

**COMMUNICATION LINKING  
UTP VOICEMAIL SYSTEM ADD-ON**

By

**CHAN PISETH**

Dissertation submitted in partial fulfillment of  
the requirement for the  
Bachelor of Technology (Hons)  
(Information and Communication Technology)

JANUARY 2011

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**CERTIFICATION OF APPROVAL**

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Approved by,

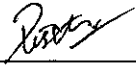


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TRONOH, PERAK  
JANUARY 2011**

## 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 has not been undertaken or done by unspecified sources or persons.



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CHAN PIETH

## **ABSTRACT**

This paper is intended to present author's Final Year Project (FYP), Communication Linking: Voicemail System Add-on (VMSA) for small and medium size company. The project dissertation explains in detail about proposed prototype, "Universiti Teknologi PETRONAS (UTP) Voicemail System", which is implemented based on VMSA to enhance current traditional phone extension in UTP. It shows the importance of this system in UTP; how it benefits both students and lecturers in UTP. Moreover, this paper will discuss the built-in features and tools to assist user accessing their voicemail more effectively and efficiently. Besides describing the system in general, it will discuss all VMSA features to be added to the "UTP Voicemail System" such as local area network (LAN) call, voicemail capability with E-mail and SMS notification, voice-prompt with text to speech tool, phone number filtering, and Voicemail Graphical User Interface (VMGUI). Furthermore, it will introduce "Asterisk" open-source Private Branch Exchange (PBX) software for handling voice communication and voice recording and some other open-source software. Lastly, it will explain result of each development phase of the prototype and some recommendations for future improvement.

## **ACKNOWLEDGEMENT**

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# CHAPTER 1

## INTRODUCTION

### 1.1. BACKGROUND OF STUDY

Nowadays, with the advancement of technology such as Internet, and Network; with the assistant of open source, many communication means such as E-mail, Voice over Internet Protocol (VoIP), Short Message Service (SMS), Multimedia Messaging Service (MMS), Voicemail service and etc. have been implemented to improve communication in an organization.

Voicemail system, known as answering machine, is one the communication means which is used as centralized electronic system to store messages from telephone callers and it can be retrieved back by its users [1]. It was first introduced with the name Telegraphone, a magnetic recorder, by Danish telephone engineer and inventor, Valdemar Poulsen in 1898. At that time, it had been used not only as voicemail system, but also as answering machine, scientific instrument as well as office dictating system.

In the late 1920s, a new approach had been used by German companies AEG and I.G. Farben to improve the efficiency of recording by using recording tape with a paper strip coated and magnetic iron powder rather than using steal recording tape. And around 1932, those two companies introduced a new recorder “magnetophon” [2]. However, the term “Voice mail” or “Voicemail” was not using widely until it was introduced and patented by Gordon Matthews who was known as the “Father of Voice Mail” and founder of the VMX Company in Dallas, Texas [3].

Voicemail system using recording tapes to store the voicemail fades away or getting less popular. They are now digitized and stored in the form of data file in a server which has

overcome the limitation of number of voicemail for each user. Moreover, each user can have more than one voicemail box.

As Internet was widely used with its affordable price, voicemail system is transformed and combines with VoIP to provide not only message storing and retrieving capability but also a surprisingly low cost calling capability.

Initially, the purpose of voicemail system and service is to increase production in business by reducing the number of missing customer calls and providing a way to handle the call when user is not around or in busy mode. However, voicemail system is not applied only in business, but also in other areas such as:

- Campus/ College/ University
- Government sector
- Hospital
- Contact Center

The proposed VMSA offers additional features to current traditional phone system. For instance, UTP voicemail system which is a prototype implemented with VMSA plays an important role to close communication gap between students and lecturers. It provides either traditional way of accessing voicemail whereby lecturers call to their voicemail main menu extension to listen to their voicemail or using Voicemail Graphical User Interface (VMGUI) whereby they can listen and retrieve their voice message anywhere and anytime from the server through web interface. Moreover, lectures have a control over some customizable features which are provided by VMSA such as E-mail and SMS notification, phone number filtering, and voice-prompt with text to speech tool.

## 1.2. PROBLEM STATEMENTS

### 1.2.1. PROBLEM IDENTIFICATION

The limitation of traditional voicemail system, which requires users to go through a long list of menus to perform specific tasks such as listening to new voicemail, deleting voicemail, and etc, makes voicemail system become less user-friendly and inadequate. In addition, whenever the extension is busy or unavailable, the call will be routed to voicemail server without any indication to the caller when user is able to receive the call.

There are a lot of issues on the ways to improve communication between people and people. For instance, let's look at the problems that are faced by lecturers and students in UTP. It is good that each lecturer's office is equipped with a telephone extension as one of communication means to help students and lecturers communicating with each other. But is it sufficient?

This is a challenge question to the author. But before answering this question, let's look at some aspects below:

- *Have you ever faced an unanswered call when you are trying to make an appointment with lecturers?*

Yes, of course. Imaging that student A made a call to lecturer to schedule a meeting for class project and at the same time there was student B making another call to schedule for FYP discussion. While lecturer was entertaining student A, what would happen to student B? With current phone system implemented in UTP, it was clear that the call from student B was dropped or would route to lecture's voicemail server. So, student B either needed to wait until student A finished the call, or he needed to call in later, or he needed to leave a voicemail. Figure 1.1 shows the process of a call terminated when the channel was occupied. In this case, student A and lecturer occupied the channel so when student B tried in, the call was dropped.

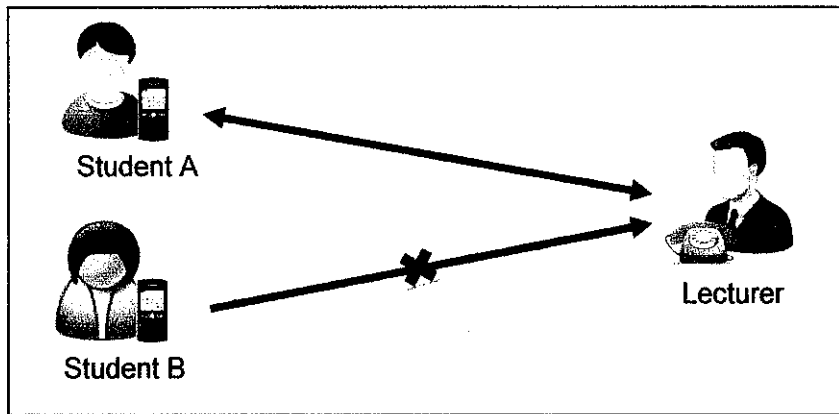


Figure 1.1: Call termination on busy channel

- *How can lecturers know if there are any new voicemail in his voicemail box when they are not around?*

With current system, lecturers can identify new voicemail only by browsing through their phone extension in their office or sometimes there will be SMS alert to their mobile phone. Thus, even though they receive a notification, they cannot listen to their voicemail if they are not in the office. For SMS notification, the mobile phone number which was used for receiving SMS notification must be set by administrator. Let's imagine that, if the phone number is changed or lost what would happen? Lecturer need to contact directly with administrator to change for them.

- *How can lecturers retrieve or listen to their voicemail if they are away from office?*  
To retrieve or listen to the voicemail message they must access from their phone extension in the office. Thus, if they were on field work, some important information would be missing because they had to wait until they came back to their office to retrieve their voicemail.

After looking at above aspect, current system needs to be improved. Thus, the proposed VMSA comes in handy.

## 1.2.2. SIGNIFICANT OF THE PROJECT

The proposed VMSA is designed to provide additional features to voicemail system. For instance, in UTP, if this system is approved to be implemented, it will help lecturers and students improve their Communication as following:

For lecturer:

- Record all incoming voicemail information and are able to display it anywhere through VMGUI.
- Record and update greeting voice-prompt.
- Alert or notify to or of new voicemail through E-mail and SMS.
- Access voicemail message anywhere through VMGUI.
- Filtering unnecessary phone number (Blacklist the number)

For student:

- Call from dormitory to lecturer extension by softphone
- Leave voicemail message when no one answers the call or when it is busy.
- Receive the voice-prompt notification
- Reduce time to go all the way to academy building to make appointment with lecturer.

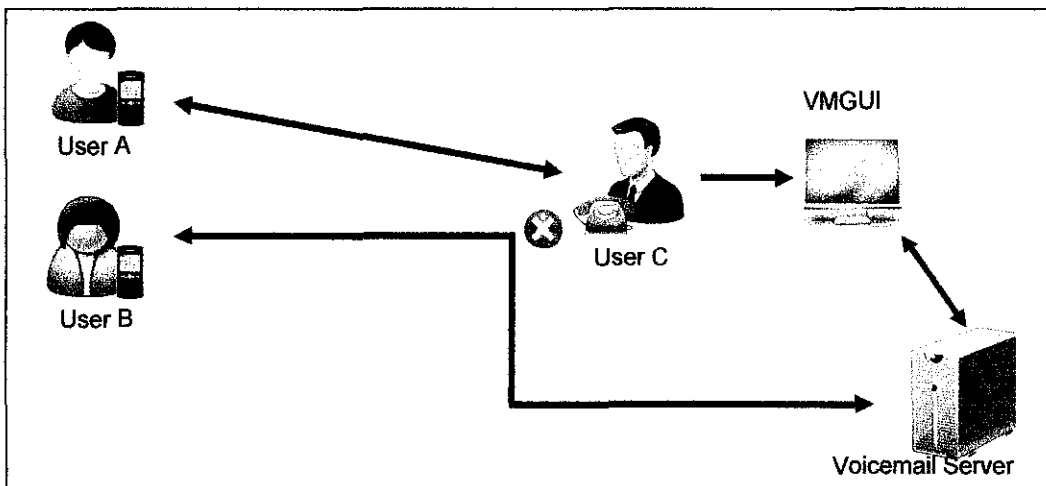


Figure 1.2: Voicemail server handles a redirected call on busy channel and VMGUI user accessibility

Figure 1.2 shows that User A and User C occupy the channel and at the same time User B calls in. Then User B is redirected to voicemail server to leave the voicemail for User C. On the other hand, after finishing the call, User C checks call history to observe if anyone leaves any messages in his voicemail box through VMGUI.

