Integration of PSTN and IP Based Telephony for Voice Messaging System

by

Md. Soheb Ahmed

MASTER OF SCIENCE IN INFORMATION AND COMMUNICATION TECHNOLOGY

Institute of Information and Communication Technology

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY

2008

The thesis titled "Integration of PSTN and IP Based Telephony for Voice Messaging System" submitted by Md. Soheb Ahmed, Roll No: MP0231016, Session 2002-2003 has been accepted as satisfactory in partial fulfillment of the requirement for the degree of Master of Science in Engineering (ICT) held on May 28, 2008.

BOARD OF EXAMINERS

1. ______________________________

Prof. Dr. S. M. Lutful Kabir — Chairman
Professor and Director
Institute of Information and Communication Technology
BUET, Dhaka 1000

2. ______________________________

Mr. Md. Saiful Islam — Member
Assistant Professor
Institute of Information and Communication Technology
BUET, Dhaka 1000

3. ______________________________

Dr. Md. Liakot Ali — Member
Assistant Professor
Institute of Information and Communication Technology
BUET, Dhaka 1000

4. AzHoque

Mr. Azimul Huq — Member
CEO, New Edge Infotech Services Limited — (External)
42/1 Indira Road, Dhaka 1215

CANDIDATES DECLARATION

It is hereby declared that this thesis or any part of it has not been submitted elsewhere for the award of any degree or diploma.

Md. Soheb Ahmed

Dedicated

To

My Parents

Page No.

CID	Caller ID
CLI	Command Line Interface
DDN	Digital Data Network
DTMF	Dual Tone Multi Frequency
EDGE	Enhanced Data rates for GSM Evolution
GNU	GNU General Public License
GPRS	General Packet Radio Service
HVMS	Home Voice Messaging System
ID	Identification
IPT	IP Telephone
ISP	Internet Service Provider
IVRS	Interactive Voice Response System
LAN	Local Area Network
MAN	Metropolitan Area Network
NSP	Nation Wide Service Provider
OS	Operating System
PSTN	Public Switch Telephone Network
SVMS	Single Voice Messaging System

TCP/IP	Transmission Control Protocol/Internet Protocol
VMS	Voice Messaging System
VOIP	Voice Over IP
VSAT	Very Small Aperture Terminal
WAN	Wide Area Network

Acknowledgements

First and foremost I would like to thank Prof. Dr. S. M. Lutful Kabir, Director, Institute of Information and Communication Technology, Bangladesh University of Engineering and Technology, Dhaka, Bangladesh who gave intellectual latitude to perform this task and provided stimulating encouragement for my research. His guidance, cooperation and suggestion were a continual source of inspiration. His dedication to his chosen field, genuine love of science and incisive advice was a welcome anchor throughout my post-graduate career.

I thankfully acknowledge advice and encouragement of Mr. Md. Saiful Islam, Assistant Professor, IICT, BUET. I would like to give thanks to Dr. Md. Liakat Ali, Assistant Professor, IICT, BUET for his suggestion and help.

Mr. Md Khairul Alam, Senior Program Coordination Officer, IICT, BUET is remembered for his help and support.

I also would like to thank officials of IICT, my colleagues, my friends and my family members for helping me during this work.

Abstract

This thesis describes the implementation of Voice Messaging System, such system has been used by organizations as a non-real-time communication system for a long time. In Voice Messaging System when a user dials to a system using any telephone network, he has to authenticate himself and than he can hear voice messages sent by other users of the system. User can also record and send voice messages to other users of the system.

The limitation of the present Voice Messaging System has been discussed in the thesis. Their proprietary nature and capabilities of handling specified number of users prohibit their use in cost effective manner. The thesis also identifies that source codes of commercial systems are not open. Moreover, they are limited to a fixed number of nodes.

After identifying limitations of existing systems, this work developed a new concept of implementation of Voice Messaging System which is open, cost effective and capable of future expansion. In our proposed architecture Single Voice Messaging System developed by open source tools shall cover small geographical area. than these Single Voice Messaging Systems will be liked with WAN infrastructure to cover large geographical area. Prototype of our proposed system with open source tools has been developed and tested to examine the possibility of implementing large scale Integrated Voice Messaging System.

CHAPTER 1

INTRODUCTION

1.1 INTRODUCTION

Voice Messaging System [1,2,3,4,5] has been used as a non real-time communication system for a long time. Often we want to send message but the intended recipient of the message cannot be reached. Using Voice Messaging System we can send the message to the recipient's voice mail address as like as usual postal services. In a very simple Voice Messaging System an ID and password are assigned to every user. User logs in to the system using PSTN/IP/Mobile telephone set, uses handset buttons for authentication, for receiving and sending voice messages. These types of services have been used by Community Service Providers, Educational Institutions and Commercial Organizations for many years

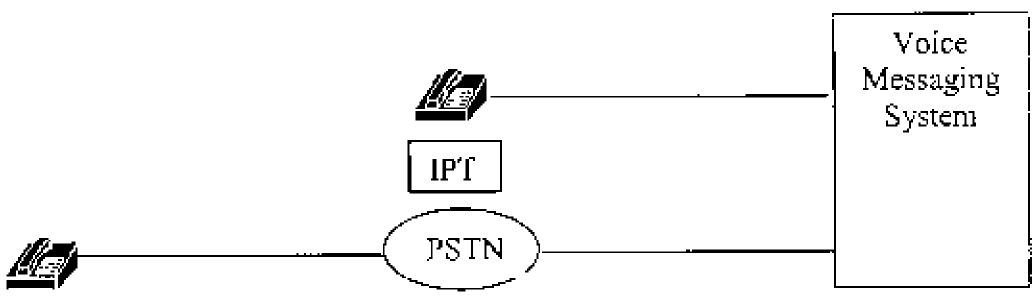

Fig 1.1: Simple Voice Messaging System

Voice Messaging System has similarity with Answering Machine but has many advanced features. Voice Messaging System record voice messages from anyone but delivers only to the registered user or users group. Each registered user is associated with a message box and message box is private for that particular registered user. Voice Messaging Systems are sized according to required numbers of ports/connections, these ports/connections establish connection between user phone system and the Voice Messaging System. More ports in a Voice Messaging System will allow more users to send and retrieve voice messages simultaneously.

Voice Messaging System uses pre recorded message or spoken instruction to guide user. These prerecorded messages or personalized messages are played based on time and events such as line busy, business hour closed etc. User may receive message-waiting indication by pager, stutter dial tone or indicator light on telephone.

Voice Messaging System provides subscribers to check and access messages from any phone and to perform the following tasks [3]:

- User can place multiple new messages and respond to multiple messages with a single call to the Voice Messaging System. User of a Voice Messaging System can respond to a message, place a new message or forward a message and can return to the messaging system to process other messages with a single call.
- User can place unicast, multicast or broadcast message to other user or group
- User can retrieve messages based on message priority.
- User can easily identify other user/group by user/group ID.
- User can forward voice messages as e-mail attachments to any e-mail user, enabling users of different voice-mail systems to share voice-mail messages.

1.2 BACKGROUND AND PRESENT STATE

We have analyzed features and characteristics of existing Voice Messaging Services, Commercial Voice Messaging Systems and Development Tools for voice messaging systems. We have also discussed pros and cons of such systems and possibility of using such systems to cover large geographical area.

1.2.1 EXISTING VOICE MESSAGING SERVICES

Voice Messaging System has been used for community services and social services in many countries. VMS has been used in agricultural sector, medical sector, weather forecast sector, disaster control sector etc. Examples of VMS services are Community Voice Mail (CVM) in USA and Australia, Lifelines in India, Teleminder in USA etc..

Community Voice Mail (CVM) was founded in seattle, USA in 1991 and CVM is a nationwide program for helping people in emergency [1]. CVM provides free 24-hour voice message access to people with 32 CVM sites. Each year CVM serves 40,000 people in 400 towns/cities nationwide. Common goals of CVM are employment, housing, domestic violence avoidance, accessing vital services etc. Target users of CVM are people in crisis, homeless, laid off workers, violence victims, runaway youths, foster childrens, migrant workers etc. CVM has also started working in Australia with help of six government and non government agencies.

Lifelines India is a telephone based information services that provides guidance on improved farming methods and provides advice on market access to rural farming communities [6,7]. Farmer can query on an automated voicemail system with community fixed phone or with mobile phone. Community workers search frequently asked question database in the system or communicate with experts to find an answer. The answer is attached as a voice file with the original query mail and send to the farmer to pickup within 24 hours. Lifeline India was started in 2006 and now serving 40,000 farmers in 700 villages. System receives 250 calls in average everyday and 88,000 frequently asked questions are now stored in database.

TeleMinder developed by "Community Voice Mail for Routine and Disaster Services" is a map based system which automatically calls people and give them emergency instructions [8]. Teleminder notify users in times of imminent danger, ask to evacuate older residents before disasters, helps getting emergency services after disasters, helps relief agencies to communicate during disaster recovery. In normal time system provide community wide services such as daily check-up calls by old people, telephone bulletin board on community events and services.

