# mvsip.confsip.conf.sample`
- `# touch sip.conf`

After including the following scripts in a sip.conf file, the user will be able to register in the system.

- [general]
- [805733]
- type=friend
- context=phones
- host=dynamic

Even though we have named this SIP device [805733] and we are probably going to assign that extension number to it whatever he/she wants. Since we want to be able to do both send calls to the soft phone and allow the client to place calls, we have defined the type as friend. The other two types are user and peer. At Asterisk side, a user is configured to enter the dial plan, and a peer is created for calls. A friend is simply a shortcut that defines both a user and a peer. In the case of doubt, define the type as friend.

The host option is used to define where the client exists on the network and when the VoIP server needs to send a call to it. This can either be defined statically by defining something like host=192.168.1.100, or if the client has a dynamic IP address, then we set host=dynamic. When the host option is set to dynamic and the client is configured to register, VoIP server will receive a REGISTER packet from the endpoint, telling that VoIP server which IP address the SIP peer is currently used. If the network is not trusted, it is better to use a password by adding the following to the device definition. This is one of those things that are not technically necessary, but it is probably a good idea:

- secret=guessthis

It has elected simply to call it the Unique Identifier. This is much better than the usual suffering that new users go through, as they change settings in both places and have no luck getting a phone to register.

4.2.1.2. Dial Plan Scripts

In order to test dial plan which allows dialing back into soft phone, we need to do the required configurations and using the function named Echo (). This function will allow testing bidirectional audio. Now, add the italicized bits to existing extensions.conf file then the code expands on it in certain sections. After entering the text into extensions.con f, reload the dial plan by running dial plan reload from the Asterisk console:

- [globals]
- [general]
- [default]
- exten => s,1,Verbose(1|Unrouted call handler)
- exten =>s,n,Answer()
- exten =>s,n,Wait(1)
- exten =>s,n,Playback(tt-weasels)
- exten =>s,n,Hangup()
- [incoming_calls]
- [internal]
- exten => 500,1,Verbose(1|Echo test application)
- exten => 500,n,Echo()
- exten => 500,n,Hangup()
- [phones]
- include => internal

4.2.1.3. Application Testing

After doing the required operations for setup voice call, we can now start initial test for server performance to make sure that the services work correctly and give the expected result. On the softphone side we add new SIP account by putting the user name and the domain then the user can register on the server. After the user registers successfully, the user will make a call and the server will answered by sending voice message as a test. This voice message indicates that the connection between the client and server established successfully and the services what we configured before work correctly as figure 4.7 is shown.

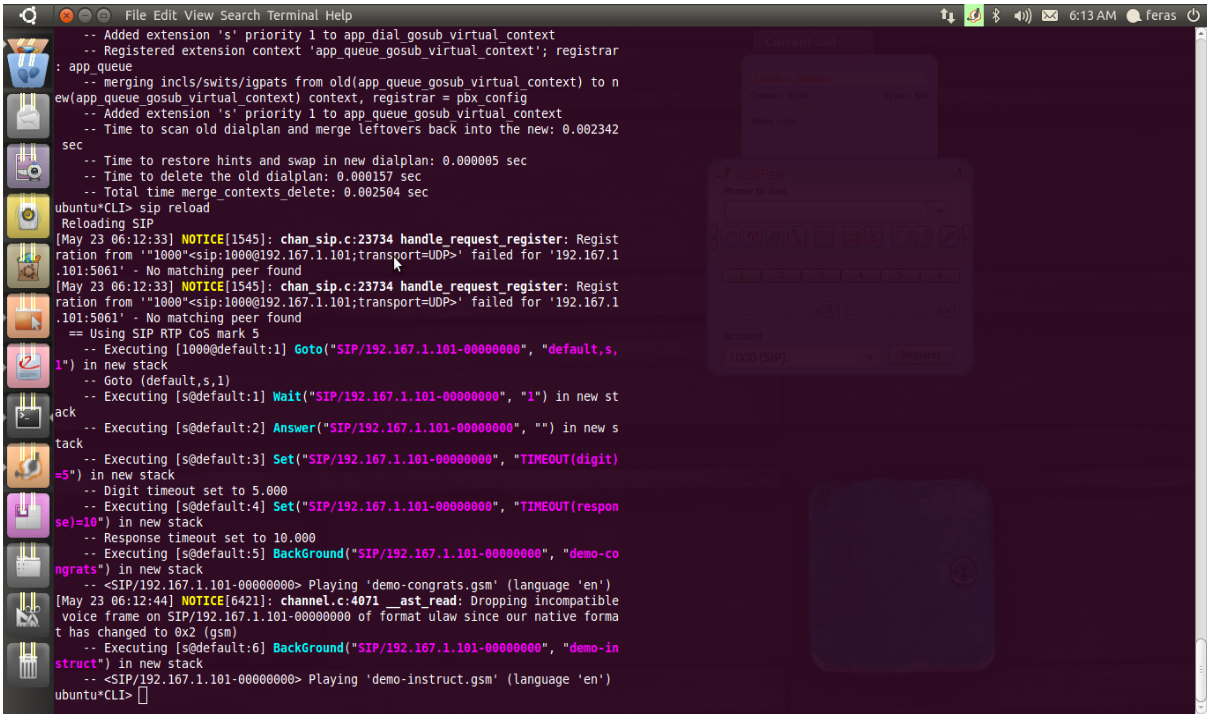

Figure 4.7: Applications Testing

4.2.2. User Configuration

In this section, we will configure the ZoIPer softphone in order to make a connection with Asterisk program. The IP address of the phone is 192.168.1.250, and Asterisk is located at 192.168.1.100. The ZoIPer is available for Microsoft Windows, Mac, and Linux. User can obtain a copy of ZoIPer from http://www.zoiper.com/download_list.php. Now let’s configure this softphone to make the connection with Asterisk box. If Asterisk has not already started, then start it now. If Asterisk was run in the background, admin can restart Asterisk system to the CLI by running the following command:

- `# asterisk-rvvv`

Admin will then be given the Asterisk CLI like so:

- `*CLI>`

If Asterisk was run before changing the sip.conf as instructed in this chapter, then reload the dial plan and SIP channel with the following two commands:

- `*CLI>dialplan reload`
- `*CLI>sip reload`

After register ZoIPer to Asterisk and if it has done successful, we can see the following at the Asterisk CLI:

- Registered SIP '805733' at 192.168.1.250 port 5061 expires 3600

Admin can verify the registration status at any time like so:

- `*CLI>sip show peers`

Name/username Host Dyn Nat ACL Port Status

805733/805733 192.168.1.250 D N 5061 OK (63 ms)

1 sip peers [1 online, 0 offline]

4.2.2.1. Choose VoIP Softphone Program

There are a lot of softphone programs which work under VoIP application. For our project, we used ZoIPer program softphone because it is simple, functional, easy to use, and it is free software. This program has friendly graphical user interface which allows the user to use the provided services easily. In addition to make voice call between two PCs, the program let the user create new account, check voice mail and sending fax as shown in figure 4.8.