### **1.3. OBJECTIVES AND SCOPE OF STUDY**

#### **1.3.1 OBJECTIVES**

The main objectives of this project are:

- To develop a cost effective system in term of operation and usage  
(See Section 2.3 for more information)
- To implement Add-on for phone system for optimizing the communication which provides following features
  - Local Area Network Call (VoIP)
  - Voicemail capability with notification such as Email and SMS
  - Phone number filtering
  - Voice-prompt notification with text to speech tool
  - Voicemail Graphical User Interface (VMGUI)

#### **1.3.2 SCOPE OF STUDY**

The overall system focuses mainly on designing and implementing an add-on package that provides some enhanced new features to traditional telephone, or Private Branch Exchange (PBX) system. As mentioned in the objectives, there are five main features which are going to be implemented, so our research will target on how to design and build those features efficiently and effectively.

First and second features will be hosted on an open source software “Asterisk” which has an ability to transform a normal PC and server to a voice communicate server [4]. It requires some additional hardware to be added; some configuration must be made in order to interface VMSA with traditional telephone line and VoIP gateway which will not cover in this scope.

For the other features, they will be designed based on available free tools and other open sources available; some necessary components will be assembled to establish a user friendly GUI which can be accessed anywhere.



## **CHAPTER 2**

### **LITERATURE REVIEW**

#### **2.1 INTRODUCTION**

In this chapter, the author will review on some related technologies to be used to develop the system. On top of that, there will be some discussion in detail on how to achieve the objectives of the project with support from some published paper from various authors.

Most of telephone system and voicemail system is still implemented with traditional PBX (Private Branch Exchange) which only provides routing functionality and basic voicemail capability. However, based on research, there are several voicemail systems which provide hosting service on the cloud. They are designed, developed, and commercialized in the market such as Voicemail Interface System and Method (E. Vallier) [5], [www.messagingservice.com](http://www.messagingservice.com), [www.multisuns.com](http://www.multisuns.com), Switchvox [6] and etc. All of those are commercial products which can be purchased or paid on demand basis. Besides, some researches, studies and projects have been carried out and implemented to improve voicemail system for the organization.

“All Talk and All Action: Strategies for Managing Voicemail Messages” is a studies on user experience with voicemail system which stated the usage problem and processing strategies. In this study, users described three main problems in processing voicemail which were scanning the inbox sequentially, extracting information from message, and searching the archive of stored messages. Furthermore, it suggested that redesigning of the system which provides more flexible tools or technologies for users to access their voicemail message is needed [7].

“Automated Message Prioritization: Making Voicemail Retrieval More Efficient” is a project which focuses on developing concepts and machine language algorithm to ease the process of retrieving the voicemail message. It has the ability to retrieve the voicemail message from voicemail box and prioritize it in urgency manner and business Relevant [8].

However, there are a few open source projects such as freeswitch [9], OpenSIP, Asterisk, and etc. which allow developer to implement in-house voicemail system for free with customizable features. In the development of this proposed system, Asterisk is used as a main tool to establish the voicemail server. There are some introduction on some technologies and process helping to successfully implement this proposed VMSA.

## **2.2 RELEVANT TECHNOLOGY**

### **2.2.1 WHAT IS ASTERISK?**

Asterisk is an open-source PBX with VoIP capabilities which is primarily hosted on Linux. It allows us creating a custom phone system to extend the functionality of traditional phone system [10]. Asterisk can be customized to work as a PBX, VoIP System, IVR (Interact Voice Response) System, and Voicemail system. It enables author to use the existing UTP LAN (Local Area Network) to provide a point to point call or PC-to-PC call with the help of SIP (Session Initiation Protocol) protocol, or IAX (Inter-Asterisk eXchange) protocol, or other popular protocols.

Asterisk allows connecting to PSTN (Public Switched Telephone Network) and VoIP gateway to provide direct long distance call with low cost. On top of that, Asterisk can be interconnected with other Asterisk as well as other SIP gateway. Thus, users from different server or office from the other side of the world can interconnect with each other easily.

Asterisk is a fully functional system which supports voicemail context so that multiple departments or organizations can access their voicemail from the same server and one user can have more than one voicemail box with unlimited space. Furthermore, it also supports email notification with voicemail attachment [11].

Asterisk is developed in the environment which supports most of popular DBMS (Database Management System) like PostgreSQL, MySQL, MSSQL through ODBC, and etc.

### 2.2.2 WHAT IS SIP?

SIP (Session Initiation Protocol) basically is an application level signaling protocol for initiating interactive communication session in an IP network. It is a text-based protocol quite similar to SMTP and HTTP which consists of message headers and a message body. SIP is a better fit protocol for IP PBX because it helps to create, modify, and terminate communication channel. SIP has the ability to invite its users to existing session such as conferences or call queue.

SIP basically sends message through UDP on port 5060. As SIP is only capable of making connection session, SIP uses Real-Time Protocol (RTP) to provide transportation and quality of service (QoS) feedback for streaming voice media [12].

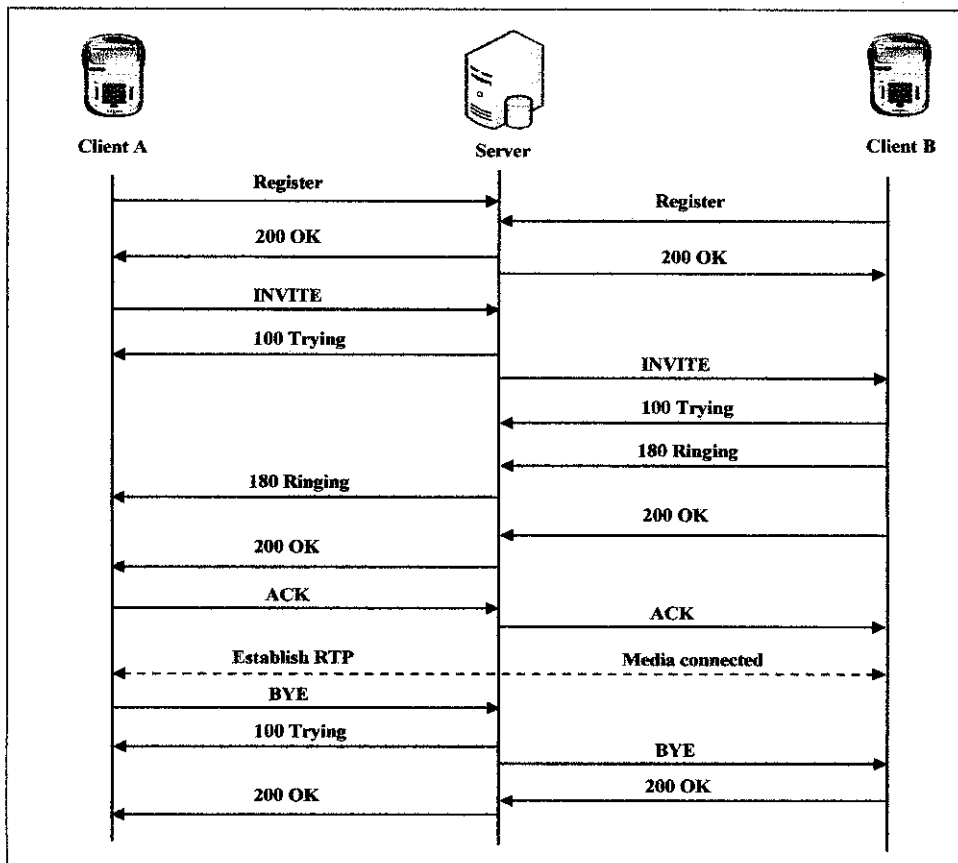


Figure 2.1: SIP requests, acceptances, setup, and terminates call diagram

Figure 2.1 shows how SIP sends request message, accepts request, setups session, and terminates the session. First, both Client A and B register to SIP server so that SIP server can locate where its clients are. Second, Client A sends request to initiate a call to Client B. SIP server locate where Client B then send request to Client B. Ring sound is generated before acceptance message is forwarded back to Client A. The session between Client A and B with RTP voice setups after acknowledgement from Client A is sent to Client B. Third, when terminating message is sent to Client B, Client B will send acceptance message back to Client A then the session will be destroyed.

### **2.3 HOW DOES IT REDUCE COST?**

There are a few factors affecting implementation cost of the proposed VMSA compared to the traditional voicemail system such as operational cost, technical support cost, and usage cost.

To add new extension for traditional PBX system, it will require a complex connection and the redesigning of network for that extension which would be considered costly. On the other hand, the proposed system can reduce all hard works while it uses existing LAN to connect to the new extension. As a result, the growth of users is not a big deal.

Upgrading voicemail feature for traditional PBX system will require additional costly hardware and technical support while the proposed VMSA uses Asterisk running as a core server handling all configuration and customization with a one-time hardware purchase. For instance, the cost of hardware to connect to PSTN line or ISDN (T1/E1) line for Asterisk is affordable. Most of PBX functionality could be configured in the configuration file or Database. Thus, it can be customized to meet its user's needs.

Cost of usage can be viewed as network cost versus telephone service cost. This system makes use of LAN and internet to provide a free PC-to-PC call between users. For instance, UTP voicemail system with VMSA allows students to make a call to lecturers whenever they have access to Campus's LAN and where Internet connection is accessible. On top of that, the system makes use of VoIP gateway to initiate the call to outside world which helps reducing the telephone service bill.

## **CHAPTER 3**

### **METHODOLOGY**

#### **3.1 PROCEDURE IDENTIFICATION**

This proposed project will be divided into two sub-methodologies, Research Methodology and Development Methodology. Research methodology basically focuses on collecting useful information from all parties related to the development of the system. On the other hands, design methodology discusses the methods used for implementing the proposed system.

#### **3.2 RESEARCH METHODOLOGY**

##### **3.2.1 INTERVIEW**

First of all, an interview session was conducted with Mr. Ruslan, Telecommunication Lab Technician, to collect some information regarding equipment in VoIP lab and the process to implement the existing VoIP in UTP. The objective of this interview is to gain a better understanding of the process flow of the VoIP and to assess whether the new system could make use of the equipment from VoIP lab to provide additional functionalities.