Medical service providers like lifeline USA, Bexhill Hospital UK are using Voice Messaging Systems to give routine medical services to aged people. Enterprises like Wall Street Journal, AGIA Alliance, One Account Solutions Group are also using Voice Messaging Services for sending inbound and outbound messages to customer. Campus Voice Messaging Systems are widely used in Universities like MIT, University of IOWA, Yale University, Concordia University, University of Victoria, University of Arizona etc [1]. These Voice Messaging Systems are single system in nature and provides telephone bulletin board services to users.

1.2.2 EXISTING COMMERCIAL VOICE MESSAGING SYSTEMS

VOCP system founded in 2000, is a full featured Call Answering and Voice Messaging System [5]. It is an open sourced system and now under Psychogenic INC's open projects department VOCP is a software based single site centralized system, VOCP requires voice modem and Linux/Unix variants operating system. VOCP supports unlimited number of voicemail, user can access and navigate with their touch tone telephone pad. User can send and receive faxes, listen to email using text-to-speech conversion, filter and redirect calls based on caller ID information etc. A single executable program and a number of Perl modules comprise the core of the VOCP system. VOCP handles incoming calls, processes user input and allows callers to navigate the system to leave and retrieve voice messages, email messages, pager numbers and faxes. The core program is free but any technical support or customization requires considerable amount of charge.

IVM is a telephone answering machine, voice massaging system, call attendant, and interactive voice response (IVR) program for Windows [9]. It connects to the phone using a standard voice modem, professional telephony board or to a VoIP service. IVM is software based single system and an effective voice message server, call attendant, info-line and autodial solution for small to medium businesses IVM can redirect in coming calls during office hours or act as a PC answer machine and take messages for a number of voice mail boxes after business hours. All calls answered or non answered are logged with date, time and caller ID. The recorded messages can be played at any time, forwarded to an email address, accessed via the internet and if necessary, saved for future reference. IVM can run on Windows 95/NT4/98/2000/Me/XP/2003/Vista or Linux variants. Licensing is free to use as a personal answering machine, but licensing price is higher for commercial use. IVM can be installed with failover option in two machines.

Panasonic has hardware based Voice Messaging system ranging from 6 ports/ 64 voice box to 64 ports/1024 voice boxes [3]. Panasonic VMS are non configurable and costly. SoundBite Communication's Voice Messaging Solutions are software based and highly efficient system. SoundBite provides customized solution based on customer requirements. AVAYA has Voice Messaging Solution but these solutions are integral part of IPT or IPCC solution. AVAYA IPT/IPCC solution can be purchased as software or can be purchased as bundled with hardware. Cisco Unity offers a reliable, secure, scalable, and full-featured voice and unified messaging platform [10]. Cisco Unity can be purchased as software or can be purchased as bundled with hardware. Cisco Unity is basically designed for LAN users and has failover option. It can be designed for clustered implementation. Both AVAYA and CISCO solution are of single centralized architecture.

1.2.3 EXISTING DEVELOPMENT TOOLS FOR VMS

Proprietary and non proprietary tools are available to develop telephony and voice applications. These Application Development Tools can be used to build full featured Voice Messaging System on windows/linux/unix operating system.

Telephony Application Programming Interface (TAPI) was developed jointly by Microsoft and Intel, is set of classes to write application for telephone communication devices [11]. TAPI is set of API and by using these set of API, application can be written on windows platform for sending and receiving data, voice and fax. TAPI provides interfaces between computer telephony applications and telephony services It's possible to develop wide range of systems like VOIP, PBX, CallControl, IVR, VMS, Video

Conferences with TAPI. TAPI interfaces can be called from standard programming language like C/C++, Java, VB and C#.

Java Telephony Application Programming Interface (JTAPI) is an object oriented Application Programming Interface for computer telephony applications based on Java language [12]. JTAPI is platform independent and serve as an interface between Java applications and telephone systems. JTAPI has sets of function packages, core function package provides basic structure for telephony processes as call placing, call answering and call dropping. Other function packages provide additional telephony application features. JTAPI is similar to Microsoft and Intel's Telephony Application Programming Interface (TAPI).

IVRS (Interactive Voice Response System) is relatively new technology for Linux OS, aimed to integrate computer and telephony applications [13]. IVRS module is developed in Perl, VMS can easily be developed by using voice modem or voice telephony cards and Perl language. It can play message (sampling rate 11025 bytes/sec), get Caller ID, send /receive fax, Record message, dial out side number, generate fast response (no need to wait for message to complete), interface with data base, in short almost every thing that a commercial IVRS can do. The only limitation is number of telephone line that can be supported on one machine.

One of the most widely used module in Linux World for Voice Application is the Modem::Vgetty module [14]. Modem::Vgetty module is a Perl interface to the vgetty program, which can be used for communication with the voice modems. Vgetty is an encapsulation object for writing applications. The answering machines and sophisticated voice applications can be written using this module. Vgetty is distributed with linux/unix mgetty peackage and allows to write voice application in Perl Language.

1.2.4 FINDINGS

Existing Voice Messaging Systems are of small scale and confined to a small geographical area, based on dedicated hardware or customized software. Access to the local system is by POTS/PSTN or IP Telephony [15,16]. These systems have the basic feature of Voice Messaging but not suitable for covering large geographical area, because

- Commercial Voice Messaging Systems [3,.5,9,10] are proprietary system, costly and require huge physical implementation which is impractical for large scale projects.
- Many Commercial Unified Messaging Systems are for IP Telephony or Call Centre Solution Voice Messaging is Optional Feature in those Systems [9].
- Developing non proprietary Voice Messaging System is possible [11,12,13,14], but such implementation usually covers a small geographical area.
- Deploying Single Voice Messaging System to cover large area is impractical due to requirement of huge ports or connection points in hardware.
- Centralized Single Voice Messaging System is not suitable for covering large geographical area. User access to such centralized system by POTS/PSTN is costly.
- Use of IP Telephony system [15,16] instead of PSTN requires IP telephone set and Network Infrastructure and it is not suitable for a large number of people and large scale implementation.

1.3 OBJECTIVES WITH SPECIFIC AIMS AND POSSIBLE OUTCOMES

Objective of this thesis is to develop a Voice Messaging System for mass people covering large geographical area. In our proposed system we want to develop an Integrated Voice Messaging System by linking multiple Single Voice Messaging Systems (SVMS) with TCP/IP network. Each SVMS will cover a small geographical area or a predefined small number of users, multiple SVMS will be linked together to cover a large geographical area or a large number of users. SVMS are building block of our proposed Integrated Voice Messaging System and each SVMS will act as a node (in upcoming sections "SVMS" and "node" will have same meaning for our Integrated Voice Messaging System) SVMS will reside in the local telephone exchange and linking between these nodes will be by TCP/IP Network.

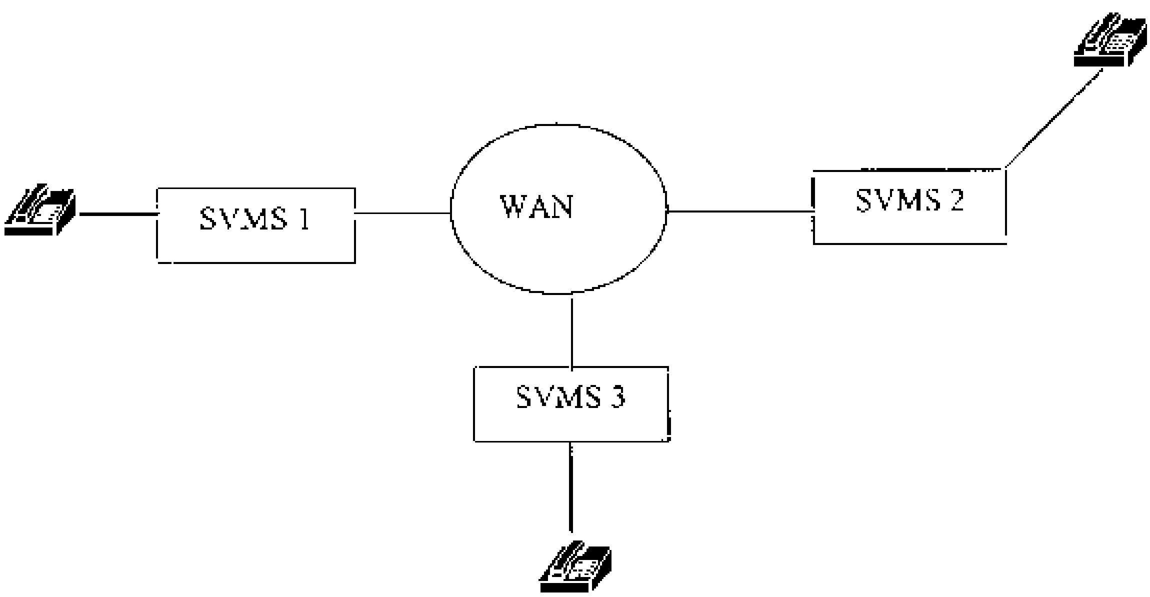

Fig 1.2: Proposed Integrated Voice Messaging System

Every user will be registered to a particular SVMS (node) and that system will be called his Home Voice Messaging System (HVMS). User group can be registered to a particular SVMS or multiple SVMS User will access his HVMS using any PSTN/Mobile/VOIP telephony system. User will be able to send voice messages to any registered user/group residing in any system but only the registered user will be able to retrieve message from his home system.