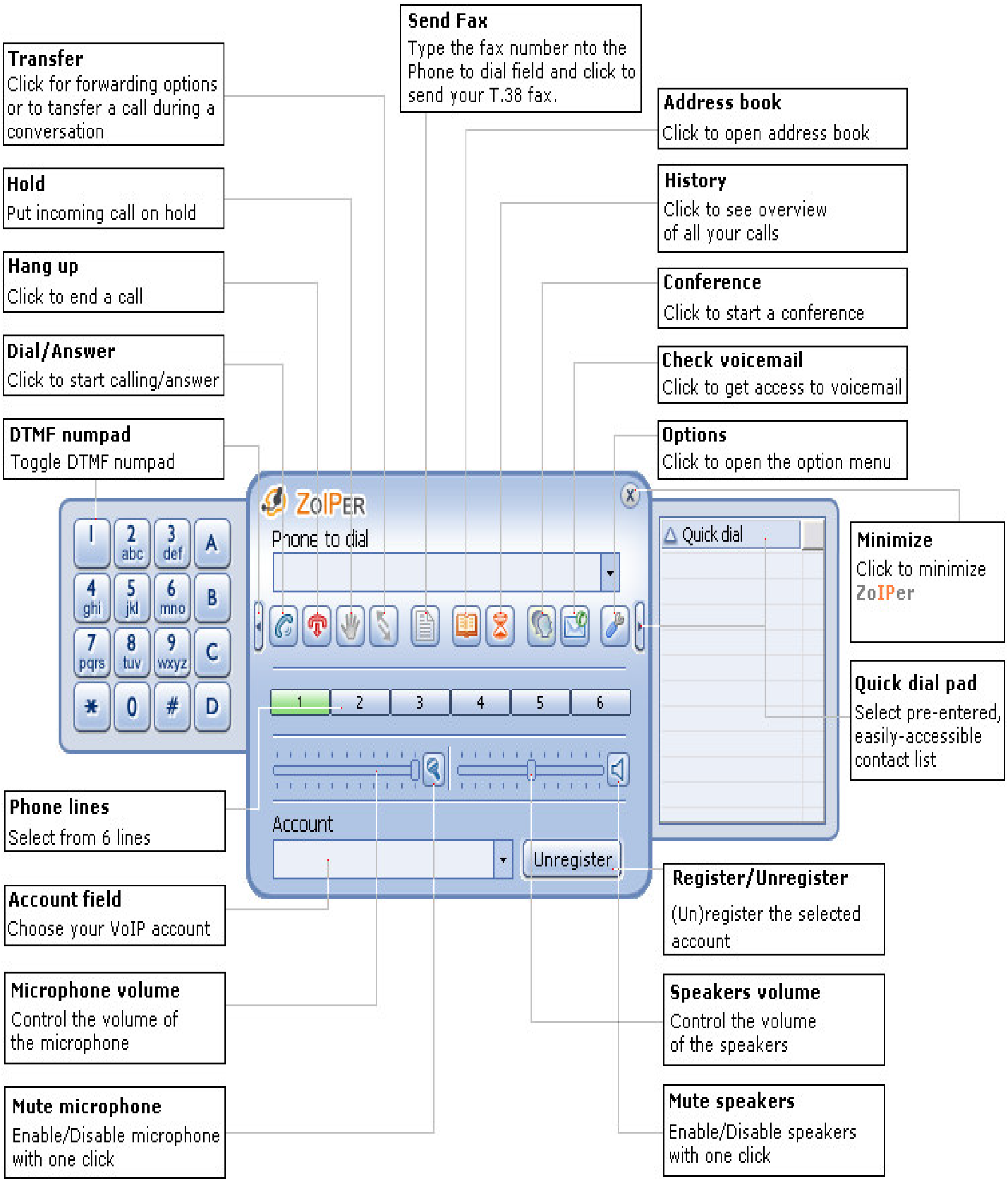

Figure 4.8: ZoIPer Interface

4.2.2.2. Testing VoIP System By Creating SIP Account

To let the user communicate with the Asterisk server, we should define a new SIP account and fill the information at it. From ZoIPer program, press on "option" bottom and it will open new windows as figure 4.9 is shown.

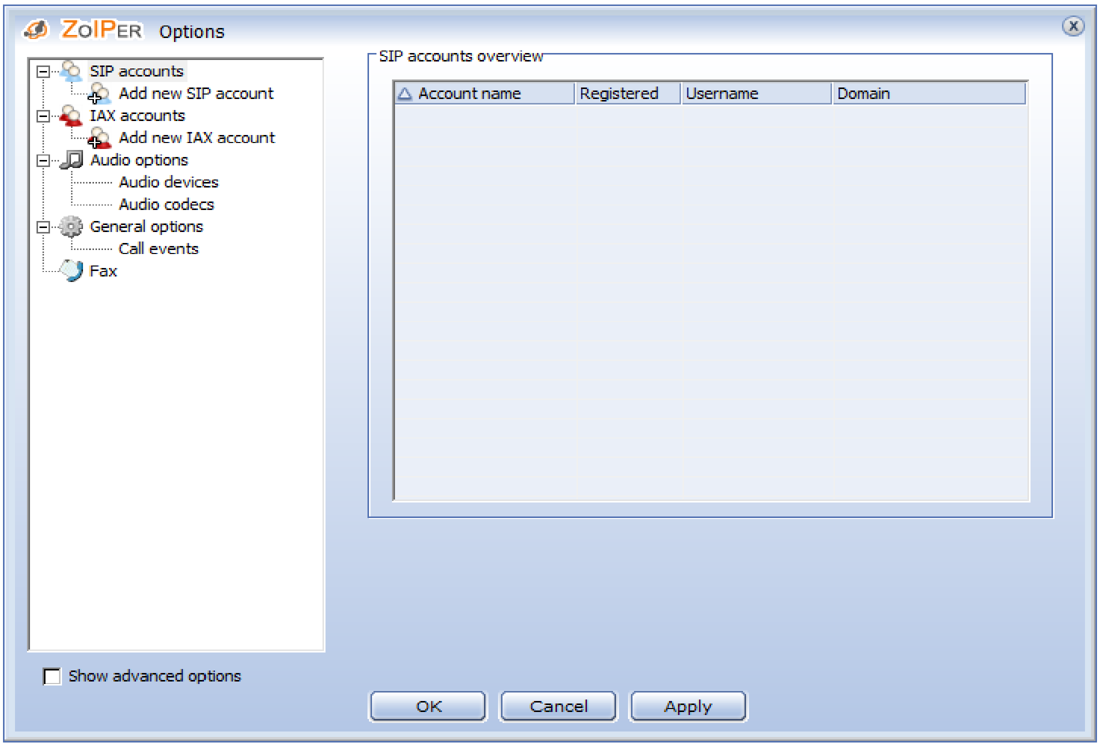

Figure 4.9: Option ZoIPer window

At this widow, we will create a new SIP account by pressing on add new SIP account then it will appear widow to enter the name of this account as figure 4.10 is shown.

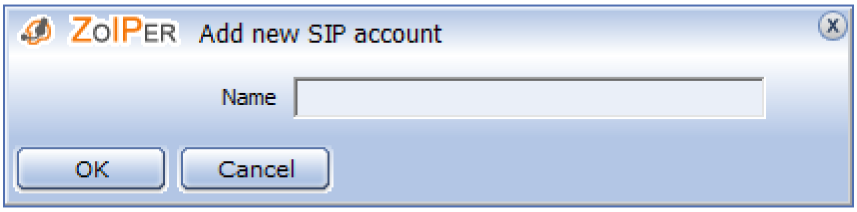

Figure 4.10: Add SIP account

After entering the name and pressing on OK button, we will enter to SIP option window to fill in the information as figure 4.11 is shown.