##### **3.2.2 QUESTIONNAIRE**

Questionnaire is chosen as a tool to gather information from users on the usage of all technologies which closely related to the proposed project, the way they interact, and how they comment on the current system compare to previous system. The main reason for the usage of questionnaire is because it is an inexpensive way to collect data in large scale. It also saves a lot of time because it is easily distributed both with hard copy and online. In addition, data interpretation is much easier and faster.

This questionnaire is designed so that results will be fair and distributed. In addition, it will also be supported with observation and survey to assess the performance of the system.

### 3.3 DEVELOPMENT METHODOLOGY

There are many development methodologies available for the development of this project. However, the best methodology for this project is Phase Development Methodology. The main reason of choosing this methodology is because of time constraint and the overall system requirement and concept are identified.

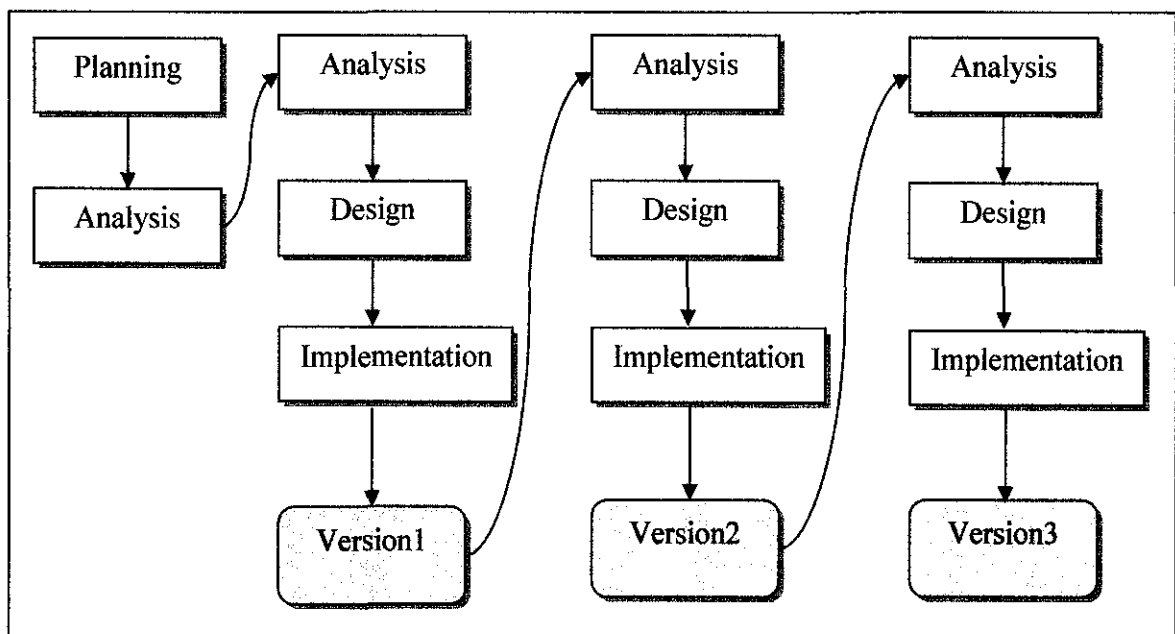


Figure3.1: Phase Development-based Methodology

Major phases and deliverables:

- Project initiation and proposal
  - a. Conduct further research on topic to be proposed for FYP topic and on its feasibility
  - b. Topic proposal is the task which is initiated first two week after new semester starts. The proposed FYP topic was sent to FYP committee and lecturer for approval.
  - c. Preliminary report included background, problem identification, significant of the project, objectives, and scopes of study of the proposed project.

- d. Progress report included some studies related to proposed project such as literature review, methodology, result and discussion, and conclusion.
- Planning and analysis
  - a. Identify problem and opportunity will be done in planning phase to find out the problem with the current system and to determine how to solve that problem.
  - b. Identify available equipment and technology will be done in planning phase. This phase helped to determine required equipments for the propose system.
  - c. Get user requirement and system requirement will be done in analysis phase in every version of the system to review user requirement in each version of the system.
  - d. Refinement the requirement will be done in every version of the system
- Design phase
  - a. System architecture design will be defined in first version of the system
  - b. UML diagram design will be defined in first version of the system
  - c. Database design will be defined in second and third version of the system
- Implementation and system version
  - a. Server configuration and backup will be done for every version of system to add new features to the system and to make sure the system can be rolled back and recovered to a stable stage.
  - b. Develop GUI for retrieving voicemail will be implemented in version three
  - c. End user testing will be done for every version of the system to determine the usability of the system.

The project will be divided into three major versions whereby version one will be implemented in FYP1 and other two major versions will be implemented in FYP2. The purpose of this division is to make sure fast delivery of the system to end user for testing.

Following are the key functionalities and deliverables in each version:

*Version one*

- Server Installation and configuration
- Configuring VoIP functionality for Local Area Network call (PC to PC Call)

#### *Version two*

- Customizing voicemail feature to allow its user to retrieve the voicemail from the server traditional way by dialing to voicemail main menu extension.
- Sending E-mail notification with voicemail as attachment to its user once there is new voicemail has been dropped in the box.
- Sending SMS notification if there is any missed call.

#### *Version three*

- Providing Voice-prompt notification features
- Providing phone number filtering feature
- VMGUI implementation which provides user with a graphical user interface to access voicemail and features setting.

### **3.4 SYSTEM PLANNING AND ANALYSIS**

#### **3.4.1 DATA GATHERING AND ANALYSIS**

The interview session were scheduled on 7<sup>th</sup> September 2010 with Mr. Ruslan, Telecommunication Lab Technician. This session basically focus not only on collecting information regarding the components used for implementing a VoIP system but also on the demonstration of the VoIP system implemented in VoIP lab.

The VoIP system implemented in VoIP lab provided a simple point-to-point call service between two PC and four analog phones with the help of CISCO 3700 series multiservice access router and SIP protocol to bridge those two PC and those fours analog phones.

Figure 4.1 illustrates the architecture of VoIP system implemented in VoIP lab which provided a point to point communication in local area network environment. CISCO 3700 series multiservice access routers were configured to receive authentication request from phone extension connected to router's through FXS port as well as other router. Even though this system only provided basic functionality which allowed workstation to call in a Local Area Network environment, it helps the author to determine that the proposed system can be implemented in UTP environment as well.



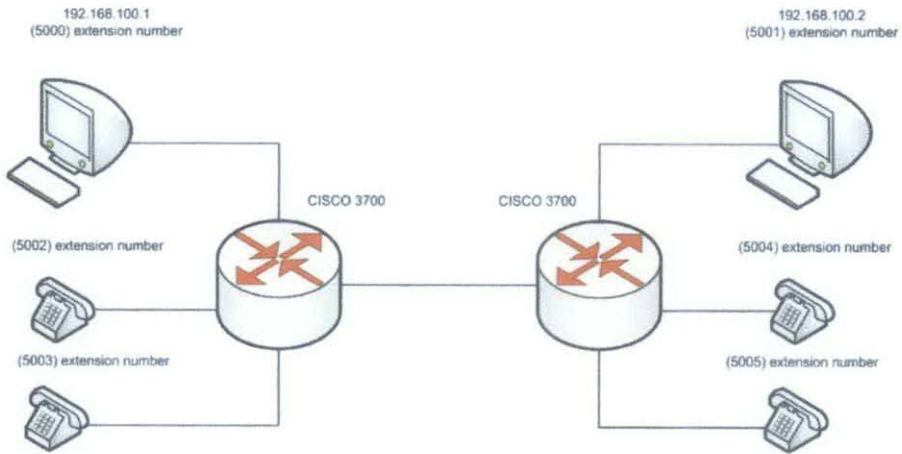


Figure 3.2: VoIP system architecture in multimedia lab

A survey is conducted with the objective to study on students' behavior towards existing UTP telephone system. This study helps to determine some of current system characteristic and some requirements to enhance the proposed system's features. Out of fifty questionnaires, there were thirty seven respondents taking part in the conducted survey. The analysis of data from the survey will be shown in graphical chart in the next section.

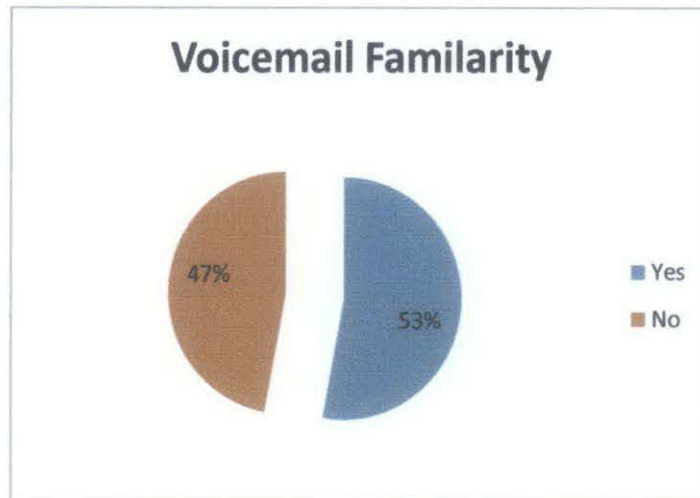


Figure 3.3: The familiarity with Voicemail Service provided for lecturer's extension

Pie chart above depicts students' familiarity to voicemail service. Out of 34 respondents for the questionnaires, there were 46% did not familiar with voicemail service embedded in current lecturer's extension. This means the service that was provided by current voicemail system was not really effective and students tended to change their direction to use other service instead.