Message retrieval, storing, sending and forwarding can be done using telephone set. If a user wants to send voice message to another user/group in the same system (node), transfer is within that system and inter system communication will not occur. For sending to a user/group in another system, TCP/IP network will be used for inter system communication. ISP, NSP and Local Exchange can be used to provide such facility. For inter-system group messaging, only one message will be forwarded to other system and distribution to group member's message box within that system will be done locally.

1.4 USES OF INTEGRATED VOICE MESSAGING SYSTEM

- Integrated Voice Messaging System can be implemented as nation wide system for poor and roaming people to find jobs and other social services. An organization can send interview schedule by voice message to a roaming user or any healthcare organization can broadcast message regarding spread of diseases to all user.
- Broadcast Messages can help to inform Business Opportunities, Training Opportunities and any other Community Announcements to user and users group.
- Centralized Group Messaging Facility can be used for agricultural sector, medical sector, weather forecast, disaster control sector etc.
- Commercial Organization can use Integrated Voice Messaging System for nation wide commercial services such as advertisement, user billing information, offline support services etc.
- Voice Messaging System can be used to provide telegraph like facility in post offices by using speech to text conversion.
- Data gathered from Voice Messaging System database can be used for Socio Economics Analysis.

1.5 ORAGANIZATION OF THE THESIS

- Chapter 1 of the thesis starts with an introduction to Voice Messaging System. It is followed by study on existing services and systems, development tools, objectives and organization of the thesis.
- In chapter 2 Design of proposed system has been discussed. User/Group ID Scheme, User Authentication, Message Storing, Node IP Addressing, Inter Node and Intra Node Communication etc. were discussed in this chapter.
- In chapter 3 Components, Features and Security of the proposed system has been discussed.
- In chapter 4 development of the proposed system has been discussed. Prototype system of our proposed Integrated Voice Messaging System has been developed and this prototype system is discussed in this chapter.
- In Chapter 5 Conclusion and Recommendation for future work have been discussed. The thesis ends with citing references and appendices.

CHAPTER 2

DESIGN OF AN INTEGRATED VOICE MESSAGING SYSTEM

2.1 INTRODUCTION

Proposed Integrated Voice Messaging System is designed by linking multiple Single Voice Messaging System (SVMS) with TCP/IP network [17]. Every user or group must be associated with a home node (HVMS) and user can be local or user can be roaming user. Message Delivery within a system is straight forward, just copying the message to the recipient box. Delivery to a remote user is little complex, first the sender's system have to find the IP address of the remote system and then it will transfer the message to the remote system. Message retrieval by any user is only possible from user home node, user will not be permitted to retrieve message from any non home node. Retrieval voice messages from non home node will bring complicacy like user authentication and downloading full message box from remote node.

2.2 PROPOSED ID SCHEME

We need to assign ID to users/groups with particular internal structure for our proposed system. Every user/group must have a HVMS (node) and must register with his home node. Group will reside within a single node or will reside in multiple/every node. Same ID structure will be used to identify both normal user and group.

In our proposed scheme first 3 digits of the ID will be called System Digits and will identify any SVMS. It is possible to address nine hundred ninety nine (999) nodes with three (3) system digits. These 3 digits must be unique and every node must have an IP address associate with these 3 digits. Next digit will be called Type Digit and will identify whether the ID is for normal user or for group. Last 5 digit will be called Identification Digits and will identify particular user or group. If required number of digits for any part of the ID scheme can be increased.

xxx	–	x	–	xxxxx
VMS		User or Group		ID
(System Digit)		(Type Digit)		(Identification Digit)

ID with all "0" (000-x-xxxxx) in first three fields means that it is applicable to all system.

"1" in the fourth field (xxx-1-xxxxx) means normal user and "0" (xxx-0-xxxxx) means user group.

Voice Messaging System Location	User/Group Name	System Digits	Type Digits	Identification Digits	Proposed ID
Moghbazar, Dhaka	Mr. Amin (Normal User)	001	1	11111	001111111
Mirpur, Dhaka	Mr. Sohrab (Normal User)	002	1	11111	002111111
Phulbari, Dinajpur	Mr. Kamal (Normal User)	111	1	11112	111111112
Faridgonj, Chandpur	Doctors (Group)	222	0	11100	222011100

Table 2.1: Proposed ID Scheme

2.3 DTMF INPUT AND ASSOCIATED ACTION

Voice Message will be send and retrieved by PSTN/Mobile telephone set or IP telephone set [1,2,3,4,5] User calls to a node (HVMS), node answers the call, plays pre recorded message and asks for DTMF input. In response to the caller's DTMF input, information may be retrieved or any other service requests will be executed. System guides user with pre recorded digitized messages to select different options. After making call to a node, user must enter into message delivery or retrieval mode. User enters special characters/numbers to enter delivery or retrieval mode, a simple dial plan is given below:

DTMF	Action
111#	Delivery mode
222#	Retrieval mode

In voice delivery mode authentication is not needed, user will be asked to start voice message for recording. User needs to confirm recorded message and than will be asked for recipient ID. User can perform following by pressing appropriate keys in his keypad.

DTMF	Action
1#	Review recorded message
2#	Discard recorded message and record again
3#	Confirm recorded message
7#	Hear help message
8#	Exit Voice Delivery Mode
9#	Exit Voice Messaging System

When in message retrieval mode, he must provide his ID and password. After authentication the user may select the following options: play or delete messages in his message box, hear help message or quit. Each action has an associated special characters/numbers:

DTMF	Action
1#	Play first message
2#	Play next message
0X#	Play message number X (X is numeric digit), if message not found system will play the first message
3#	Delete mode and delete current message
7#	Hear help message
8#	Exit Voice Retrieval Mode
9#	Exit Voice Messaging System

Combination of these digits and associated actions will be customized according to need during implementation phase.

2.4 USER AUTHENTICATION

User must call to HVMS and authenticate himself before retrieving messages. We can authenticate user by plain text password file, system password file, access server or by using database system [18,19]. Plain text file authentication is not secured and we can ignore this method. We can write and use programming codes to authenticate user from system password file. We can also use authentication server like Radius/TACCAS, open source authentication servers are available in net. System password file is highly secured and suitable for stand-alone system but not suitable for our integrated system, query to remote database is less complex than query to remote system password file. Best way to authenticate a user for our proposed system is from database system. Query to a database is first and it's easy to write code to make query to a local or remote database system. Tables will be created to store user ID and password as shown below.

User ID	Password
001111111	999#888
002111111	123*123
222011100	321#123

Table 2.2: Database Authentication Table

2.5 GROUP MEMBERSHIP

A group can reside in a single node or a group can reside in multiple/all nodes. Every node can keep group ID and associated members ID within that system in database table. Its easy to make query to database tables to find out members ID of any group. Tables will be created with group ID name and group members ID will be stored in those tables.

Group ID: 222011100

Table Name: 222011100

User ID
001111111
001111112
001111113

Table 2.3 : Group Member ID

2.6 MESSAGE BOX LOCATION

Standard Database System will be used to store information of user voice message box location. Standard database systems are highly flexible and have enormous feature for software development. Query to a local database or remote database is easy, faster and secured. Database tables will be used to store voice messages folder location and any prerecorded message location. Message folder will be named according to user/group ID or user/group "Type Digits + Identification Digits". Implementation will be make OS independent by keeping message folder information in database. Folder path should have unique standard in every node else message uploading may become complicated. If any node uses folder path structure "/aaa/bbb/" than all other node should use the same path structure.