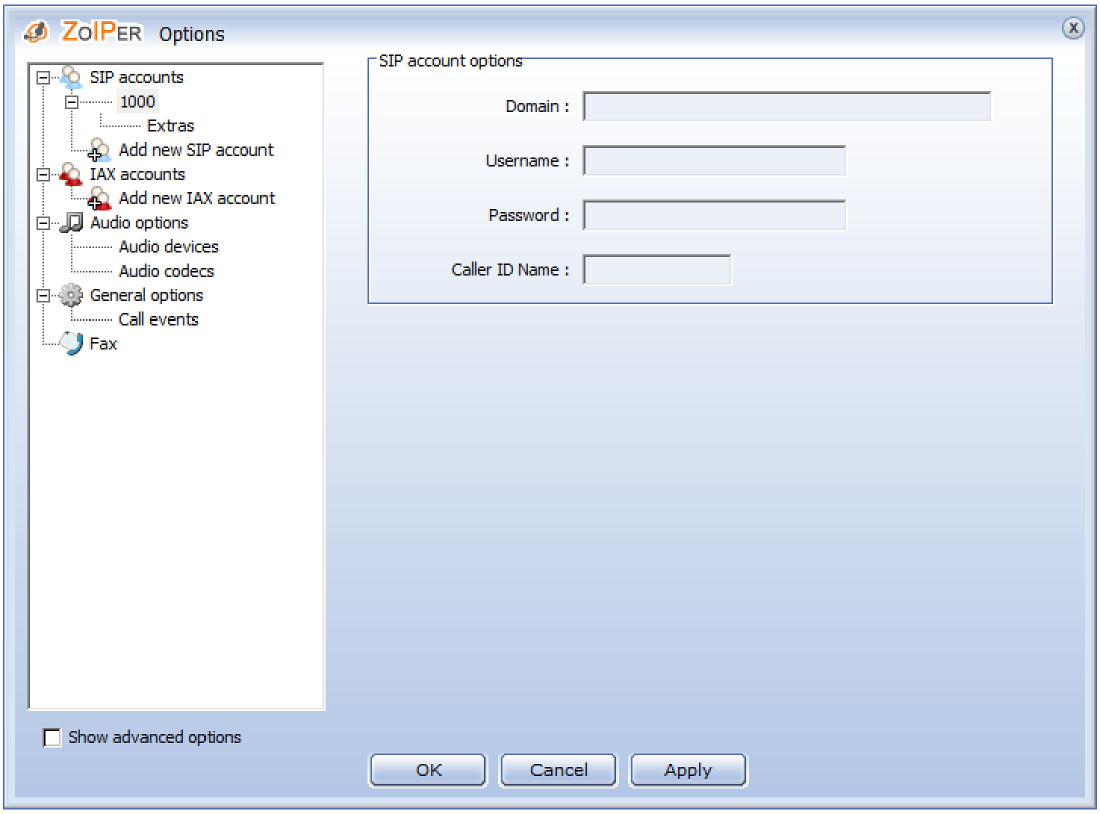

Figure 4.11: Field of account

In figure 4.11 the user will add a value as a following:

- Domain: enter the host name or IP
- Username: enter the username given from server for registration.
- Password: if there is password, if not no need.
- Caller ID Name: enter it as it given from server.

When the user finishes filling in the information, he/she has to press on apply to let the user register on Asterisk server and verify them. Finally, the user now can make a call and communicate with another user in the destination side.

4.3. Summary

In concluding, this chapter covers the design of VoIP network structure depends on UUM network structure. On the other hand implementation has been discussed widely in this chapter in order to give a clear overview about installing Asterisk, configuring operation, basic dialplan and testing.

CHAPTER FIVE

EVALUATION OF THE SYSTEM

5.0. Introduction

This chapter introduces the evaluation phase of the proposed VoIP system for UUM campus. The system has been evaluated by performing a monitoring user's performance on a carefully constructed interview and questionnaire.

5.1. User Interview

Many interviews had been done among the UUM students. First, we explained the system to the users in order to make the idea of VoIP services provided by the system clear. Moreover, a detailed illustration has been discussed about the actions that the students can perform using the system. In addition, some examples and scenarios had been given to make the idea more clear. Ideas related to the system had been shared with the users and they also gave us some suggestions related to the softphone properties and responsibility of the system. Thus, the users that had been interviewed were really satisfied of using the system.

5.2. Evaluation Process Of The Application

Most of the students that had been interviewed are master and PhD. This study applied a descriptive statistic technique to measure the mean value of the participants towards the usage of VoIP system for UUM campus. Due to the very short time until completing this study, the domain of the population has been narrowed to include just 30 respondents in this study.

5.2.0. Evaluation Of Demographic Data

Table 5.1 shows the distribution of both respondent's age and degree. The respondents of the study divided into (9 respondents) of PhD student which is represented as 30 % and (21 respondents) of master student which is represented as 70 %. Moreover, the majority age of respondents presented 26.7% (8 respondents from 18-25), 56.7 % (17 respondents from 26 - 34) and 13.3 % (4 respondents from 35-44). With the observation that is one missing data have included in the age property.

Table 5. 1: Distribution of demographic data

Demographic Data	Frequency	Percentage (%)
Degree		
Phd	9	30
Master	21	70
Degree	0	0
Age		
18-25 Years old	8	26.7
26-34 Years old	17	56.7
35-44 Years old	4	13.3

Using the SPSS software, figure 5.1 and figure 5.2 describe the representation of both age and degree of the respondents as a Pie chart.

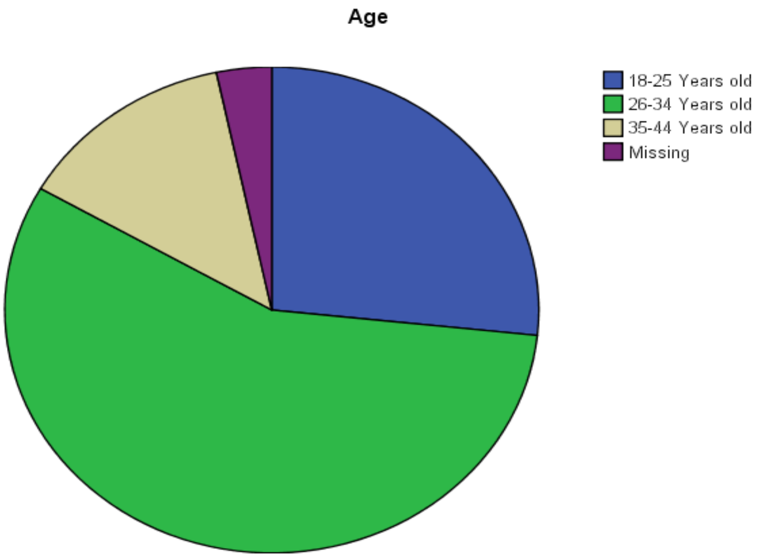

Figure 5.1: The distribution of the respondents' age as a pie chart

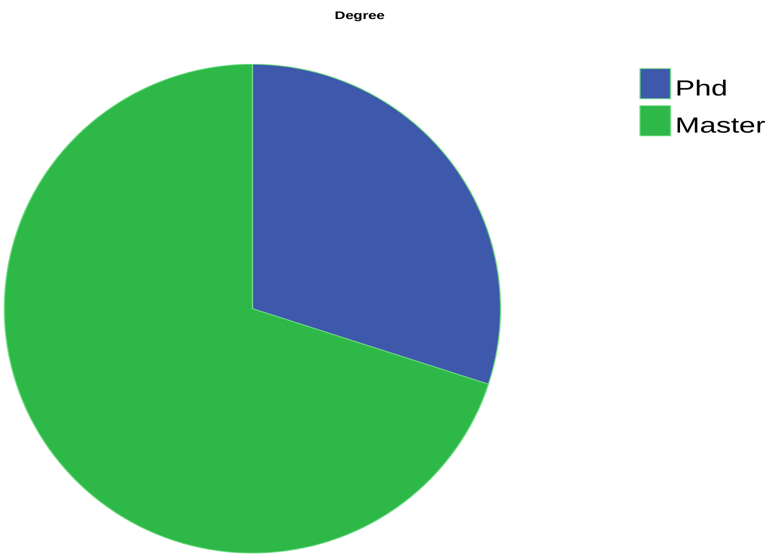

Figure 5.2: The distribution of the respondents' degree as a pie chart

5.2.1. Evaluation Of The Application Usefulness

Respondents were asked to indicate the usefulness of the application which includes several points such as delays of calls response. Table 5.3 shows the Mean and Std. Deviation of each question in the usefulness section that had been asked to the respondents. Then, by calculating the Mean value, the agreement of the application usefulness from the respondents can be seen easily.