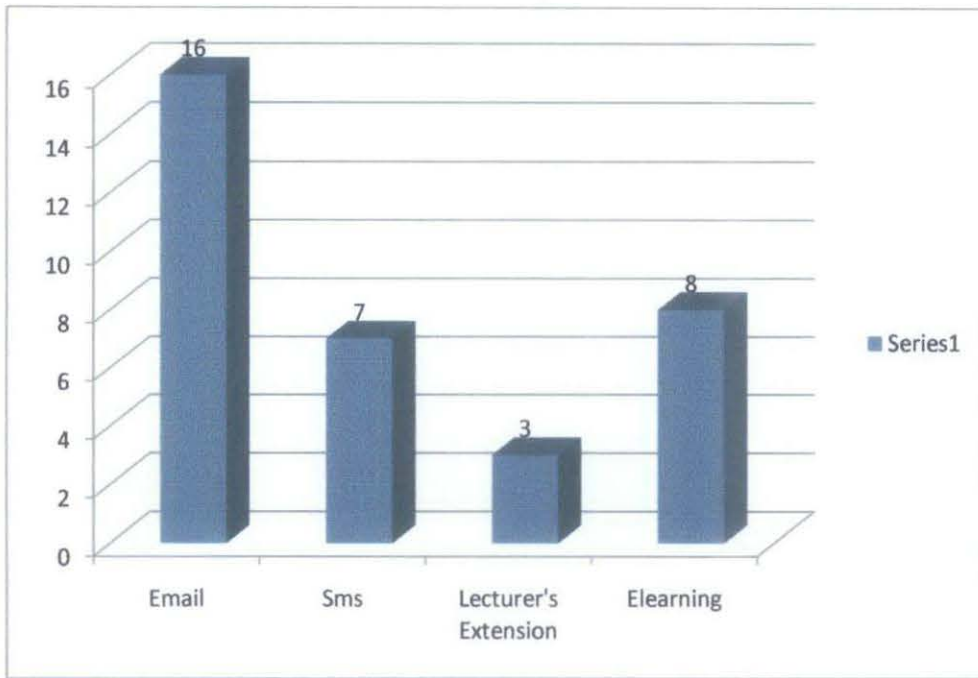


Figure 3.4: Communication means between students and lecturers

Bar chart above depicts that Lecturers' extension were less popular compare to other communication service. It is because of the service that was provided by lecturers' extension were limited and not effective. For instance, it does not notify students when lecturers can entertain their call.

A few comments were made by the respondents which related to lecturer's response to their voicemail. Some of them received their response a few days and some never received any response back. However, by using other communication service, for instance, email which had been using the most, they received the response within a day or a few hours. Thus, notification through email will be implemented in the proposed system so that missing communication between lecturers and students will be cut down.

All the responses from students show that the voicemail systems they used were still using traditional way of accessing voicemail. User needed to dial to voicemail server extension and follow instruction in order to listen to their voicemail message. For instance, a user made a call to extension 9000 to listen to his voicemail. Voice instruction was play to input mailbox and password following by please press '1' for listening to new message; press '2' to go to main menu.

### 3.5 SYSTEM DESIGN

#### 3.5.1 ACTIVITY DIAGRAM

The activity diagram for prototype UTP voicemail system contains two main activities which are call flow of the system, and accessing VMGUI and features setting flow of the system and it is shown in figure 3.5 and figure 3.6.

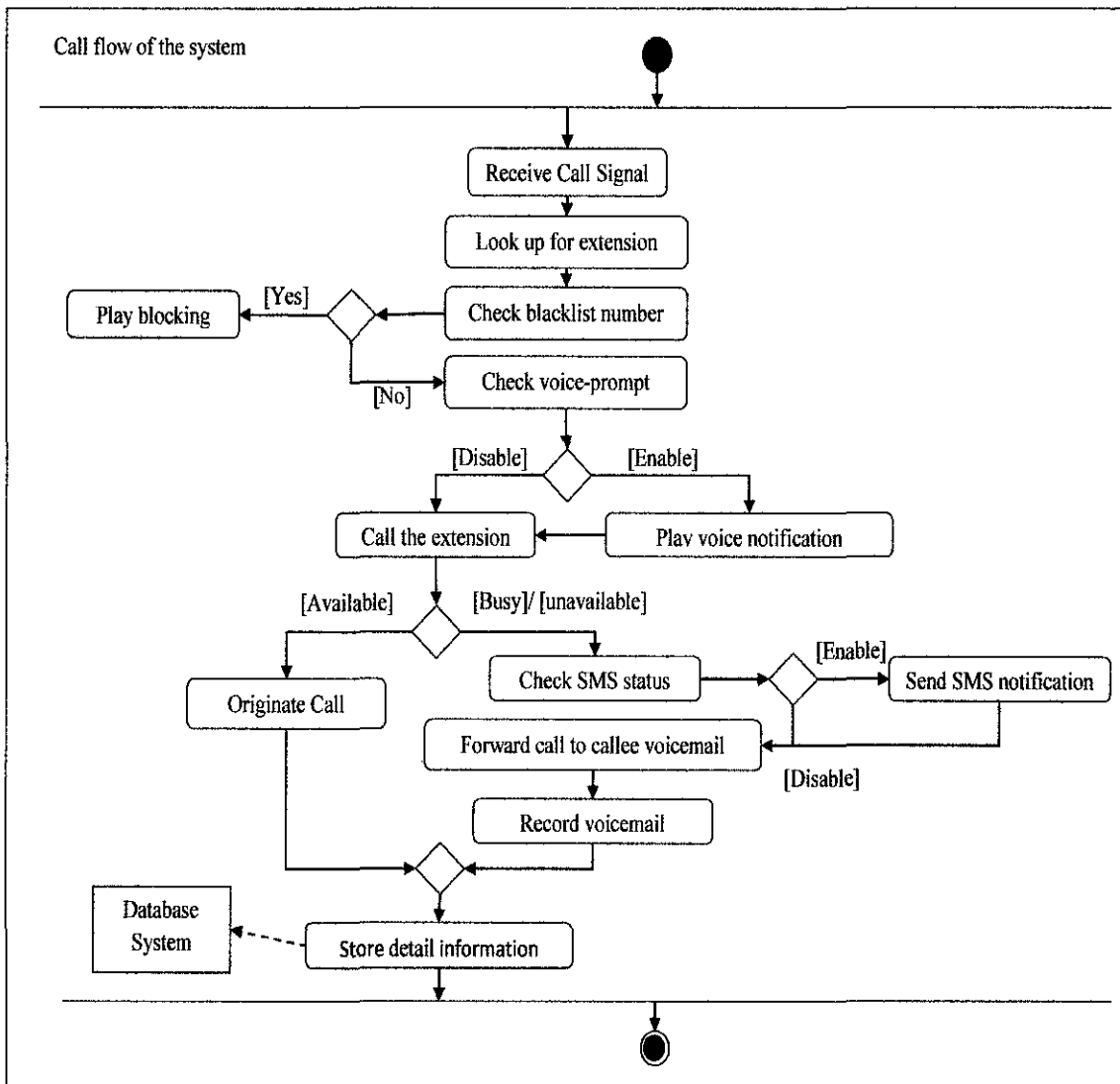


Figure 3.5: Activities diagram for UTP voicemail system (Part I)

For call flow, the process starts when server received call signal then it looks for extension which maps to the request from the signal. After mapping to the correct extension, it will check the caller number if it is available in the blacklisted list. If so, the call will be routed to

a blocking audio. On the other hand, it will continue checking if the voice-prompt feature enables. If it enables, the caller will be played a recorded voice notification by its user. Then the call will be routed to the correct extension. If the extension is unavailable or busy, it will check if the SMS notification is enable. If so, SMS will be sent to its user to notify about the missed call and then forward the call to user voicemail box to leave voice. And if the call is accepted, the call will be originated and the channel will setup. After the call is terminated, all the information will be stored in database.

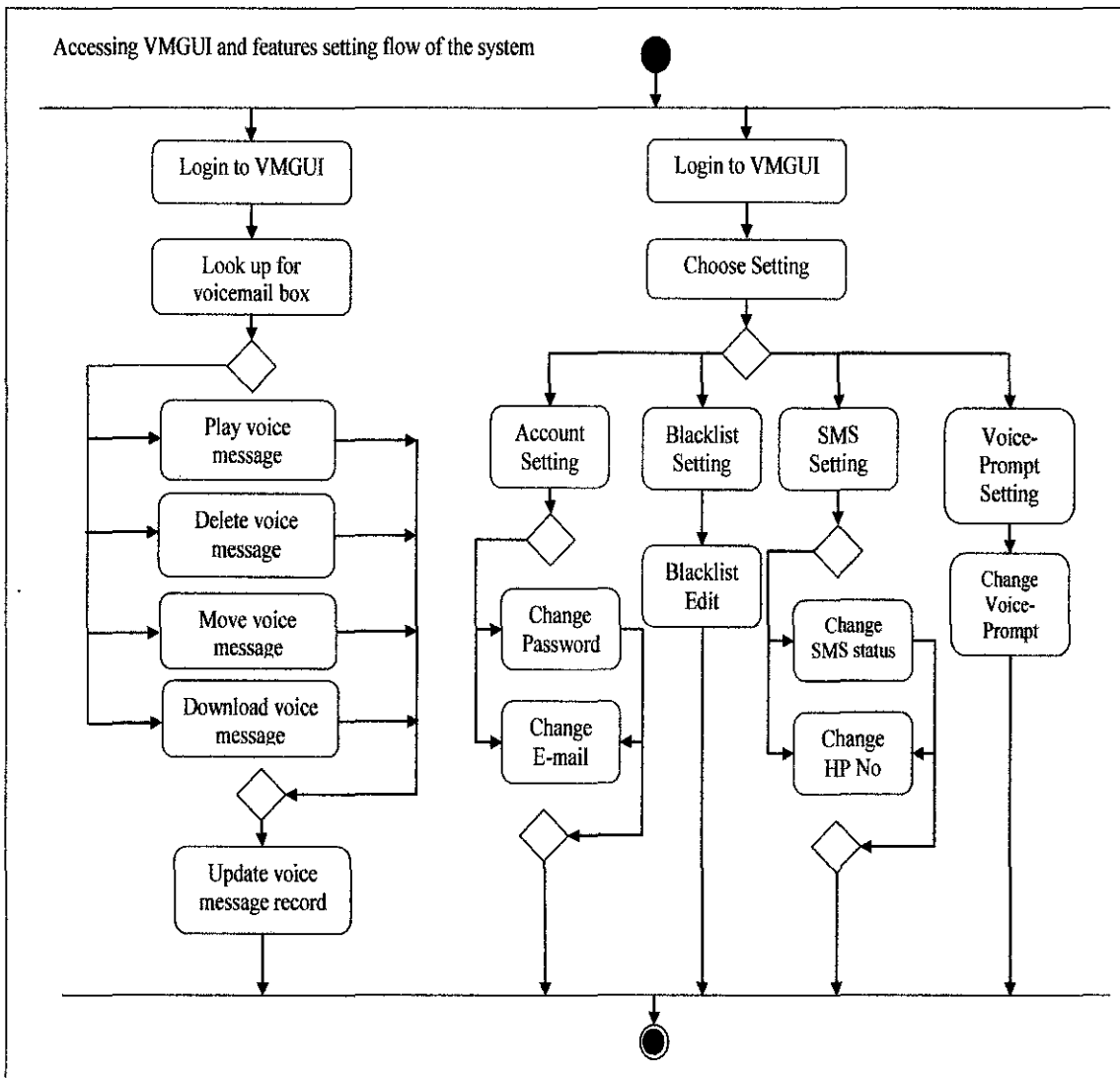


Figure 3.6: Activities diagram for UTP voicemail system (Part II)

For accessing VMGUI flow, it starts by authenticating its user and mapping them to the right voicemail box. Each user has a choice whether they want to play, delete, move, or download their voice message.