User/Group ID	Message Folder Location
001111111	/User_Voice_Folder/001111111
002111111	/User_Voice_Folder/00211111
222011100	/Group_Voice_Folder/22211100

Table 2.4: User/Group Message Folder location

2.7 NODE IP ADDRESSING SCHEME

Every node must have an associated IP address and it must know System Digits and associated IP addresses of all other nodes of the integrated system. We will use database system in every node to store information about system digits and associated IP addresses. Procedure to update database information will be customized according to need during implementation phase, update will be initiated by node administrator or will be initiated by a central database administrator

System Digits	IP Address
001	192.168.1.1
002	192.168.2.1
111	192.168.3 1
222	192 168.4.1

Table 2.5: Node IP Address Scheme

2.8 INTER NODE COMMUNICATION

We will design WAN connectivity structure to connect all the nodes with open standard TCP/IP protocol. Existing MAN, Radio, V-SAT, GPRS, EDGE and DDN infrastructure will be used to support major locations throughout the country [20,21,22,23]. We propose two different type of connection for every node, one as main link and one as backup link. Combination of existing WAN technology will be used according to geographic factors. Bandwidth allocation for any node is specific to no. of users of that node. Reasonable delay is acceptable as communication between nodes is not real time communication

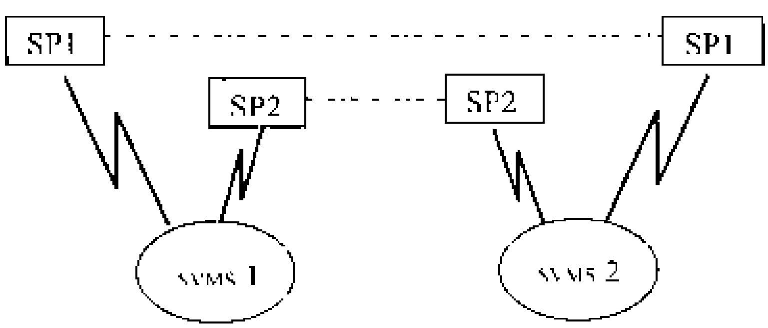

Fig 2.1: Inter Node Communication

2.9 MESSAGE SENDING

In our proposed system user access to HVMS is by PSTN/Mobile/VOIP telephony system User will access his HVMS in any convenient method, user will use PSTN or IP telephony or Mobile Network. For sending messages user authentication is not needed. User just need to give recipient ID after voice recording and rest will be done by the system. System will make query to find the location of the recipient home node by using system digits of the recipient ID. System will examine the type digit to find whether the message is for any group or for any single user. Message delivery within the system is easy, first three digits of recipient's ID will prove the user as a local user. We will create user's voice mail folder using user's "type digit + identification digits" and system will copy the message to user's home folder. In case of group messaging, we will find group identity by last five digits of the recipient's ID. We will keep ID of particular Group Members in database and by making query to the database system we will copy the message to members message box. Suppose, a user send a message to 121-1-44444, the local system will query the database service to find the IP address of system 121. Fourth bit is "1", means it's the ID of a normal user. Now, if the IP address is of the local system it will copy the message to the folder 144444 If the IP address is of a remote server it will upload the file to the remote server's 144444 folder.

For group messaging if the address is 121-0-44444, the server will copy the message to local 044444 folder or remote 044444 folder based on system identification digit "121". We will write program, that will check every 0xxxxx folder after particular time period to see whether any message reside. If any message is found it will make query to the database table to find the ID of the group members belongs to any particular group and will copy the message to those group members voice message folder. In the above example, system will find all group members ID belongs to 44444 group and will copy message to those users voice message boxes. If for any reason destination folder is not found, any intra system or inter system voice message delivery will be cancelled. Normal users will be allowed to send message to groups resides only in his HVMS but not to any group that resides in multiple nodes. Any message with leading 000 will go to every node of the system and sending such broadcast message should be done only by node's system administrator not by any normal user or group member. Successive broadcast message may bring network storm.

2.10 MESSAGE RETRIEVAL

Authentication from home node is not difficult but complexity arises with authentication from remote node. System will authenticate user by local authentication mechanism or remote authentication mechanism, it's easy to write code to make query to a local database system but complicacy arises with query to remote database system Suppose, any system Y needs to authenticate a user "AAA", system Y will examine first three digits of user "AAA" ID to locate the home node. If user "AAA" reside on the same node, query to the local system database is straight forward. System will find "AAA" voice message folder and will play voice file to "AAA".

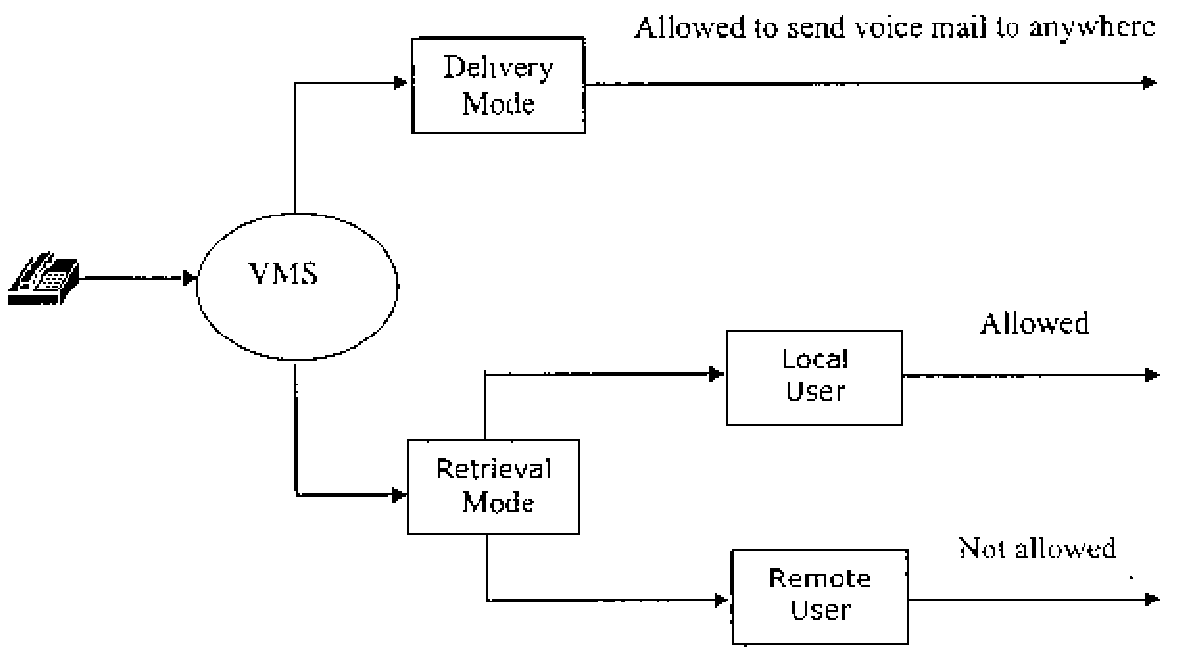

Fig 2.2: Message Retrieval

Problem arises with remote authentication, if user "AAA" has remote home node (System X), system Y will authenticate "AAA" by making query to system X. Remote authentication is not very difficult, but system Y also needs to provide voice messages information to user "AAA" of X after authentication. System Y will do it in two ways, it will copy the whole message folder of the user from the remote system or it will initiate an outgoing call and redirect the user to remote system. But both are not feasible, In first case copying the full directory is time consuming, any network interruption will halt the operation and will annoy user. Call redirection is possible, but unnecessary traffic will flow over the local system as every piece of information will be redirected through the local system. Call redirection also makes the system more complex.

We propose any user will send inter system messages but for message retrieval user must connect to local/home messaging server. User of any particular node will not be able to retrieve voice messages by log-in to another node but only from his local/home node.

2.11 SVMS FOR PROPOSED SYSTEM

We could have used existing commercial VMS as SVMS for our proposed system but proprietary nature and customization complexity of commercial systems prohibited us from such use. IVM is developed for Windows platform and it is suitable for single site implementation. Proprietary nature of Windows operating system is also a problem for considering IVM.

VOCP can be a candidate for our SVMS as it is developed on Linux platform and by open source tools. Source code of VOCP is open but technical support and any customization requires considerable amount of charges. VOCP is also designed for single site implementation

SoundBite's Voice solution are proprietary and closed system. Hardware based systems like Panasonic's VMS are pre configured system, suitable for single site small scale implementation and any customization is not possible. AVAYA don't have separate solution for VMS but bundled with IP Telephony or IP Call Centre Solution. Cisco Inc. has Unity System as VMS but number of users is predefined. Both AVAYA and Cisco solutions are for LAN specific implementation, very costly and not suitable for our Integrated System.

One of our goals is to develop our Proposed Integrated System by using open source platform and open source tools [24,25,26,27]. We can build SVMS for proposed Integrated system by using open source tools. Developing SVMS on open source platform and tools will reduce design and development complexity, maintenance and customization cost and dependency on third parties. Possibility of using TAPI to build our SVMS has been discarded due to proprietary nature of Windows operating system. JTAPI can be used to develop our SVMS as Java is supported on open source operating systems. IVRS and Modem::Vgetty both are open source tools for Linux/Unix environment. Full featured VMS and sophisticated voice application can be developed by using IVRS or Vgetty modules.

2.12 SCALABILITY AND ADAPTABILITY OF PROPOSED SYSTEM

Every node of our proposed architecture is independent of any other node, there is no direct link between nodes. Every node is connected to WAN cloud and addition or deletion of any node will not affect other nodes. Every user is registered to a certain SVMS and deletion of any SVMS will only affect user of that SVMS. Proposed ID Scheme is composed of nine digits and that will support 999 nodes and 99,999 users per node. Numbers of digits in our proposed ID Scheme can be increased to accommodate more nodes or users.