Table 5.2: Descriptive statistics of application usefulness for all items

Item	Mean	Std. Deviation
1. The VoIP system is useful	4.40	.770
2. The delays of calls response are neglected.	4.23	.728
3. It saves my money when I use the VoIP system locally.	4.20	.714
4. The VoIP system provides the kind of content that i would expect to find inside UUM campus.	3.83	.913
5. The proposed VoIP system helps the UUM users to get better way of communication with each other.	4.63	.615
6. It does everything I would expect it to do.	4.30	.837
7. Using the VoIP system helps me to contact with the other users without Internet access.	4.33	.802

In terms of application usefulness, the analysis on respondents' background showed that the highest value of the mean is (4.63) and it means most of respondents agreed that the VoIP system helps the users to get better way of communication with each other. The table 5.3 Also shows the mean value of the first question which is (4.40) shows that VoIP system is useful. Table 5.4 shows that (26.7 %) of the respondents were strongly satisfied with browsing the system, also (36.7 %) of the respondents were just satisfied. While (30.7 %) of the respondents were Not Sure, and finally 6 % of the respondents were Strongly Disagree. Regarding the values of standard deviation as table 5.3 is shown the values of standard deviation for each question. The standard deviation value of the first question (.770) indicates that the data of this question tend to be close to the mean since it is the lowest values between the other values of standard deviation for the rest of the questions. Whereas, high standard deviation values indicate that the data are spread out over a large range of values. Figure 5.3 shows the bar graph of the distribution of the application usefulness for the question four.

Table 5.3: Distribution of question three in terms of application's usefulness

	Percent	Frequency
Disagree	6.7	2
Not Sure	30.0	9
Agree	36.7	11
Strongly Agree	26.7	8
Total	100.0	30

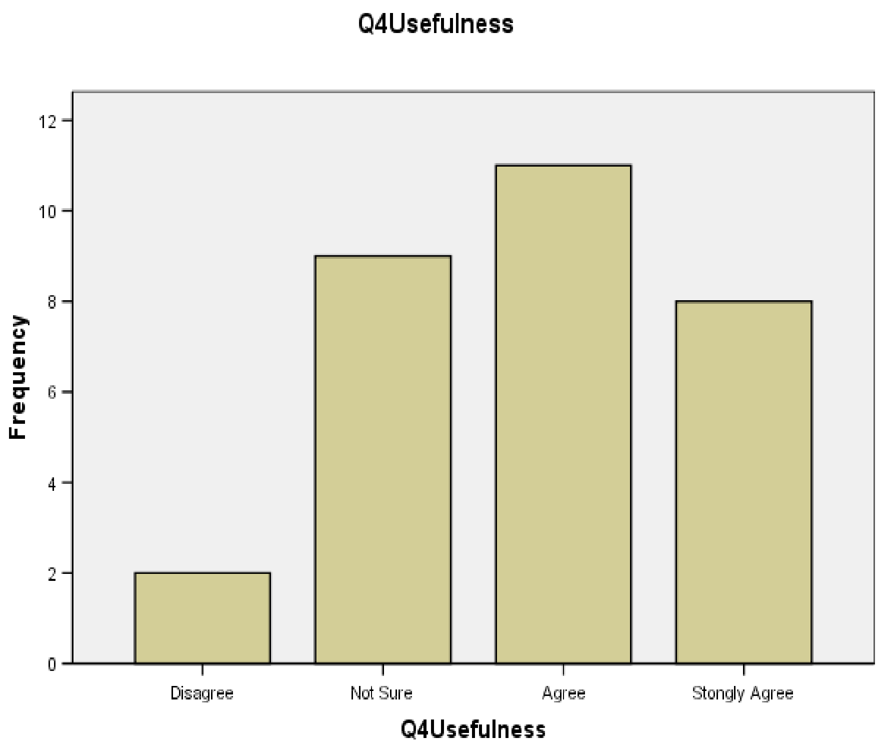

Figure 5.3 : Bar chart of the question four of the usefulness

5.2.2. Evaluation Of The Application Ease Of Use

In this step, the respondents were asked to indicate the ease of use of the application which includes several points such as easy to use the services provided by the VoIP system and the way to add new SIP account is easy. Table 5.5 shows the Mean and Std. Deviation of each question that had been asked to the respondents. By knowing these values, we can easily see the agreement of the application usability from the respondents.

Table 5.4: Descriptive statistics related to the application's ease of use

	Mean	**Std. Deviation**
1. The services provided by the VoIP system are easy to use.	4.50	.682
2. The way to add SIP account is easy.	4.33	.547
3. Register at Asterisk SIP server is simple.	4.13	.681
4. I need mental effort in order to understand the operations using the VoIP system.	2.17	.747
5. The way of using softphone programs is easy.	4.27	.740
6. All the actions and the operations in the VoIP system are controllable.	4.17	.747
7. It requires doesn’t require many steps to establish the connection between the users.	4.17	.699

Table 5.5 shows descriptive statistics for all the items in the Application ease of use distribution. The analysis on respondents' background showed that the highest value of the mean is (4.50) and it means, most of respondents agreed that the services provided by VoIP system are easy to use. Also the mean value of the second question which is (4.33) shows that also most respondents agreed that the way to add new SIP account is easy. The mean value of question four (2.17) shows that less than half of the respondents agreed that they need mental effort in order to understand and perform the operations using the system.

Table 5.6 shows that (60 %) of the respondents were strongly satisfied with browsing the system, also (30 %) of the respondents were just satisfied. While (10 %) of the respondents were Not Sure. Regarding the values of standard deviation as table 5.4 is shown the values of standard deviation for each question. The standard deviation value of the second question (.547) indicates that the data of this question tend to be close to the mean since it is the lowest values between the other values of standard deviation for the rest of the questions. Whereas, high standard deviation values indicate that the data are spread out over a large range of values. Figure 5.4 shows the bar graph of the distribution of the application ease of use for question one.

Table 5.5: Distribution of question one in terms of application's ease of use

	Frequency	Percent
Not Sure	3	10.0
Agree	9	30.0
Stongly Agree	18	60.0
Total	30	100.0

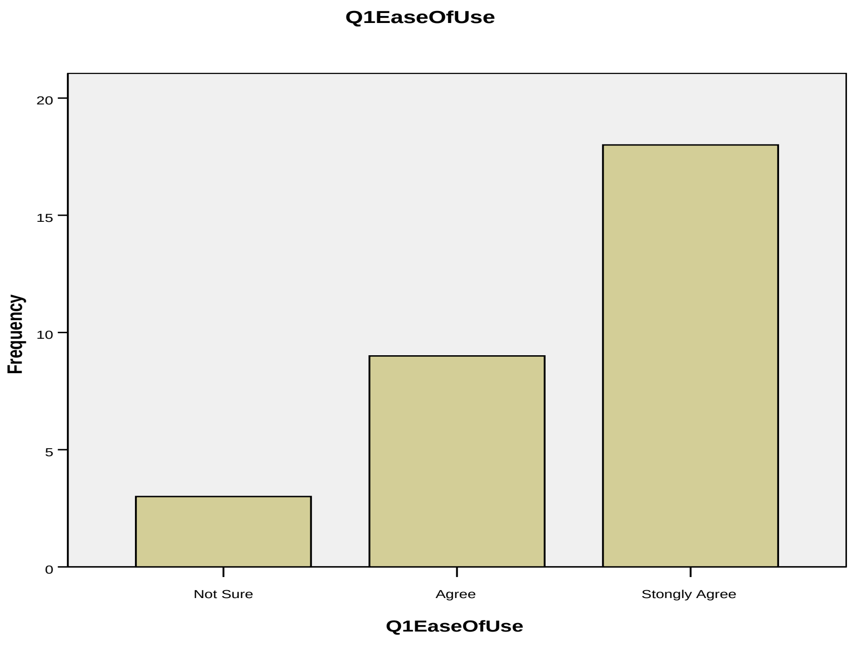

Figure 5.4: Bar chart of the question one of the application ease of use

5.2.3. Evaluation of the application satisfaction

Respondents were asked to indicate the satisfaction of the application which includes several points such as the time of responding, and quality of voice. Table 5.7 shows the Mean of each question that had been asked to the respondents and by the Mean and the St. Deviation we can easily know the agreement of the application's satisfaction from the respondents.

Table 5.6: Descriptive statistics related to the application's satisfaction

	Mean	**Std. Deviation**
1. The VoIP system responses to the user requests on time.	4.00	.871
2. The quality of voice in the proposed VoIP system is suitable.	4.10	.803
3. The VoIP services are available every time the users request it.	4.43	.774

Table 5.7 shows descriptive statistics for all the items in the Application satisfaction distribution. The analysis on respondents' background showed that the highest value of the mean is (4.43), it means that the VoIP services are available every time the users request it. Also the mean value of the first question (4.00) shows that most of respondents agreed that the VoIP system responses to the user requests on time.