For features setting flow, after authenticating, user is able to choose the setting which they prefer to make a change. For account setting, user is able to change their current password, and email which is using for email notification. For blacklist setting, user is able to add or edit the filtering number. For SMS setting, user is able to enable or disable SMS notification feature, and to change the phone number for SMS notification to be sent to. Lastly, for voice-prompt setting, user is able to change the recorded voice notification to be played to the caller.

### **3.5.2 USE-CASE DIAGRAM**

A use-case diagram is created to describe prototype UTP voicemail system's system behavior as shown in Figure 3.7. This system has three main actors, student, lecturer, and system admin. For student and lecturer, they are able to make request for call. If the channel busy or unavailable, they will be asked to place voicemail in the voicemail box. And if voicemail server main menu extension is called, they will be routed to their voicemail box to retrieve the message traditional way. Moreover, both actors are able to retrieve their voicemail through VMGUI. For system admin, they will be able to create new users to access VMGUI, edit client's features setting, backup system configuration, and monitor voice channel.

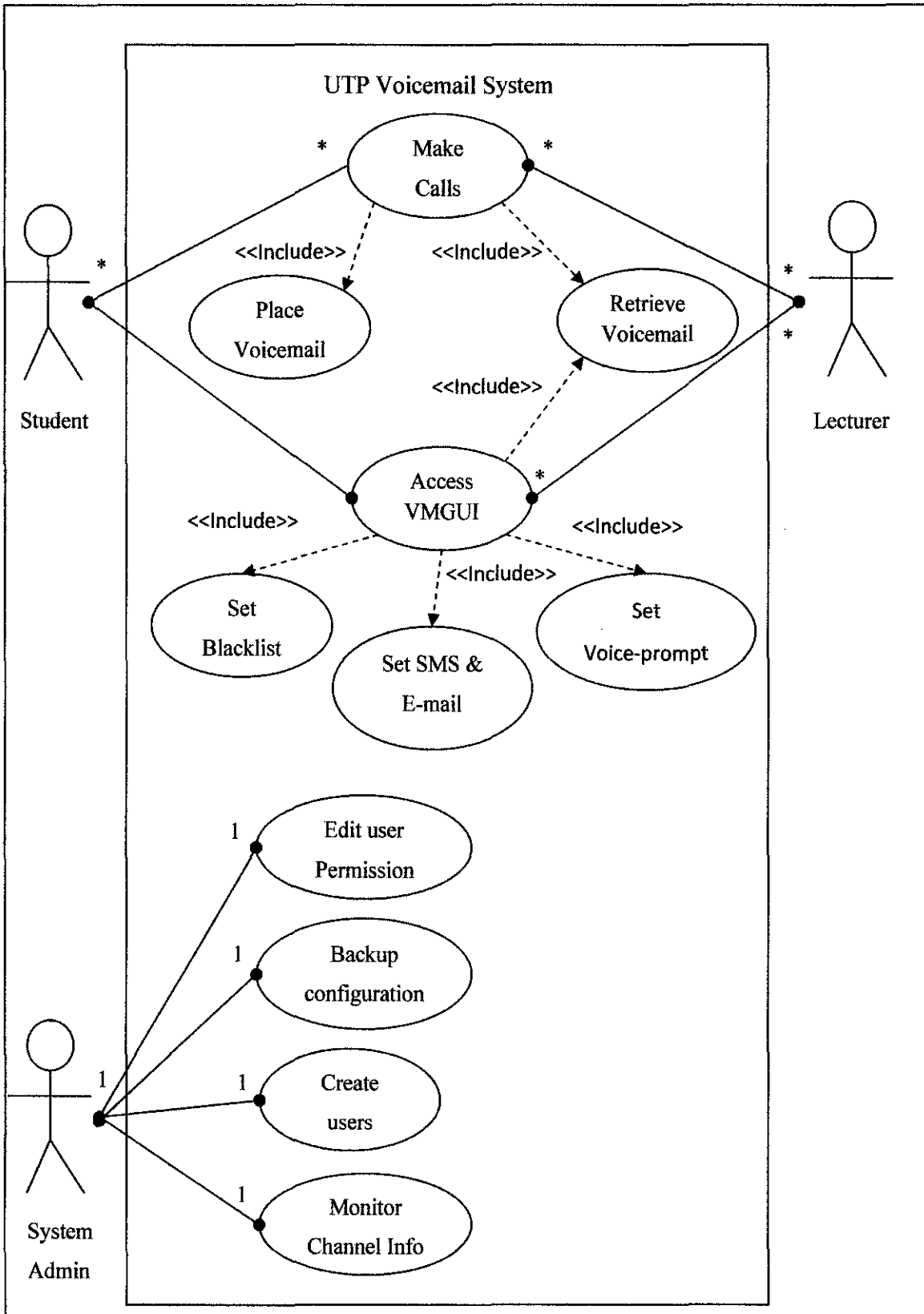


Figure 3.7: Use-case diagram for UTP voicemail system

### 3.5.3 SYSTEM ARCHITECTURE

VMSA is designed to work in four different environments which are LAN, Internet, PSTN, and VoIP gateway environment as shown in figure 3.8. However, the prototype UTP voicemail system will only support one of the four environments which are LAN because of lacking of hardware and equipment to connect to PSTN and static IP to enable Internet and VoIP gateway environment.

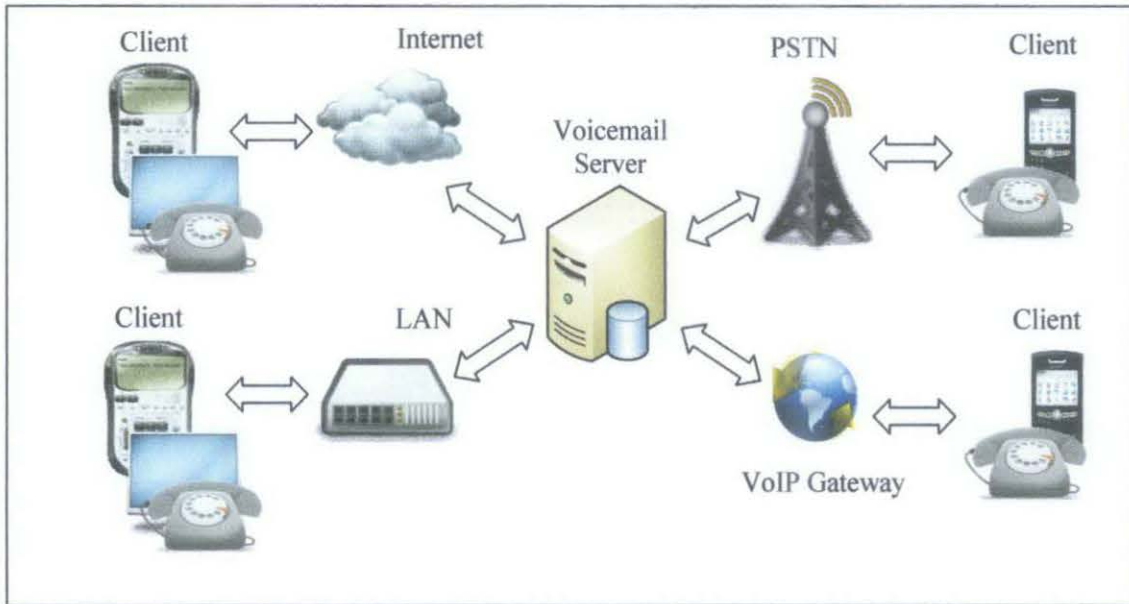


Figure 3.8: Full voicemail system architecture

This system is a client-server system whereby server's resources are handled by system admin and client's resources are requested by clients who are lecturer or student. Voicemail server in this system is basically the combination of three main components which are database, web server, and Asterisk server. Database is being used to store and retrieve information regarding all the calls. It is also used for creating new user and new mailbox for the system. Web server is used to host VMGUI which will handle a direct request from the client. And Asterisk server is being used to handle the call and the storage of voicemail for users. For clients, they are divided into three types which are IP Phone, Softphone, and VMGUI.

### **3.5.4 SYSTEM DATABASE**

In terms of database, there are a lot of databases available: Oracle, MySQL, Microsoft SQL Server, SQLite, PostgreSQL, etc. However, VMSA is only support MySQL for the time being. VMSA requires six tables which are “blacklist”, “cdr”, “extensions”, “sipusers”, “tbl\_admin”, “voicemail\_users”.

“*blacklist*” table is mainly used by phone number filtering feature which contains list of phone numbers added by user. For “cdr” table, it contains all call detail records for the system and it can be used for displaying the call history. For “extensions” table, it is used for creating the dial plan that determines the call routing operation of the system. “sipusers” table is used for creating user for LAN call. For “tbl\_admin”, it contains administrator authentication information. And for “voicemail\_users” table, it contains mailbox authentication information, SMS status, SMS phone number, voice-prompt status.

### **3.6 SYSTEM IMPLEMENTATIONS**

This is the phase where the system is built and implemented. Since the system is developed under phase-development methodology, the implementation is divided into three versions.

#### **3.6.1 VERSION ONE**

In this version, all registered users are able to call each other in a LAN environment using softphone or IP phone. This version is where operating system and all necessary software are installed such as Ubuntu 10.10, Asterisk 1.6, Apache server, gammu, festival, sox, and Mysql. Moreover, some basic configurations are made to create some users and some testing phone extension numbers. For instance, three users is created namely 2001, 2002, and 2003 which map with extensions and mailboxes 2001, 2002, and 2003 respectively.