Our proposed architecture can adapt new technologies and changes. We have proposed to develop every SVMS with open standard tools and protocols. New features can be added to any node without affecting any other nodes due to independent architecture of nodes.

We can allow web access, outward dialing, message forwarding features to users of some particular node while only basic messaging features to users of other nodes. Communication between nodes depends only on user/group ID and node IP addresses, so nodes can be developed with different open source platforms and tools. We can use any of Modem::Vgetty, IVRS or JTAPI to develop SVMS in different sites or locations. Our proposed system is highly manageable due to distributed nature, problem in one node will not hinder operation of other nodes. Any node can be taken offline for reengineering, for maintenance or for troubleshooting without affecting other nodes.

CHAPTER 3

COMPONENTS AND FEATURES OF THE DESIGNED SYSTEM

3.1 INTRODUCTION

Like other Voice Messaging System [6,7], every SVMS of our integrated system has following basic components :

1. Access Component
2. Application Component
3. Database Component
4. Messaging Server and Storage Components

3.2 ACCESS COMPONENTS

Access components provide access to Application Components and the front-end user tools of any Single Voice Messaging System. Subscribers will access Application Components with traditional telephony equipment, IPT or Mobile Network.

Access Components include the following components:

- PSTN and its components
- IPT and its components
- Mobile Network and its components
- Analog to Digital and Digital to Analog conversion equipment

3.3 APPLICATION COMPONENTS

Application Components is the software that provides following features:

- User authentication
- Storing and retrieving messages

Application Component query standard database or system password file to authenticate user. Application Component uses the Database Services to find user/group information in the system. Application Component queries the database server with System Digit to find recipient Home Voice Messaging System.

1. If the recipient is local to the SVMS, Application Component delivers the voice message to the local voice message box.

2. If the recipient HVMS is remote, the Application Component performs the following tasks.

a) Finds the IP address of the remote SVMS

b) Establishes a TCP/IP connection to the remote SVMS

c) Uploads the message to the remote SVMS

SVMS can deliver voice message to any local user or to any roaming user. To deliver voice message to a roaming user Application Component must find the IP address of the remote SVMS, Application Component than retrieve message from remote SVMS. Application Component will complete the following steps to retrieve message from a remote node:

a) Application Component find the remote SVMS IP address by querying database.

b) Establish WAN connection with remote SVMS.

c) Retrieve the voice message from remote node.

3.4 DATABASE COMPONENTS

The basic function of a database in our proposed integrated system is to store and retrieve additional information by few digits of user id or by user input. User/Group details such as ID, password, Message Folder and any other information will be kept in standard database. Database tables will also be used to store node IP addresses. Application Components uses Database Tables to store, retrieve and update information. Free DBMS like MySQL will be used to store user/group ID and any other information.

3.5 MESSAGING SERVER AND STORAGE COMPONENT

SVMS needed to run on stable hardware and OS platform. Storage device is needed to store voice messages reliably and securely. Database Components also store database information in storage devices. Server hardware and storage device will be selected based on number of user and volume of messages. Storage redundancy is suggested for Messaging Server.

3.6 ADVANCED FEATURES OF PROPOSED INTEGRATED SYSTEM

Proposed Integrated Voice Messaging System can have many features [6,7] in addition to basic voice messaging service. Some of attractive features are:

1. Caller ID detection
2. Text-to-Speech Conversion
3. Group Messaging
4. Web Access
5. Message Forwarding
6. Outgoing Call

3.6.1 Caller ID Detection

Voice messaging system may have provision for caller ID detection. Voice messaging system may detect and use Caller ID to view the origin of incoming calls or filter calls based on the CID information. It is also possible to redirect a call based on caller ID.

3.6.2 Text-to-Speech Conversion

Text-to-speech and speech-to-text support can be added to a standard voice messaging system, can be used to convert emails to voice mail messages and voice mail messages to email. Integration of Fax service is possible due to this feature. Speech-to-text conversion technology can be used to provide postal like services in rural areas, voice message can be converted to text message in rural post offices and can be delivered to recipient address.

3.6.3 Group Messaging

Voice messaging system can have group messaging facility. A single message to the group ID will forward the voice messages to a number of different group member boxes. Group Messaging Facility can be used for agricultural sector, medical sector, weather forecast, disaster control sector etc.

3.6.4 Web Access

Voice messaging system can forward voice file to user E-mail address and can receive voice file attached with E-mail from the user. It's also possible to develop software to allow user to access his voice mail folder from internet. User email address can be kept in user information database and new voice mail can be forwarded to user email address.

3.6.5 Message Forwarding

Message forwarding allows messages left for one individual to be forwarded to another individual. It is very similar to mail forwarding, can be done manually or can be done automatically. User can have alias ID or user can ask node administrator to forward all his voice mail to some other user ID.

3.6.6 Outgoing Call

Voice Messaging system can place an outgoing call to a designated remote telephone number when a message is left in a user's mailbox. User telephone numbers can be kept in database and SVMS can originate outgoing calls to those number to deliver voice messages. Such outgoing calls can bring outgoing call cost on SVMS and may result toll fraudulent.

3.7 VOICE MESSAGING SYSTEM SECURITY

Provision must be made to secure SVMS from any unauthorized access [28,29]. Messages left on voice message boxes can contain critical information or corporate secrets. Gaining access to any SVMS or to any voice message box may permit an attacker to take away valuable information and by using any compromised box attacker can further attacks on other infrastructure component.

Data Networks are usually kept secured with use of OS firewall, dedicated firewall, IDS/IPS sensor etc. In our proposed system SVMS to SVMS connections are data network and tight security measures needed to be implemented. Uses of dedicated firewall is costly but we can use OS based firewall, every Linux/Unix OS comes with built in firewall like Iptables or PF. We can configure the SVMS to prevent logins of any user after a predetemined number of unsuccessful attempts. User password should be stored in OS password file or in database table, not in plain text file. We can enforce user to choose long password with combination of different digits and different signs. Confidential information must not be send by voice messages, if necessary such message must be deleted after reception. Following two events should be examined very carefully:

- Sign of unusual activity in any voice message box.
- Problem accessing the SVMS and password has been changed

Advanced feature like Call Forwarding and Call Transfer can bring risk of toll fraud [30]. Recommendation is not to implement Call Forwarding feature in our SVMS and if mandatory restrict calls only to local numbers.

CHAPTER 4

DEVELOPMENT OF PROTOTYPE SYSTEM

4.1 INTRODUCTION

We have developed a prototype distributed system to judge the possibility of our proposed Integrated Voice Messaging System. We have used Redhat Linux and US Robotics 56K Voice Faxmodem to develop and test our prototype system. We have tested followings in our lab:

- Voice message sending
- Voice message retrieval
- Intra System and Inter System voice message delivery

4.2 PLATFORMS AND TOOLS

Objective of this thesis is to develop nation wide distributed Voice Messaging System and integration of those VMS with PSTN and TCP/IP Network.

We have used Redhat AS 4 as our OS and Pentium 4 desktop pc as our voice messaging server. US Robotics 56K Voicefax modem has been used as our voice modem. We have used Modem::Vgetty voice module as our development tool and Perl was installed during the installation of Linux OS. Both system (system A and system B) was in similar platform and were connected directly to LAN switch.

4.3 VOICE MODEMS AND TELEPHONY CARDS

Voice Messaging System records sounds it hears from the phone line as voice file and we need voice modem or voice telephony cards to do so. Voice modem record and play audio over a telephone line. In addition to playing and recording audio many voice modems are capable of advanced features like DTMF detection, call transfers and outbound dialing. Example of voice modems are ZyXEL U1496, US Robotics Sportstar etc. It is possible to use PCI multi port serial interface card to allow multiple modem to connect to the same PC or Server.

Telephony card is very similar to Voice Modem but provide much better quality audio recording and output and may support multiple phone lines on a single card. Example of telephony cards are Octopus, Dialogic etc.

The modem/telephony card understands only rmd (raw modem data) format and record as rmd file. Voice format tool like pvftools (Portable Voice Format Tools) which is included with Vgetty has excellent utilities to convert sound files from one format to another format. Voice Modem or Telephone Card [10,11] should have some basic features as:

- It should initiate and terminate hardware perfectly
- It should receive and terminate call
- It should be able to play voice file and record voice file
- It should detect DTMF properly
- It should be capable of call forwarding, outward dialing
- The modem should detect bong tone, busy tone, call waiting, dial tone, handset on/off hook etc.

4.4 MODEM VOICE MODULE INSTALLATION

We have used Redhat Linux AS 4 as our OS and it comes with following rpm packages necessary for installing mgetty-1.1.31-2.i386.rpm and mgetty-voice-1.1.31-2 i386.rpm module. Perl development package was installed during installation of the operating system.