Table 5.8 shows that (33.3 %) of the respondents were strongly agreed that the quality of voice in the proposed VoIP system is suitable, also (46.7 %) of the respondents were just agree. While, (16.7 %) of the respondents are Not Sure, and finally 3.3 % of the respondents were Strongly Disagree. Regarding the values of standard deviation as table 5.8 is shown the values of standard deviation for each question. The standard deviation value of the second question (.803) indicates that the data of this question tend to be close to the mean since it is the lowest values between the other values of standard deviation for the rest of the questions. Whereas, high standard deviation values indicate that the data are spread out over a large range of values. Finally, figure 5.6 shows the bar graph of the distribution of the application satisfaction for the question three.

Table 5.7: Distribution of question one in terms of application's satisfaction

	Frequency	Percent
Disagree	3.3	1
Not Sure	16.7	5
Agree	46.7	14
Stongly Agree	33.3	10
Total	100.0	30

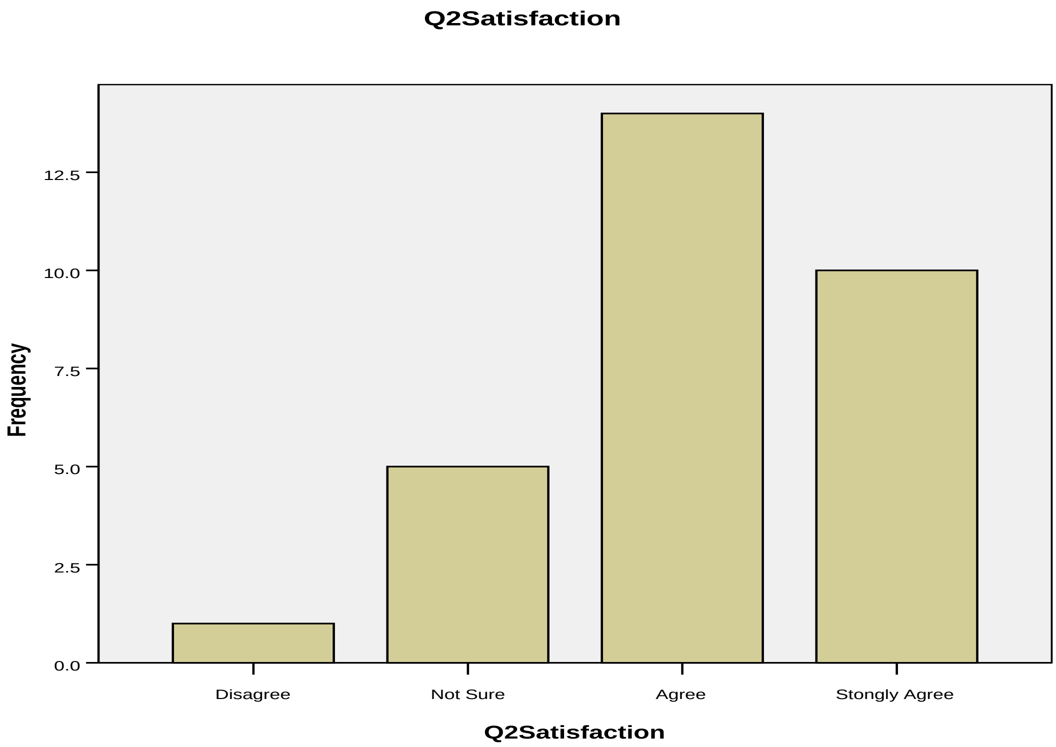

Figure 5.5: Bar chart of the question one of the application satisfaction

5.3. Summary

This chapter describes the evaluation process of the implementation of VoIP system in UUM campus. Questionnaire was distributed to 30 UUM residents. In addition, the evaluation process helped us more to understand the user's opinion, requirements and their expectations. Finally, and referring to the last objective of this proposed study which is the evaluation of the proposed VoIP system, we can say that this objective has been achieved successfully during the process of the evaluation that had discussed previously and also based on the very satisfied and acceptable results that we got at the end of this step.

CHAPTER SIX

CONCLUSION OF THE STUDY

6.0. Introduction

This chapter explains the conclusion of the study, in addition to include the study contribution for providing better communication services to the users at UUM campus using VoIP technique. The limitations of the research are also proposed in this chapter.

6.1. Project Conclusion

Based on identifying the limitations of the current communicating mechanisms used by UUM campus the result obtained at the end of this study meets the objectives identified in chapter one. Moreover, in this research study we tried to depend on our current available network equipment at UUM campus and create new network structure design using VoIP technology to complete the study successfully. Furthermore, implementing a VoIP system at UUM campus sounds a good idea to enhance the communicating mechanisms that let the different users at UUM to communicate with each other and with a low cost, using the current network equipment and with a convenient speed. Thus, the proposed VoIP system has been implemented and evaluated among a certain group of users and the results was quite satisfaction and acceptable.

6.2. Study Contributions

The proposed system is implemented based on VoIP technique, this technique used to transport voice communication over IP network. In addition, it provides the capability of making phone calls over the packet switched networks instead of traditional circuit switched networks. Thus, this technology is expected to become an important Internet application in the future. Some other strengths of the proposed system are listed as following:

i. Services provided to user

The structured of proposed system allows users to connect with each other regardless where ever the source and destination exist at UUM campus. The VoIP system presents to user many services such as voice call and text message. These services enable the user to make free calls and high voice quality.

ii. Services provided to UUM

This proposed VoIP system added several effective properties to the current UUM network. One of these services is reducing the usage of network bandwidth since there is no need to have Internet access in order to communicate with each other.

6.3. Limitation Of The Study

This study only concentrates on implementing of communication system using VoIP technique in order to enhance the current limitations of using the traditional communication mechanism at UUM campus. However, there are some limitations of the study as following:

i. In order to use the proposed VoIP system properly in any mobile device, we need to make sure that the mobile device support SIP protocol which is important to provide VoIP services.

ii. In terms of cost, any new service that we want to add to the VoIP system requires adding additional equipment to run this service smoothly.

iii. Not all VoIP services provide by VoIP system due to the limited time of doing the research and the lack of network equipment required to establish the services.

iv. The study doesn't cover deeply all the topics that are related to the VoIP technique because of also the limited time of doing the research.

6.4. Future Works

For better VoIP system, some future works can be formulated in order to improve the communication equipment and provide better services in terms of quality and effectiveness. One of the suggestions is to create a new softphone programs depends on the VoIP services which are existing in the system as following:

i. Voice call.

ii. Text message.

iii. Address book.

iv. Voice mail.

v. Fax.

vi. Video call.