Zoiper and X-lite softphones are program software which is used as a medium to provide voice communication over Internet. They are configured to register users that were created previously in the installation process. The free version of the both softphones has enough function for simple users to make a call to each other.



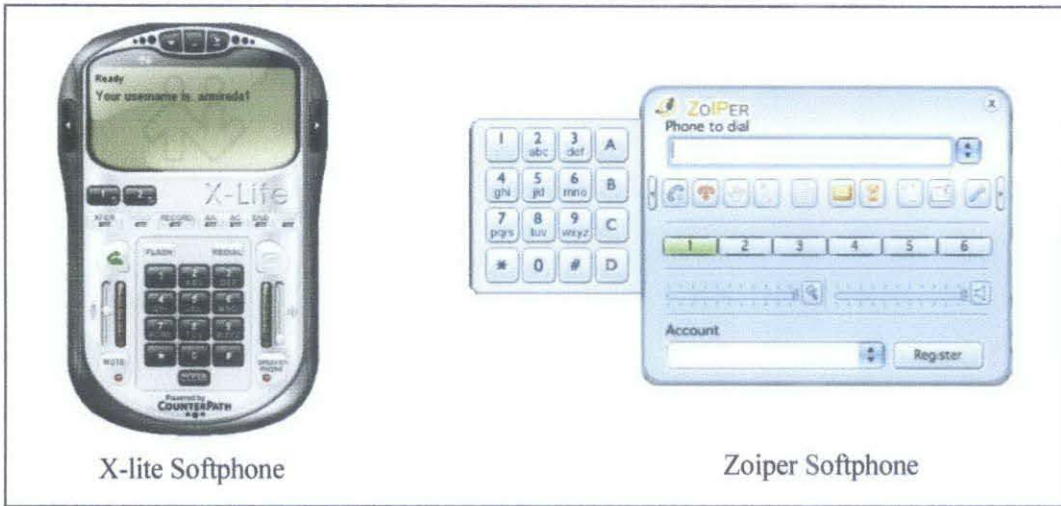


Figure 3.9 Free X-lite and Zoiper softphone for version one

### 3.6.2 VERSION TWO

In this version, voicemail capability with E-mail and SMS notification have been configured and customized. Thus, each user has its own mailbox, E-mail, and SMS associated with their phone extension. Each user is able to retrieve their voicemail with traditional way by dialing to voicemail main menu extension which in this case is 9000. Each user will be prompted to input their mailbox and password to login to the mailbox.

For call handling, if the extension is busy or unavailable, caller will be routed to voicemail server to leave voicemail for its user. Moreover, SMS notification will be sent to its user to notify if there is a missed call. Moreover, it will send notification to its users' E-mail along with voicemail as attachment if there are any new voicemail.

### 3.6.3 VERSION THREE

In this version, each user is given their own VMGUI to access their mailbox listening, downloading, archiving, and deleting the voicemail. Furthermore, VMGUI provides features setting for previous version features as well as new features including phone number filtering and voice-prompt notification feature with text to speech.

For phone filtering feature, it allows user to block the number which is suspicious to be a marketing number or number that its users do not want to entertain. For voice-prompt feature, it allows user to enable voice notification to the caller. This feature has two built-in tools which are text to speech tool and audio upload tool. Text to speech tool is tool which converts the text from the user to audio. Besides text to speech tool, users can use audio upload tool if they have their recorded audio.

For administrator, VMSA gives some control over VMGUI features such as creating new user, deleting user, and enabling or disabling user's features.

### **3.7 PROJECT TIME LINE AND MILESTONE**

This project will be broken into two main schedules which depict in two separate Gantt Charts. Each chart lists down all the activities to be completed for each semester both for FYPI and FYPII. Please refer to Appendix A.

### **3.8 DEVELOPMENT TOOL**

Below are all hardware and software to be used for the development and implementation of this project.

#### *HARDWARE*

- Two laptops (one for server and one for client)

#### ASUS F5

- Intel Pentium Dual Core 1.67Ghz
- 2GB DD2 RAM
- 80GB Hard Disk
- 128MB ATI Radeon Xpress 1100

#### Sony Vaio

- Intel Atom 1.67Ghz
- 1GB DD2 RAM
- 250GB Hard Disk
- 128MB Graphic Card

- IP phone (optional)

- Wireless Access Point
- LAN switch and cables

### *SOFTWARE*

- Ubuntu 10.10 (for Operating System)
- Asterisk 1.6 (for Voicemail Server), Asterisk-gui 2.0 (for managing Asterisk server)
- Softphone: Zoiper and X-lite (for PC to PC call)
- Apache HTTP server (for hosting web)
- Gammu (for communication with GSM modem)
- Festival (for converting from text to speech)
- Sox (for converting from .wav file to .mp3 file)
- PHP5 (for PHP script)
- MySQL (for database)
- Phpmyadmin (for managing mysql database)
- Openssh (for remote control)
- Adobe Photoshop (for graphical design) and Adobe Dreamweaver

## **CHAPTER 4**

### **RESULTS AND DISCUSSION**

In this chapter, all the results and discussions will be illustrated and discussed. The results to be included in this report are extracted from system testing throughout the whole development of the project to determine the limitation and the efficiency of VMSA. On top of that, there is some description of all system's functionality and it also shows how the system works and how to achieve in implementing each or phase of the system.

#### **4.1 LOCAL AREA NETWORK CALL**

With three username created in version one, the author try to do a testing in the hostel environment to provide a PC-to-PC call for student. After a few attempts, the author find out that softphone cannot register to the server because of network port required by softphone to send authentication request has been blocked by network administrator. However, to do successful LAN call testing, the author come out with a solution by using wireless access point to provide a sharing network connection so that users who are in the range are able to register and make a test call. This solution works fine which allows all three users registering their softphone and they are able to make a call to each other.

#### **4.2 VOICEMAIL NOTIFICATION**

For voicemail notification which has been implemented in version two, only SMS notification feature is successfully tested with Sony Ericsson W800i phone using as GSM modem. However, for E-mail notification feature, it only can send E-mail notification to local mail server because the port 25 that is used for sending E-mail has been blocked.

### **4.3 PHONE NUMBER FILTERING**

This feature has been successfully tested. Whenever receiving a call from number in blacklisted list, the call will be blocked then routed and play blocking audio to the caller.

### **4.4 VOICE-PROMPT NOTIFICATION**

This feature has been successfully tested with both text to speech tool and audio upload tool. Moreover, it is controlled by the user through VMGUI in voice-prompt setting panel.

### **4.5 VOICEMAIL GRAPHICAL USER INTERFACE**

VMGUI is a product which comes from the combination of all VMSA features. For this version, it is used as an interface for prototype UTP voicemail system which will be used by two major users: lecturers as client, and administrator. They have different permission to access VMGUI.

In the following section, the author will describe step by step all the functionality provided by VMGUI.

#### **4.5.1 LOGIN PAGE**

Both client and administrator share the same login page with different login panel to login to VMGUI as shown in Figure 4.1 and Figure 4.2. However, there is an icon on the top right corner of each login panel to identify which login mode the user is in. It allows each user to change login mode from client to administrator and vice versa by clicking on it.

With too many things to do, users usually forget their password. Thus, VMGUI provides password recovery panel as shown in Figure 4.3 to request for old password or new password. Each user can choose sending type for password recovery whether they want to request for old password for the mailbox or system reset password. After submitting the request, E-mail that contain old password or a new reset password will be sent to users' E-mail address.



Client mode ← 

## VMGUI LOGIN



Mailbox: \_\_\_\_\_  
 Context: \_\_\_\_\_  
 Password: \_\_\_\_\_

[forgotten password?](#)

Figure 4.1: Login panel for client



Administrator mode ← 

## VMGUI LOGIN



Username: \_\_\_\_\_  
 Password: \_\_\_\_\_

[forgotten password?](#)

Figure 4.2: Login panel for administrator



**Forgotten Password** X

Mailbox : \_\_\_\_\_  
 Context : \_\_\_\_\_  
 Sending Type :

Figure 4.3: Forgotten Password panel

## 4.5.2 CLIENT MAIN PAGE

After successfully authenticating, user will be redirect to main page. This page is mainly for displaying voicemail information and for user to perform operation such as playing, downloading, archiving, and deleting voicemail. If any icon is clicked, the operation will take place as shown in Table 4.1.

This page shows an alternative way to accessing voicemail rather than using traditional way to access voicemail. Furthermore, by using VMGUI, users are able to access their voicemail not only from their phone extension, but also from LAN as well as Internet which means it has improved the range of accessibility.

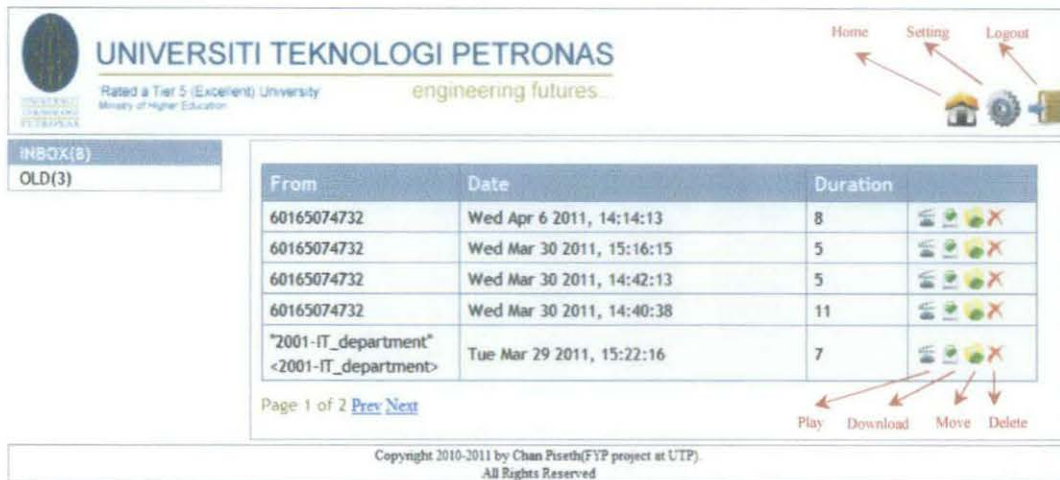


Figure 4.4: Client's main page

	There will be a pop up player page and then it autos play the voicemail.
	There will be a pop up asking whether user would like to save the voicemail.
	The voicemail will be moved to Old if it is inside Inbox folder and vice versa.
	The voicemail will be deleted from Inbox or Old archive.
	The page will be redirected to features setting page
	The session will be destroyed and the page will be redirect to login page

Table 4.1 Client main page icon and its operation

### 4.5.3 CLIENT FEAGURES SETTING PAGE

Users will be directed to features setting page whenever setting icons at the top right corner of the page is clicked. This page contains all the setting such as Account Setting, SMS Setting, Blacklist Setting, and Voice-prompt Setting which are represented by icons located on the left panel as shown in Figure 4.5. Each setting has its sub category setting. Those sub category settings works as tool which allows users to configured VMSA features such as E-mail and SMS notification, Phone number filtering, and Voice-prompt notification.