First, we made query to see whether these packages are installed with

rpm -qa mgetty

Packages were not installed, so we install with

rpm -ivh <package name>.rpm

Configuration file for Vgetty is /etc/mgetty+sendfax/voice.conf, we opened voice.conf with vi editor to make necessary changes.

First searchd for the line voice_devices, our modem was connected to ttyS0 port and we changed it to

voice_devices ttyS0

We searched for the line "rings 3" or a similar line, we wanted our modem to answer after three rings and we changed it to

rings 3

We searched for the line with "port", our modem port was ttyS0 and we changed it to

port tty S0

We kept default voice directories indicated in voice.conf by "voice_dir" keyword as /var/spool/voice, /var/spool/voice/incoming and /var/spool/voice/messages. In /etc/inittab we insert following line for invocation of vgetty by init during startup:

S1:345:respawn:/usr/sbin/vgetty ttyS0

Our vgetty log file was in default location /var/log/vgetty.ttyS0.log

We have named our calling program as answer.pl and placed it under /home/soheb/ directory. We gave appropriate input in voice.conf file

voice_shell /usr/bin/perl

call_program /home/soheb/answer.pl

4.5 MODEM VOICE MODULE METHODS

We have used some common Modem::Vgetty methods to develop our prototype system as discussed below

4.5.1 communication Initialization

We need to create Modem::Vgetty object that will initialize communication to the vgetty. communication can be terminated closed by the shutdown method:

use Modem::Vgetty;

my $v = new Modem::Vgetty;

communication can be terminated by the shutdown method:

$v->shutdown;

4.5.2 Voice Commands

beep ($freq, $len) – We can send a beep with frequency "$freq" (HZ) and length "len" (in miliseconds). vgetty Returns a defined value on success or undef on failure. The state of the vgetty changes to "BEEPING" and vgetty returns "READY" after a beep is finshed. Example:

$v->beep(50,10);

dial($number) – We can make outbound call and dial to a number

play($filename) – vgetty with play the file given in "$filename" record($filename) - vgetty with record voice as file given in "$filename"

wait($seconds) – modem will wait for "$seconds" and will return "READY" after waiting finished

$v->wait(5);

$v->waitfor('READY');

stop – vgetty stops current work

play_and_wait($file) - abbreviation of the following:

$v->play($file);

Sv->waitfor('READY'),

4.5.3 Event Handler Methods

add_handler($event, $handler_name, $handler) – This event handler will install a call-back routine "$handler" for the event type "$event". The first argument is the Modem::Vgetty object, second argument is the event name and third argument is optional event argument.

del_handler(Sevent, $handler_name) - This will deletes the handler given as second argument for the event as first argument.

enable_events – vgetty voice shell will start to dispatch event. No events are sent by default.

disable_events - vgetty voice shell will stop to dispatch event. No events are sent by default.

Some common events handled by vgetty are:

BUSY_TONE – Line is busy

CALL_WAITING - Line has received another call

DIAL_TONE – Modem has received dial tone on line

NO_DIAL_TONE - Modem has not received dial tone on line

HANDSET_ON_HOOK - Handset is on hook.

HANDSET_OFF_HOOK - Handset is off hook.

NO_ANSWER – Modem didn't detect answer for the time give in dial_timeout in voice.conf.

RING_DETECTED - Modem detected an incoming ring.

RECEIVE_DTMF - Modem detected a DTMF input code.

SILENCE_DETECTED – Modem detected no voice energy

VOICE_DETECTED - Modem detected voice energy on the line

UNKNOWN_EVENT – Modem detected unknown event.

4.5.4 The Readnum Method

readnum($message, $tmout, $repeat) -

With this method we can read user DTMF input. This routine will play voice instruction given as $"message" and accept sequence of the DTMF keys finished by the "#" key. If no key is pressed for "$tmout" of seconds, it re-plays the message again.

Number of repetation is controlled by "Srepeat" argument. It returns any available string irrespective of termination sign "#", after "$tmout" seconds or completion of repetation indiacted by "$repeat". DTMF tones are accepted during playing of message and if DTMF is received playing of the message is stopped.

4.6 PERL VOICE MODEM PACKAGE

Perl Modem package is an essential package for writing applications using voice modem or telephony. The answering machines and sophisticated voice applications can be written using this module. We have downloaded perl-Modem-Vgetty-0.03-1.2.el4.rf.noarch.rpm package and installed with

rpm – ivh perl-Modem-Vgetty-0.03-1.2.el4.rf.noarch.rpm

4.7 GNU WGET AND WPUT PACKAGE

We have used GNU wget and wput packages for uploading and downloading voice file to and from remote node. We have downloaded package wget-1.9.1-17.i386.rpm and wput-0.6.1-1.el4.rf.i386.rpm and installed with

rpm – ivh wput-0.6.1-1.el4.rf.i386.rpm

rpm –ivh wget-1.9.1-17.i386.rpm

4.8 CALLING PROGRAM TO VOICE MODULE

Main calling program to Modem::Vgetty for our prototype system was named as soheb.pl (Appendix A.1) and was kept under /home/soheb/ directory. We have used very simple programming logic for our prototype system and password authentication was not implemented, user is identified by user name. Script get_voice (Appendix A.2) to download voice messages from remote node and script put_voice (Appendix A.3) for uploading voice messages to remote node will executed from the main calling program. Both of these are shell script and placed under folder /home/soheb/voice/.

Prerecorded voice files for instructing users were stored under /var/spool/voice/messages/ directory and user's voice directory were created under folder /home/soheb/voice/. Three prerecorded voice messages are used to guide user in our prototype system. Prerecorded message "1.rmd" will guide user to choose voice retrieval mode or voice sending/delivery mode. Prerecorded message "2.rmd" is used to instruct user to provide user/group ID to the system. Prerecorded message "3.rmd" will instruct user to utter voice message for recording.

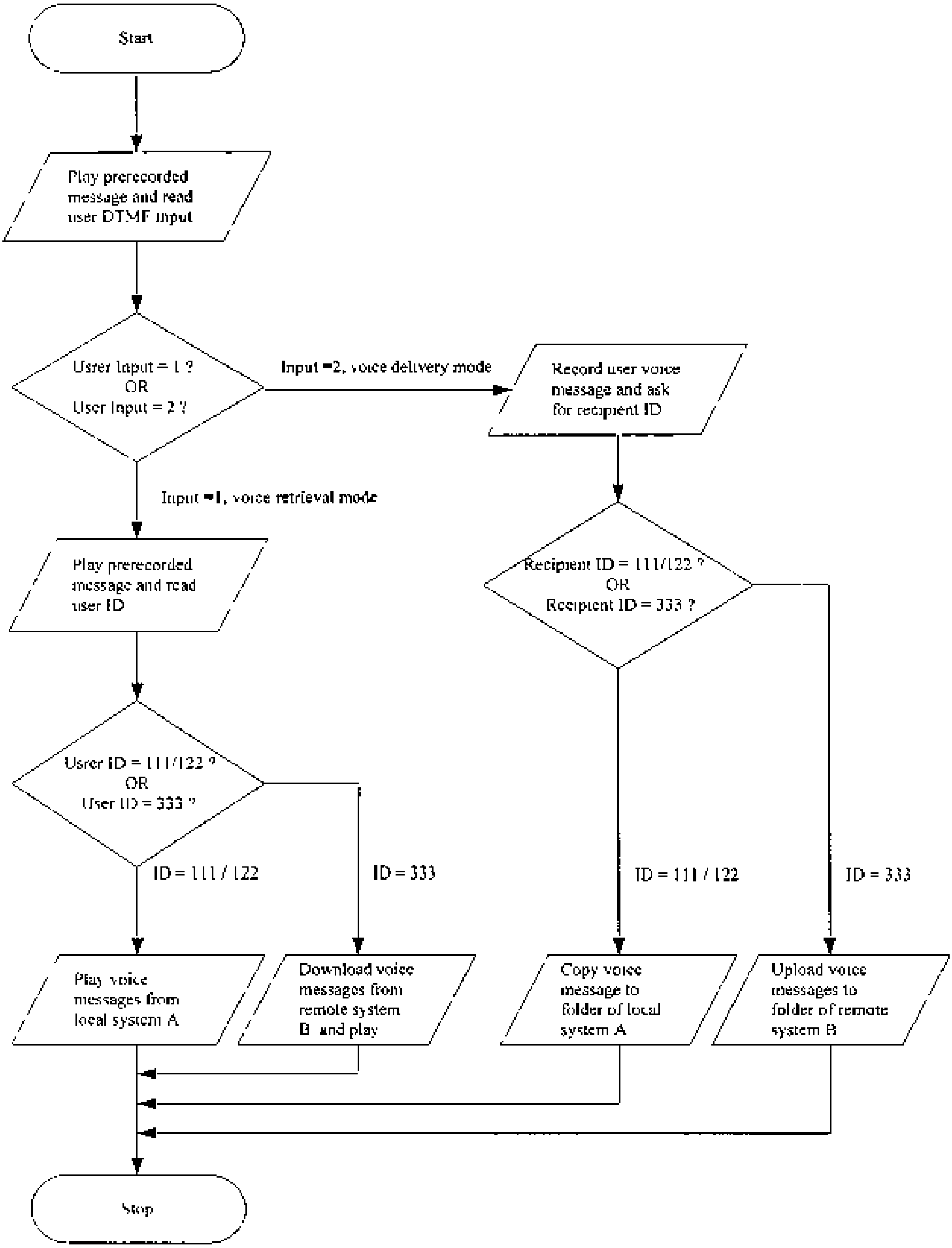

Fig 4.1: Flowchart of Calling Program for Prototype System

4.9 USER CREATION AND TESTING

To show possibility of our proposed system three users were created on two different Linux machine and both machine were in the same LAN as shown in the figure.