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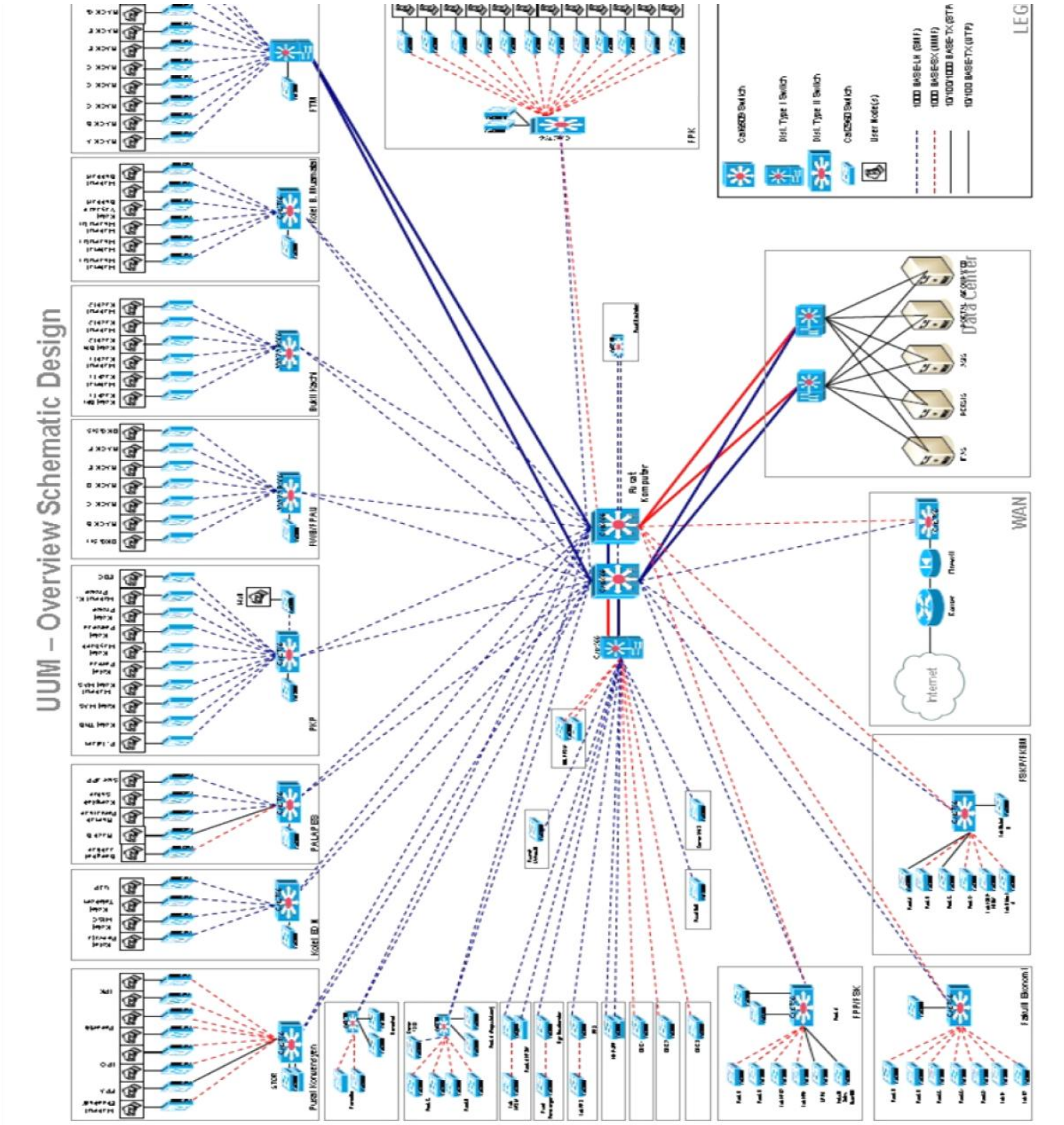
APPENDIX A

Universiti Utara Malaysia

College of Arts and Sciences



Fiqure: UUM Overview Schematic Design

APPENDIX B

Universiti Utara Malaysia

College of Arts and Sciences

Asterisk Installation

The commands lines of installing compilers packages are as the following:

- The command for GCC and G++ is:
    - `#apt-get install gcc g++.`
- The command for ncurses is:
    - `# apt-get install ncurses-base libncurses-dev.`
- The command for newt is:
    - `# apt-get install libnewt0.52 libnewt-dev.`
- The command for libxml2 is:
    - `# apt-get libxml2 libxml2-dev.`

Some of these packages may need to be installed as we can see in figure 1, and some of these packages don't need to be installed as you can see in figure 2.