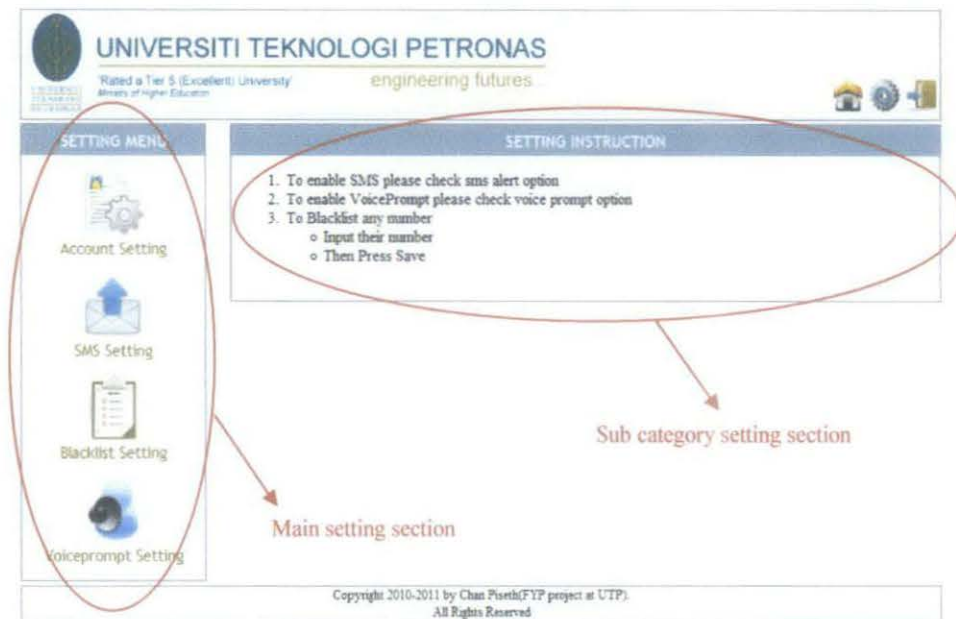


Figure 4.5: Client's features setting page

If any icons in main setting section were clicked, sub category setting section will change according to the category of the main setting icons as shown in Figure 4.6. Moreover, in Figure 4.7, it shows setting panel that associated with each sub category setting. In each setting panel, information that is submitted by user will be validated before updated in the server. It will do checking on E-mail pattern, existing user, blank field, number field, selection requires, and etc.



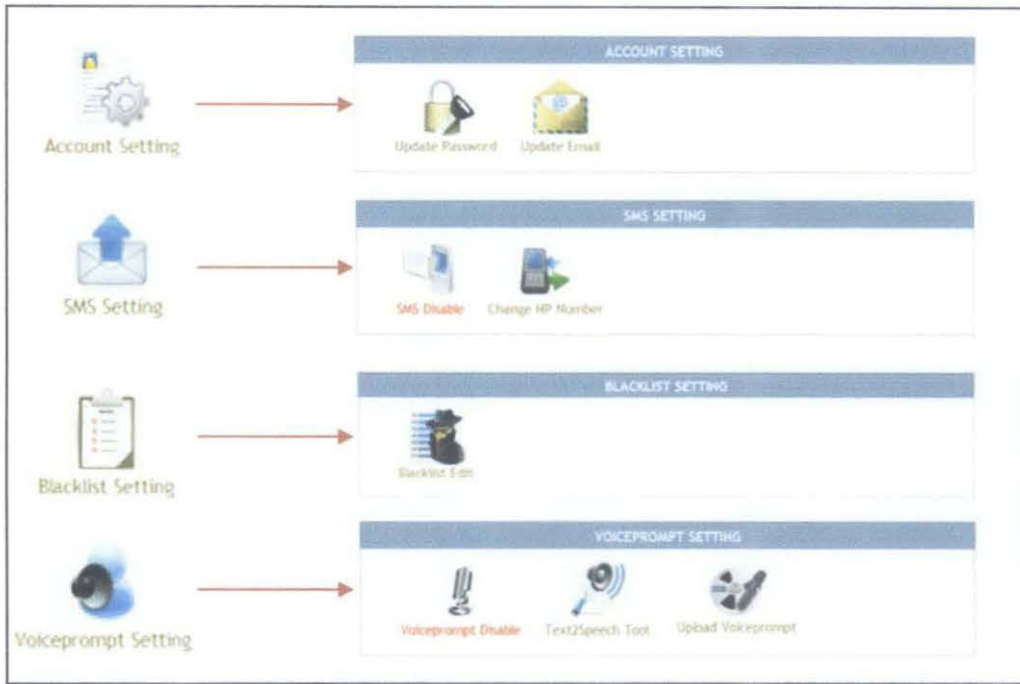


Figure 4.6: Client's features setting with each sub category setting

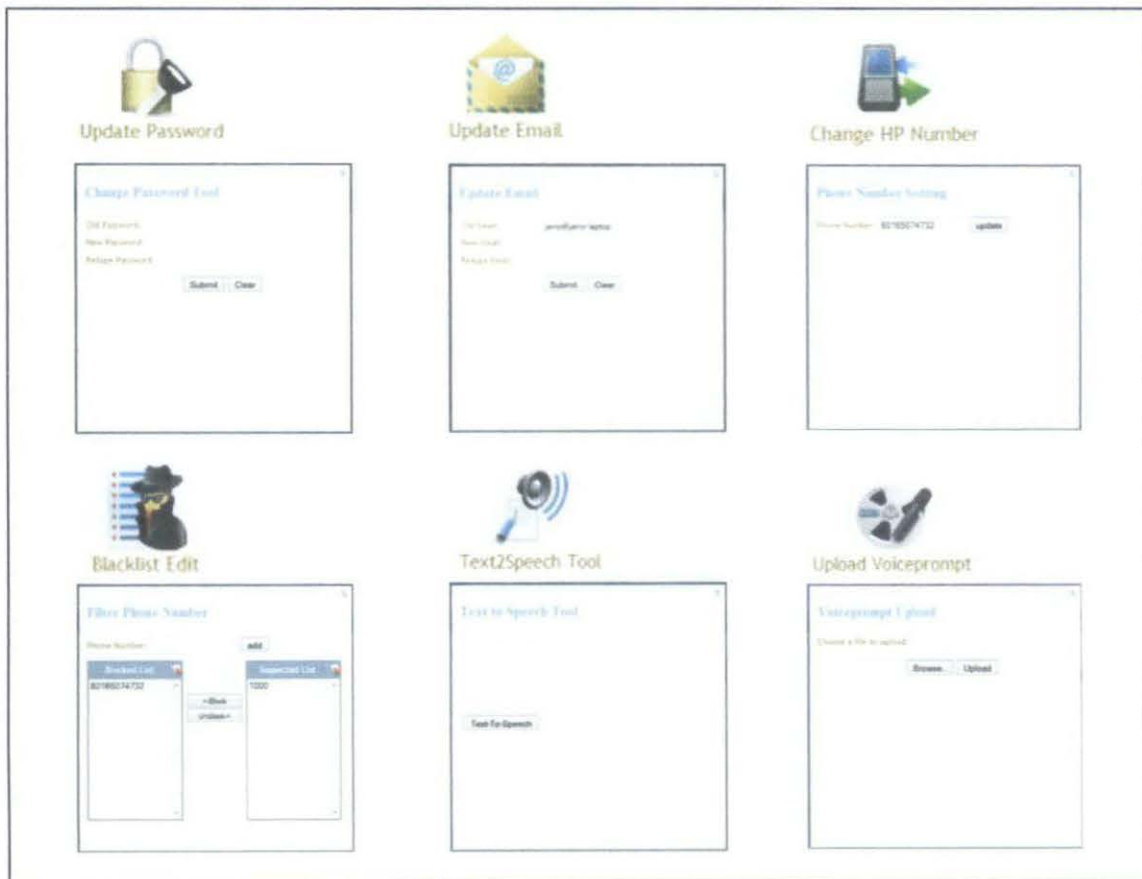


Figure 4.7: Sub category setting panel for client's VMGUI









 Update Password	<ul style="list-style-type: none"> <li>- This setting is used for changing current user's mailbox password for VMGUI.</li> <li>- If the icon is clicked, it will popup Update Password panel.</li> </ul>
 Update Email	<ul style="list-style-type: none"> <li>- This setting is used for changing current E-mail address for E-mail notification feature.</li> <li>- If this icon is clicked, it will popup Update E-mail panel.</li> </ul>
 SMS Disable	<ul style="list-style-type: none"> <li>- This setting is used for changing SMS notification feature status.</li> <li>- If this icon is clicked, it will toggle the status of SMS notification feature from Disable to Enable and vice versa.</li> </ul>
 Change HP Number	<ul style="list-style-type: none"> <li>- This setting is used for changing current user's mobile number for SMS notification feature.</li> <li>- If the icon is clicked, it will popup Change HP Number panel.</li> </ul>
 Blacklist Edit	<ul style="list-style-type: none"> <li>- This setting is used for adding or editing blacklisted number.</li> <li>- If the icon is clicked, it will popup Blacklist Edit panel.</li> </ul>
 Voiceprompt Disable	<ul style="list-style-type: none"> <li>- This setting is used for changing Voice-prompt feature status</li> <li>- If this icon is clicked, it will toggle the status of Voice-prompt notification feature from Disable to Enable and vice versa.</li> </ul>
 Text2Speech Tool	<ul style="list-style-type: none"> <li>- This tool setting is used for generating audio file from text for voice-prompt feature.</li> <li>- If the icon is clicked, it will popup Text to speech tool panel.</li> </ul>
 Upload Voiceprompt	<ul style="list-style-type: none"> <li>- This setting is used for upload user recording audio file for voice-prompt feature.</li> <li>- If the icon is clicked, it will popup Upload Voice-prompt panel.</li> </ul>

Table 4.2: The functionality for each sub category setting and its operation for client

#### 4.5.4 ADMINISTRATOR MAIN PAGE

After successfully authenticating, administrator will be redirect to administrator main page. For this prototype, this page is only design to control over user registration, user deletion, and VMGUI features for each user. This page gives some usages instruction for the setting it provided for administrator as shown in Figure 4.8.