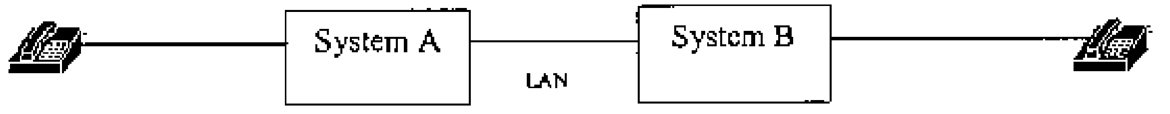

Fig 4.2: User Creation and Testing

User ID	System	IP Address
111	A	192.168.2.9
122	A	192.168.2.9
333	B	192.168.2.10

Table 4.1: User Information of Prototype System

Two users were created in system A with ID 111 and 122, voice folders were /home/soheb/voice/111/ and /home/soheb/voice/122/. System B has one user with ID 333, voice folder location was /home/soheb/voice/333/. All these voice folder has two subfolder named "in" and "out" for holding incoming and outgoing voice message. We have tested itra node messaging and inter node messaging with system A and system B.

4.10 RESULTS AND DISCUSSION

4.10.1 VOICE MESSAGE SENDING

To check message sending capability of our prototype system, we first dialed to system A. Prerecorded message instructed us to enter either in voice retrieval mode or in voice delivery mode. We pressed key '2' from our telephone dial pad as per instruction to enter into voice delivery mode. Another prerecorded message asked us to start our voice message for recording and 10 seconds were allowed for our voice recording. Pre recoded message asked us to enter recipient ID and we enter "111" as recipient ID. We repeated the cycle in system A to send message to user 122 and did the same in system B to send message to user 333. From CLI we have found voice messages were recorded and delivered successfully in both systems.

4.10.2 VOICE MESSAGE RETRIEVAL

We dialed to system A and pressed key '1' from our telephone dial pad as per spoken instruction to enter into voice retrieval mode. According to instruction we entered "111" as our ID and message sent for user 111 in "Message Sending" step was listened successfully. We repeated the same in system A with user ID "122" and message sent for user 122 was listened successfully. Next we dialed system B to hear the message from "in" box of user user 333 and message was retrieved successfully.

4.10.3 INTER SYSTEM MESSAGE DELIVERY

We entered into voice delivery mode in system A and entered recipient ID "333" after recording of voice message. User 333 was a user of system B and according to our prototype system design, script put_voice should execute and deliver the message to remote system B. Next we entered into voice retrieval mode in system B with user ID "333" and found message recorded in system A was successfully delivered to System B.

To show possibility of downloading voice message for any roaming user, script was written to download message of user 333 (User of machine B) in System A. We logged in to System A in voice retrieval mode, we gave ID "333" to download message from message folder of user 333 in system B to temporary folder of system A. We found script get_voice executed accordingly, downloaded message from system B to system A and the message was listened successfully.

4.10.4 COST EFFECTIVENESS OF PROPOSED SYSTEM

Our integrated system is of distributed architecture and every node is independent of other nodes. Number of ports in any particular SVMS will depend on number of users/groups belong to that SVMS and will be changed later. SVMS developed with open source tools are highly configurable as source code is open and reengineering is not needed as every system will be developed according to our requirements. We have estimated, budgetary cost OF main unit of 16 port SVMS FOR unlimited users as given below :

Product	Unit	Unit Price (BDT)	Total Price (BDT)
Quad Core Xeon Server, 2 GB Ram, 2 x 146 GB SAS HDD with Raid Controller, Linux OS	1	250,000.00	250,000.00
Cyclades/TELCTAS 16 port Serial PCI adapter	1	90,000.00	90,000.00
US Robotix Voice Modem or Equivalent	16	5,000.00	80,000.00
Estimated Cost:			420,000.00

Table 5.1: Estimated cost for SVMS developed with open source tools

VOCP is based on open system and it can be downloaded and used for free. License charge is not required but support and customization fees are charged according to requirements. VOCP system is designed for single system and customization is required for our Integrated Voice Messaging System.

We also have calculated cost needed to implement commercial product like IVM as our node. IVM can be used with failover option but IVM is not developed for distributed architecture. IVM needed to be customized to align with our architecture. Cost of such reengineering of commercial product is very high. Estimated cost of IVM for unlimited users is given below:

Product	**Unit**	**Unit Price (BDT)**	**Total Price (BDT)**
Quad Core Xeon Server, 2 GB Ram, 2 x 146 GB SAS HDD with Raid Controller, Windows OS	1	320,000.00	320,000.00
Cyclades/TELCTAS 16 port Serial PCI adapter:	1	90,000.00	90,000.00
US Robotix Voice Modem or Equivalent	16	5,000.00	80,000.00
Support/Year	3 years	300,000.00	900,000.00
Estimated Cost:			1,390,000.00

Table 5.2: Estimated cost for IVM

CISCO Unity is unified messaging solution suitable for LAN implementation, cost is very high and require purchasing license based on number of users. CISCO Unity implementation requires additional high end router, FXO/FXS cards to connect with telephone service provider and high capacity LAN switches. CISCO or AVAYA voice messaging solution is not suitable for our implementation. Estimated cost of CISCO Unity for 300 users is given below:

Product	Unit	Unit Price (BDT)	Total Price (BDT)
Quad Core Xeon Server, 2 GB Ram, 2 x 146 GB SAS HDD with Raid Controller, Windows OS	1	320,000.00	320,000.00
Unity Release 5	1	0.00	0.00
Unity UM Exchg, 300 users, 16 session	1	2,700,000.00	2,700,000.00
Message Store 2003	1	0.00	0.00
SP Essential SW Unity Rel 5	3	0.00	0.00
Support	3 years	144,000.00	432,000.00
Estimated Cost:			3,452,000.00

Table 5.3: Estimated cost for CISCO Unity

CHAPTER 5

CONCLUSION

5.1 CONCLUSION

Main objective of this thesis was to show the possibility of developing Integrated Voice Messaging System for mass people by using non proprietary resources. Basic idea was to develop Single Voice Messaging System by open source tools and than to interconnect those Single Voice Messaging System to build up our Integrated Voice Messaging System.

We have analyzed different possibilities and shows possibility of developing large scale Integrated Voice Messaging System with open source tools. To show certainty of our proposed system, we have developed a prototype integrated system with two voice messaging system. Simple authentication mechanism and simple programming logic were implemented to reduce complexity in prototype system. Three users were created in two systems and inter system and intra system message delivery and retrieval was done successfully. Voice message for roaming user can be downloaded from any node other than user home has been shown in prototype system. But recommendation is not to give such facilities to any roaming user as it will increase bandwidth contention in WAN connectivity.

In our proposed design operation of every node is independent. If two nodes can communicate successfully with one another, multiple nodes will be able to communicate successfully with each other. Design of our proposed system is highly flexible and any number of nodes can be added or subtracted without affecting other nodes.

We have found some problem with DTMF detection with "readnum($message, $tmout, $repeat)" method. If telephone keys are pressed while prerecorded message are played, DTMF should be detected properly. But we have found DTMF are not received properly if we press key before finishing of pre recorded message. We are working to solve this issue.

5.2 FUTURE WORK

Further work based on our proposed architecture can be carried out. A few areas for further work is outlined below

1) Highly efficient Integrated Voice Messaging System can be developed with proposed architecture and tools. Uniqueness of each message can be assured by adding time stamp and sender ID with name of the message file. Incoming Message and Outgoing Message for each user can be stored in different folders and application can be developed to retrieve messages based on user DTMF input. Prerecorded voice messages can be added in the system for guiding user for telephone key pad digit selection.

2) Advanced Voice Messaging System features like caller ID detection, Web access, Message forwarding can be implemented with our proposed system. Speech to text conversion facilities can be added with our proposed system to provide traditional mail like facilities in post offices. We have proposed not to allow any user from retrieving message from non home node, if in future high WAN bandwidth become available and affordable message retrieval from non home node can be allowed.