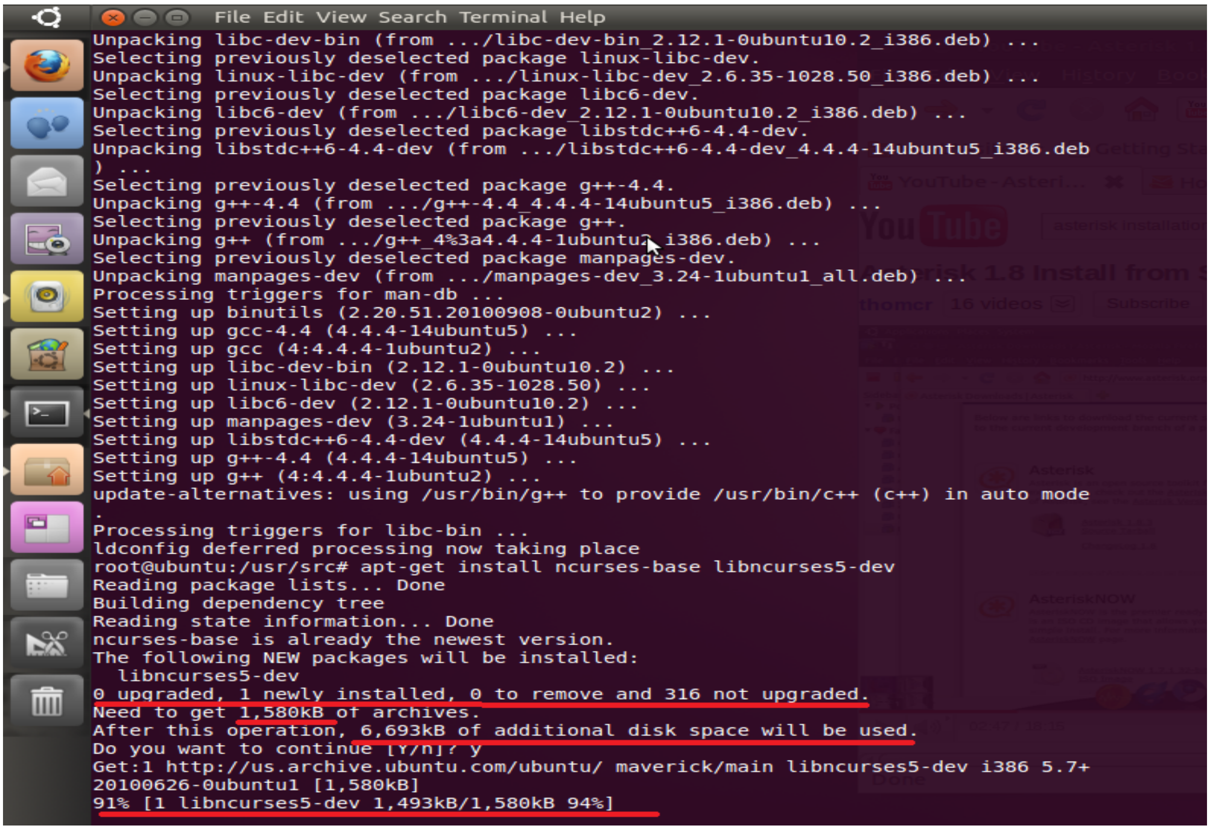


Figure 1: Installing ncurses compiler

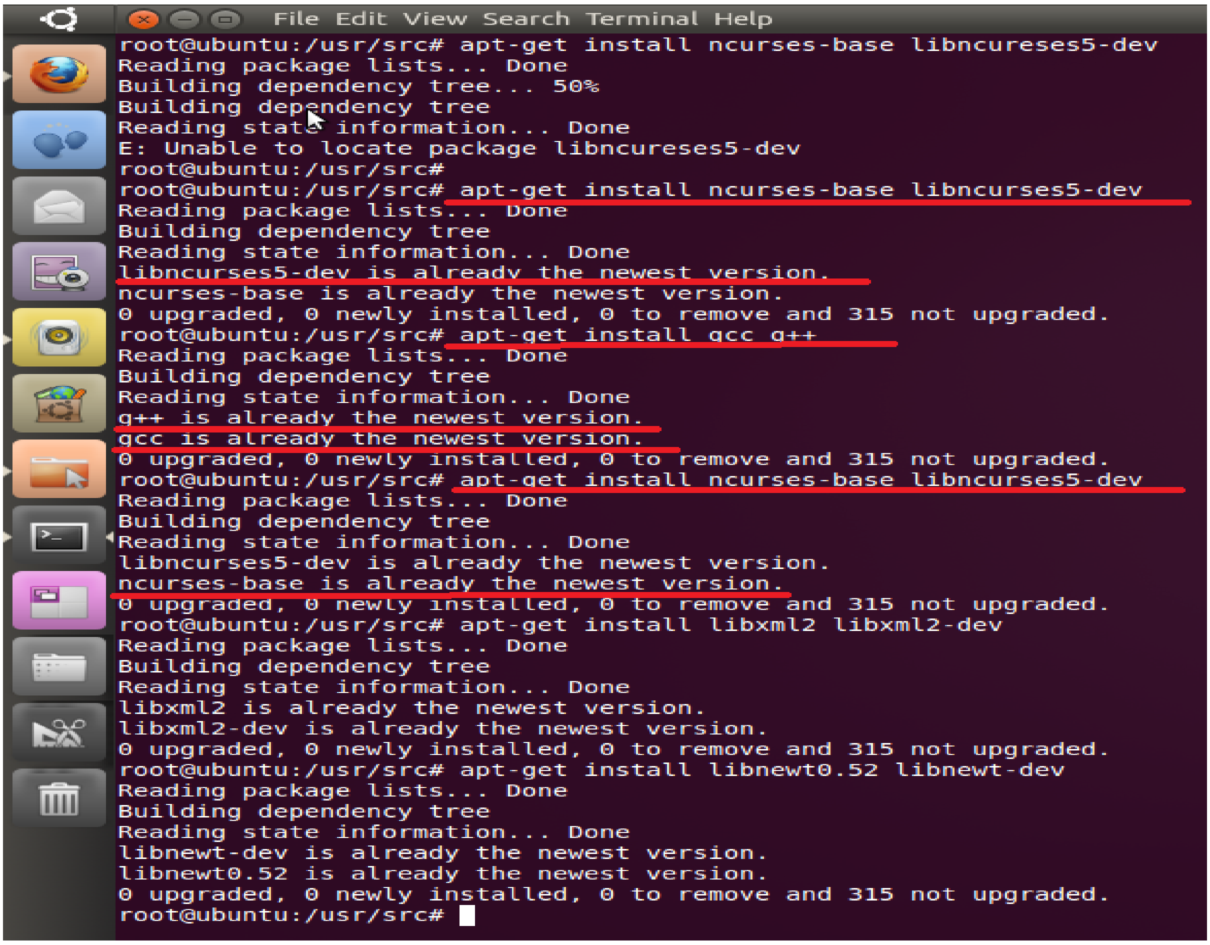


Figure 2: Check existing package

Directory /usr /src / will be used to download Asterisk source code, while admin is using the command /wget /admin can release, get the package and enter the following commands line as we can see at figure 3:

- `#cd /usr/src/`
- `#wgethttp://downloads.asterisk.org/pub/telephony/asterisk/asterisk-1.8.4.tar.gz.`

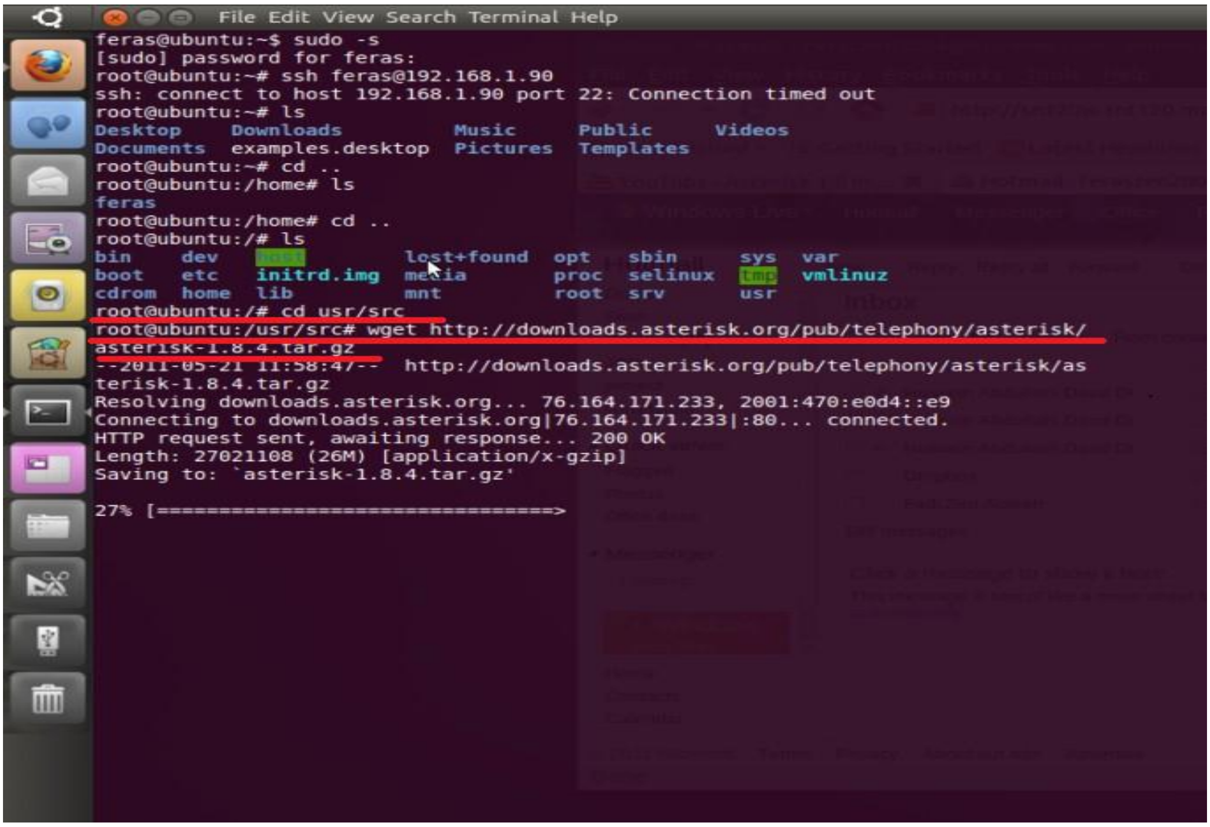


Figure 3: Download Asterisk package

Before admin can use the downloaded source code packages via FTP, admin need to extract the compressed archives files which containing the source code; thus, admin will need to extract them before compiling. This is a simple operation that can be achieved by using of the following commands:

- `#cd /usr/src/`
- `#tar zxvf asterisk-1.8.4-current.tar.gz`

These commands will extract the packages and source code to their respective directories. When admin extract the asterisk-1.8.4-current.tar.gz file, you will find that the file will extract to the current version of Asterisk, i.e. asterisk-1.8.4.

Using Menuselect

Moving between options is by navigated up and down the list using the arrow keys, and then we should select one of the menu options by pressing Enter or by using the right arrow key. The left arrow key can be used to go back. Modules to be built are marked as [*] while [ ] refers to the modules that are not being built yet. Modules that have XXX in front of them are missing a package dependency which must be satisfied before being available to be built (Van Meggelen, Madsen, et al. 2007). In Figure 4.4, we can see that the app_mysql module which cannot be built due to the missing of the dependency of Asterisk (i.e., the Asterisk module has not been built and installed on the system since the last time ./configure was run). If admin have satisfied a dependency since the last time admin ran ./configure, then run it again, and run menu select again. Asterisk system now should be available for building. After you have finished making changes to the menu select, there are two choices either save & exit or exit (the changes may be lost!).