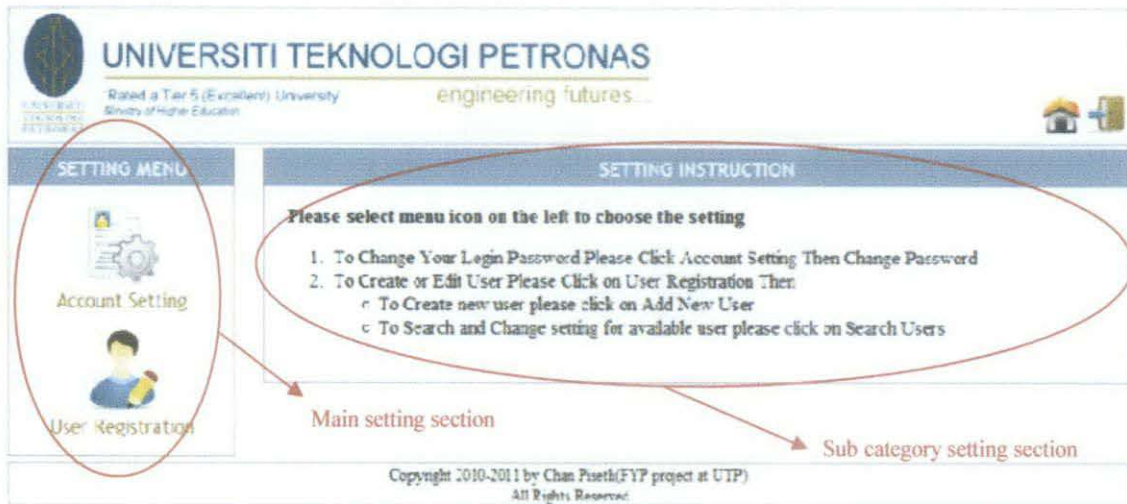


Figure 4.8: Administrator main page

If any icons in main setting section were clicked, sub category setting section will change according to the category of the main setting icons as shown in Figure 4.9. Furthermore, in Figure 4.10, it shows setting panel that associated with each sub category setting.

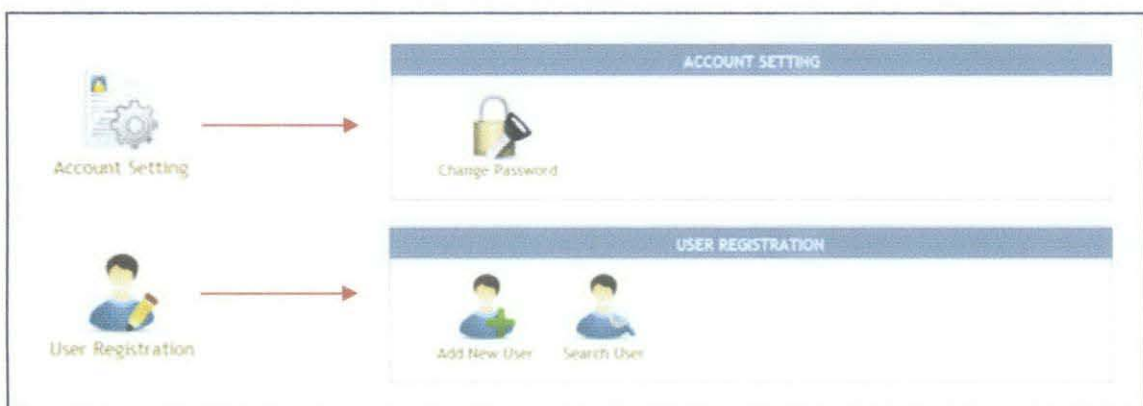


Figure 4.9: Administrator setting with each sub category setting



Figure 4.10: Sub category setting panel for administrator's VMGUI




 Update Password	<ul style="list-style-type: none"> <li>- This setting is used for changing current administrator password for VMGUI.</li> <li>- If the icon is clicked, it will popup Update Password panel.</li> </ul>
 Add New User	<ul style="list-style-type: none"> <li>- This setting is used for adding new user to access VMGUI and LAN call account.</li> <li>- If this icon is clicked, it will popup Add New User panel.</li> </ul>
 Search User	<ul style="list-style-type: none"> <li>- This setting is used for looking for existing user, deactivating account, deleting account, and changing VMSA features.</li> <li>- If this icon is clicked, it will popup Search User panel.</li> </ul>

Table 4.3: The functionality for each sub category setting and its operation for administrator

## **CHAPTER 5**

### **CONCLUSION AND RECOMMENDATION**

#### **5.1 CONCLUSION**

In conclusion, the author has introduced VMSA and a prototype UTP voicemail system which is going to be implemented to provide and to enhance communication service for current lecturer's phone extension. It started by defining the problem statement with supporting study in literature review. The objective and scope of study were well defined so that it will complete with the given time frame. At the same time, the methodology to be used for this project was presented with the support of diagrams. This was followed by some results and discussions on each features of VMSA.

With VMGUI implemented in version three, the prototype UTP voicemail system helps to improve the way its users accessing their voicemail. It provides flexible and wide range of accessing capability. By having this system implemented in UTP, the communication between lecturers and students is optimized. The number of losing communication between lecturers and students is reduced. On top of that, users have a full control over its voicemail box storing or retrieving their voicemail message through VMGUI.

#### **5.2 RECOMMENDATION**

##### **5.2.1 UNIVERSITI TEKNOLOGI PETRONAS**

The author suggests that CIS department should provide more cooperation by scheduling testing period for FYP project which involves Campus network. CIS department should provide a short briefing on how to make request to have a project testing on Campus network to ITMS. For instance, the proposed system cannot be tested in different network segment while it is firewalled by ITMS.

### **5.2.2 SYSTEM FUNCTIONALITY**

As the proposed VMSA is implemented and tested in local area network environment, it still has a lot of rooms for future enhancement. The system can be configured and customized to work with Internet environment and VoIP by assigning static IP with domain name to provide a free and cheap calling rate for an organization. Moreover, with the power of SIP and Asterisk, the system can be implemented to have an interconnection with other organization. For instance, phone extension in PETRONAS office at KL could connect to phone extension in PETRONAS office at UTP. Each user in both offices can make a call to each other easily.

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**APPENDIX A**  
**PROJECT TIME LINE AND MILESTONE**  
**(GANTT CHART)**



### PROJECT TIMELINE AND MILESTONE FOR FYPI

#	Detail/Week	1	2	3	4	5	6	7	8	9	10	11	12	13	14
1	Selection of Project Title														
	i. Research on Topic														
	ii. Submission of Project Proposal														
	iii. Topic and supervisor approval														
2	Requirement and Definition analysis														
	i. Project background study														
3	Submission of Preliminary Report														
4	Requirement and Definition analysis (Cont)														
	i. Literature Review														
	ii. Information gathering (Interview and survey)														
5	Submission of Progress Report														
6	System version 1 implementation														
7	Submission of Interim Report														
8	Oral Presentation														

**Note: Pink is Main Process and Blue is Sub-Process**

**PROJECT TIMELINE AND MILESTONE FOR FYPII**

#	Detail/Week	1	2	3	4	5	6	7	8	9	10	11	12	13	14
1	Submission of progress report														
2	Poster Presentation and Pre-EDX														
3	Submission of final draft dissertation														
4	Viva Presentation														
5	Submission of technical report and final dissertation														

**APPENDIX B**  
**VMSA INSTALLATION**  
**AND**  
**CONFIGURATION**

## **VMSA requirement**

- OS installation : Ubuntu 10.10 (a simple OS installation)
- Asterisk server installation (Asterisk 1.6)
- Other software Installation such as gammu, festival, sox
- Download some necessary packages such as “libss7-1.0.2”, “libpri-1.4.11.5”, ”dahdi-linux-complete-2.4”, ”asterisk-1.6”, ”asterisk-addons-1.6” from “<http://www.asterisk.org>”

### **1. Asterisk server installation**

Before installing Asterisk package there are a few packages to be installed such as

- libss7-1.0.2
- libpri-1.4.11.5
- dahdi-linux-complete-2.4

There are a few steps to install those packages:

- a. Extract those packages then open the terminal and go to the packages folder
- b. Compile the package using “./configure” following by “make” command and after successful compile, using “sudo make install” to install the package. (do same thing for other packages)

Asterisk-1.6 installation steps:

- a. Extract the packages then open the terminal and go to “asterisk-1.6/contrib./scripts”
- b. Execute command “./install\_prereq install” to install some dependent packages
- c. Go back to” asterisk-1.6” folder then execute “./configure” following by “make menuselect”. After select some necessary options, execute “make” following by “sudo make install” command.

Asterisk-addons-1.6 (for app\_mysql) installation steps:

- a. Extract the packages then open the terminal and go to the package folder
- b. Execute “./configure” following by “make menuselect” command. After select some necessary options, execute “make” following by “sudo make install” command.

## 2. Setting up VMGUI

There are four main things to be considered for setting up VMGUI

- For VMGUI web server, copy “FYP” folder in the VMSA folder to “/var/www”.
- For AGI script, copy all the script in “agi-bin” folder in the VMSA folder to “/var/lib/asterisk/agi-bin”
- For Asterisk configuration, replace “/etc/asterisk” folder with “asterisk” folder in VMSA folder.
- Import “asterisk.sql” from VMSA folder through phpmyadmin