3) Perl is a very strong high level programming language and efficient program can be written for text manipulation, file manipulation, system software and management tasks, database application, graphical programming, networking and web programming. IVRS functionality of VMS can be used to add additional services to the integrated system as: user commodity billing information, offline help services, social services, users mandate on common issues etc.

REFERENCES:

[1] Community Voice Mail,
www.cvm.org/whatcvm/whatiscvm.htm

[2] The University of IOWA, "Voice Telephone User Guide", 2004,
www.its.uiowa.edu/tns/voice/guides.htm

[3] Panasonic, "Voice Mail System",
www.panasonic.com/consumer_electronics/communication_systems/

[4] IVM Phone Answering System, www.nch.com.au/ivm/

[5] VOCP System, http://www.vocpsystem.com/

[6] CISCO and British Telecom, "Communications technology bridges the digital divide in India", A Case Study LifeLines India, 2007
www btglobalservices.com/business/global/en/docs/case_studies/lifelines_oneworld_india_case_study_en.pdf

[7] One World South Asia, "A Case Study on LIFELINES INDIA, VOICE ENABLED INFORMATION DELIVERY SERVICES TO BRIDGE KNOWLEDGE DIVIDE OF RURAL FARMING COMMUNITIES IN INDIA", August 2007
http://www.bcoalliance.org/system/files/070817LifelineIndiaCaseStudyOW.pdf

[8] Community Voice Mail for Routine and Disaster Services, "TeleMinder",
http://grants.nih.gov/grants/funding/sbir_successes/decision_2.pdf

[9] IVM Phone Answering System, www.nch.com.au/ivm/

[10] CISCO Systems Inc, Unified Messaging
http.//www.cisco.com/en/US/docs/ios/solutions_docs/voip_solutions/UM_ISD.html

[11] Microsoft MSDN, "TAPI 2.2 Reference", http://msdn2.microsoft.com/en-us/library/ms737219(VS.85).aspx

[12] Sun Microsystems, Java Telephony API (JTAPI), http://java.sun.com/products/jtapi/

[13] Mukund Deshmukh, "IVRS", http://search.cpan.org/~mukund/Ivrs-0.24/Ivrs.pm, http://backpan.perl.org/authors/id/M/MU/MUKUND/Ivrs-0.03.readme

[14] Jan "Yenya" Kasprzak, "Modem::Vgetty", http://search.cpan.org/~yenya/Modem-Vgetty-0.03/Vgetty.pm

[15] Wojciech Mazurczyk and Zbigniew Kotulski, "Lightweight Security Mechanism for PSTN-IP Telephony convergence", *IJCSNS International Journal of Computer Science and Network Security*, VOL.7 No.5, May 2007

[16] Wojciech Mazurczyk, Zbigniew Kotulski, "Lightweight security mechanism for PSTN-VoIP cooperation", Faculty of Electronics and Information, Polish Academy of Sciences, Institute of Fundamental Technological Research, 2006

[17] Douglas E. Comer, "*Internetworking with TCP/IP, Vol. I: principles, Protocols and Architectures*", Fourth Edition, Prentice-Hall of India

[18] H.T. Kung, F. Zhu and M. Iansiti, "A Stateless Network Architecture for Inter-enterprise Authentication, Authorization and Accounting", Harvard University, *Proceedings of the Third International Conference on Web Services (ICWS'03)*, June 2003, pp. 235-242

[19] Cisco Systems, "Network Security Policy: Best Practices White Paper", Document ID: 13601, Jul 09, 2007

[20] Kalyan K Mallik, "Some observations and recommendations for spreading integrated telecommunications infrastructure in rural Bangladesh", Electrical Engineering, Jadavpur University, Kolkata, India

[21] Abdul Hasib, "PROSPECT OF WIRELESS MAN IN RURAL NETWORK INFRASTRUCTURE OF BANGLADESH", Institute of Information and Communication Technology, Dhaka, Bangladesh

[22] Dr. S. M. Lutful Kabir, "INTERNET AND DATA COMMUNICATION INFRASTRUCTURE IN BANGLADESH", Institute of Information and Communication Technology, Dhaka, Bangladesh

[23] Sanim Alam, "TELECOMMUNICATIONS INFRASTRUCTURE OF BANGLADESH", American International School/Dhaka, Senior Project, 2004

[24] Paul P. Komkoff, Anna Anikina, Roman Zhnichkov, "Linux Telephony A Short Overview", *Proceedings of the Linux Symposium*, June 27th–30th, 2007

[25] Michael Weiss, Gabriella Moroiu, Ping Zhao, "Evolution of Open Source Communities", 2006, *The Second International Conference on Open Source Systems*, June 8-10, 2006

[26] Stefan Koch, "Evolution of Open Source Software Systems – A Large-Scale Investigation", *Proceedings of the First International Conference on Open Source Systems*, Genova, July 11-15, 2005

[27] Barbara Russo, Chiara Braghin, Paolo Gasperi, Alberto Sillitti, Giancarlo Succi, "Defining the Total Cost of Ownership for the Transition to Open Source Systems", *Proceedings of the First International Conference on Open Source Systems*, Genova, July 11-15, 2005

[28] Cisco Systems, "Network Security Policy: Best Practices White Paper", Document ID: 13601, Jul 09, 2007

[29] Kingpin, "Compromising Voice Messaging System", Case Study by Stake Inc. http://pdf.textfiles.com/academics/compromising_voice_messaging.pdf

[30] Time Warner Telecom, "TOLL FRAUD SECURITY TIPS", Network Operations Center

Appendix A

PERL PROGRAM AND BASH SCRIPTS FOR PROTOTYPE SYSTEM

A.1 Calling Program to Modem::Vgetty

```
#!/usr/bin/perl

use Modem::Vgetty;

my $voicemaster = 'root@localhost';

# Declare Voice Directory

my $voicedir = '/var/spool/voice/messages';
my $voice = '/home/soheb/voice';

# Declare Variables

my $x = "1";
my $y = "333";
my $tmout = 5;
my $finish = 0;

# Declare new Vgetty instance

my $v = new Modem::Vgetty;

# Declare Event Handler for busy tone silence etc.

$v->add_handler('BUSY_TONE', 'finish', sub { $v->stop; $finish=1; });
$v->add_handler('SILENCE_DETECTED', 'finish', sub { $v->stop; $finish=1; });
local $SIG{ALRM} = sub { $v->stop; };
$v->enable_events;
```

```
# Play instruction to user and take user DTMF input, 1 for Voice retrieval and 2 for
voice delivery

$option = $v->readnum("$voicedir/1.rmd",10,2);

# make a 50 Hz, 10 mili seconds beef to user

$v->beep(50,10);

# make the modem ready for hearing

$v->waitfor('READY');

# Check whether vgetty is busy

if ($finish == 0) {

# User input is 1 and user wants to retrieve message

if ($option eq $x) {

# Ask user to give ID and take DTMP input

$id = $v->readnum("$voicedir/2.rmd",10,2);
$v->beep(100,10);
$v->waitfor('READY');

# If ID is 333, remote home node user, run get_voice script to download voice
message to temporary folder and play

if ($id eq $y)
{
system "/home/soheb/voice/get_voice";
$v->play_and_wait("$voice/temp/0.rmd");
}
```

```
# IF ID is 111 or 122, user is local and play from his voice message directory

Else
{
$v->play_and_wait("$voice/$id/in/0.rmd");
$v->beep(100,10);
$v->waitfor('READY');
}
}

# User input is 3 and user wants to record message

Else

# Ask user to start uttering message

{
$v->play_and_wait("$voicedir/3.rmd");
$v->beep(100,10);
$v->waitfor('READY');

# Record the message to a temporary location

$v->record("$voice/temp/0.rmd");
alarm(8);
$v->beep(100,10);
$v->waitfor('READY');

# Ask user to give ID and take DTMP input

$other_id = $v->readnum("$voicedir/2.rmd",10,2);

# If ID is 333, recipient is remode node user, run put_voice script to send voice
message to remote node
```

```
if ($other_id eq $y)
{
system "/home/soheb/voice/put_voice";
$v->beep(100,10);
$v->waitfor('READY');
}

# If ID is 111 or 122, recipient is local, just copy the message to user voice box

else
{
system "cp $voice/temp/0.rmd $voice/$other_id/in/";
}
}
$v->beep(100,10);
$v->waitfor('READY');
}

# kill vgetty instance and exit from the script

$v->shutdown;
exit 0;
```

A.2 Bash Script for Downloading Voice Message

```
#! /bin/bash
cd /home/soheb/voice/temp/
wget ftp://soheb:test123@172.16.12.136/voice/333/in/0.rmd
chmod 777 ./0.rmd
cd /home/soheb/
```

A.3 Bash Script for uploading Voice Message

```
#! /bin/bash
wput --reupload /home/soheb/voice/temp/0.rmd
ftp://soheb:test123@172.16.12.136/voice/333/in/0.rmd
```