Executed Command (sip show peer)

More detailed stats of the peer can be shown as follows with sip show peer 1000:

```
*CLI>sip show peer 1000
* Name: 1000
Secret :<Not set>
MD5Secret :<Not set>
Context: phones
Subscr.Cont. :<Not set>
Language:
AMA flags: Unknown
Transfer mode: open
CallingPres: Presentation Allowed, Not Screened
Callgroup:
```

```
Pickupgroup:
Mailbox:
VM Extension: asterisk
LastMsgsSent: 32767/65535
Call limit: 0
Dynamic: Yes
Callerid: "" <>
MaxCallBR: 384 kbps
Expire: 1032
Insecure: no
Nat: RFC3581
ACL: No
T38 ptUDPTL: No
CanReinvite: Yes
PromiscRedir: No
User=Phone: No
Video Support: No
Trust RPID: No
Send RPID: No
Subscriptions: Yes
Overlap dial: Yes
DTMFmode : rfc2833
LastMsg: 0
ToHost :
Addr->IP : 192.168.1.250 Port 5061
Defaddr->IP : 0.0.0.0 Port 5060
Def. Username: 1000
SIP Options: (none)
Codecs: 0x8000e (gsm|ulaw|alaw|h263)
Codec Order: (none)
Auto-Framing: No
Status: Unmonitored
Useragent: X-Lite release 1105d
Reg. Contact: sip:1000@192.168.1.250:5061
```

Basics Dialplan

The dial plan is truly the core of any Asterisk system, as it defines how Asterisk handles inbound and outbound calls. Unlike traditional phone systems, Asterisk's dial plan is fully different. The Asterisk dial plan is specified in the configuration file named extensions.conf. The dial plan is made up of four main concepts: contexts, extensions, priorities, and applications. In the next few sections, there is explanation about these concepts which work together

- **Contexts**

Dial plans are broken into sections called contexts. Contexts are named groups of extensions, which serve several purposes. Contexts keep different parts of the dial plan from interacting with one another. An extension that is defined in one context is completely isolated from extensions in any other context, unless interaction is specifically allowed. Contexts are denoted by placing the name of the context inside square brackets ([ ]). The name can be made up of the letters A through Z (upper- and lowercase), the numbers 0 through 9, the hyphen and underscore.* For example, a context for incoming calls looks like this:

[incoming]

All of the instructions placed after a context definition is part of that context, until the next context is defined. At the beginning of the dial plan, there are two special contexts named [general] and [globals]. The [general] section has a list of general dial plan settings, "Global variables" section; these two contexts are special. As long as admin avoid the names [general] and [globals], admin may name contexts anything he/she likes.

- **Extensions**

In Asterisk, however, an extension is far more powerful, as it defines a unique series of steps (each step containing an application) that Asterisk will take that call through. Within each context, we can define many (or few) extensions as required. When a particular extension is triggered (by an incoming call or by digits being dialed on a channel), Asterisk will follow the steps defined for that extension. It is the extensions, therefore, that specify what happens to calls as they make their way through the dial plan. Although extensions can certainly be used to specify phone extensions in the traditional sense (i.e., extension 153 will cause the SIP telephone set on John's desk to ring), in an Asterisk dial plan, they can be used for much more. The syntax for an extension is the word extension, followed by an arrow formed by the equals sign and the greater-than sign, like this:

exten =>

This is followed by the name (or number) of the extension. When we are dealing with traditional telephone systems, we tend to think of extensions as the numbers you would dial to make another phone ring. A complete extension is composed of three components:

• The name (or number) of the extension.

• The priority (each extension can include multiple steps; the step number is called the "priority").

• The application (or command) that performs some action on the call.

These three components are separated by commas, like this:

```
exten =>name,priority,application()
```

Here's a simple example of what a real extension might look like:

```
exten => 123,1,Answer()
```

In this example, the extension name is 123, the priority is 1, and the application is Answer(). Now, let's move ahead and explain priorities and applications.

- **Priorities**

Each extension can have multiple steps, called priorities. Each priority is numbered sequentially, starting with 1, and executes one specific application. As an example, the following extension would answer the phone (in priority number 1), and then hang it up (in priority number 2) :

```
exten => 123,1,Answer()
exten => 123,2,Hangup()
```

The key point to remember here is that for a particular extension, Asterisk follows the priorities in order.

- **Unnumbered Priorities**

In older releases of Asterisk, the numbering of priorities caused a lot of problems. Imagine that we have an extension that had 15 priorities, and then we need to add something in step 2. All of the subsequent priorities would have to be manually renumbered. Asterisk does not handle missing steps or miss numbered priorities, and debugging the types of errors was pointless and frustrating. Beginning with version 1.2, Asterisk addressed this problem. It introduced the use of the n priority, which stands for "next". Each time Asterisk encounters a priority named n, it takes the number of the previous priority and adds 1. This makes it easier to make changes to the dial plan, as admin don't have to keep renumbering all his/her steps. For example, the dial plan might look something like this:

```
exten => 123,1,Answer()
exten => 123,n,do something
```

```
exten => 123,n,do something else
exten => 123,n,do one last thing
exten => 123,n,Hangup()
```

Internally, Asterisk will calculate the next priority number every time it encounters an n. You should notice, that you must always specify priority number 1. If accidentally put an n instead of 1 for the first priority, admin'll find that the extension will not be available.

- **Priority Labels**

Starting with Asterisk version 1.2 and higher, common practice is to assign text labels to priorities. This is to ensure that admin can refer to a priority by something other than its number, which probably unknown, given that dial plans now generally use unnumbered priorities. To assign a text label to a priority, simply add the label inside parentheses after the priority, like this:

```
exten => 123,n(label),application()
```

- **Applications**

Applications are the workhorses of the dial plan. Each application performs a specification on the current channel, such as playing a sound, accepting touch-tone input, dialing a channel, hanging up the call, and so forth. In the previous example, you were introduced to two simple applications: Answer() and Hangup(). Some applications, such as Answer()and Hangup() do not need other instructions to do the jobs. Other applications require additional information. These pieces of information, called arguments, can be passed on to the applications to affect how they perform the interactions. To pass arguments to an application, place them between the parentheses that follow the application name, separated by commas.

APPENDIX C

Universiti Utara Malaysia

College of Arts and Sciences

Implementation of Voice over Internet Protocol (VoIP) in UUM Campus

SECTION A: Demographic Background

Please kindly tick (√) your answers to the given statements

Degree:

☐ PhD

☐ Master

☐ Degree

AGE :

☐ 18-25 Years old. ☐ 26-34 Years old. ☐ 35-44 Years old.

☐ 44-54 Years old. ☐ Above 55 Years old.

SECTION B: Visual aspects of the application usefulness

Please check the appropriate column. The numbers 1 to 5 represent the following:

1= Strongly Disagree. 2= Disagree. 3= Not Sure. 4= Agree. 5= Strongly Agree.

	Question	1	2	3	4	5
Q1	The VoIP system is useful	☐	☐	☐	☐	☐
Q2	The delays of calls response are neglected.	☐	☐	☐	☐	☐
Q3	It saves my money when I use the VoIP system locally.	☐	☐	☐	☐	☐
Q4	The VoIP system provides the kind of content that i would expect to find inside UUM campus.	☐	☐	☐	☐	☐
Q5	The proposed VoIP system helps the UUM users to get better way of communication with each other.	☐	☐	☐	☐	☐
Q6	It does everything I would expect it to do.	☐	☐	☐	☐	☐
Q7	Using the VoIP system helps me to contact with the other users without Internet access.	☐	☐	☐	☐	☐

SECTION C: Visual aspects of the application ease of use

Please check the appropriate column. The numbers 1 to 5 represent the following:

1= Strongly Disagree. 2= Disagree. 3= Not Sure. 4= Agree. 5= Strongly Agree.

Question		1	2	3	4	5
Q1	The services provided by the VoIP system are easy to use.	☐	☐	☐	☐	☐
Q2	The way to add SIP account is easy.	☐	☐	☐	☐	☐
Q3	Register at Asterisk SIP server is simple.	☐	☐	☐	☐	☐
Q4	I need mental effort in order to understand the operations using the VoIP system.	☐	☐	☐	☐	☐
Q5	The way of using softphone programs is easy.	☐	☐	☐	☐	☐
Q6	All the actions and the operations in the VoIP system are controllable.	☐	☐	☐	☐	☐
Q7	It requires doesn’t require many steps to establish the connection between the users.	☐	☐	☐	☐	☐

SECTION D: Visual aspects of the application satisfaction

Please check the appropriate column. The numbers 1 to 5 represent the following:

1= Strongly Disagree. 2= Disagree. 3= Not Sure. 4= Agree. 5= Strongly Agree.

	Question	1	2	3	4	5
Q1	The VoIP system responses to the user requests on time.	☐	☐	☐	☐	☐
Q2	The quality of voice in the proposed VoIP system is suitable.	☐	☐	☐	☐	☐
Q3	The VoIP services are available every time the users request it.	☐	☐	☐	☐	☐

THANKS A LOT