VIDEO CALLING & LOCATING APPLICATION
(FOR ANDROID SMARTPHONES)

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THE COMPUTER SOFTWARE ENGINEERING, MILITARY COLLEGE OF SIGNALS,
NATIONAL UNIVERSITY OF SCIENCES & TECHNOLOGY, RAWALPINDI
IN PARTIAL FULFILLMENT FOR THE REQUIREMENTS OF A B.E. DEGREE IN COMPUTER
SOFTWARE ENGINEERING
JUNE 2012

ABSTRACT

VOCAL is an Android mobile application which can be used to make video calls as well as to locate other users through Google Maps. It will allow users to stay in touch with one another throughvideocalls. Users will addothers in their buddy lists so that they mayreconnect with them at any later time. Besides video calling,users can also locate each other through Google Maps.

This project focuses on developing a Java application which is to be installed on Android smartphones. Once the user starts the application he has an option ofregistering or loggin-in.After a user is logged-in, he can view other users in around him in a selected radius. Such users'location-coordinated are obtained from a central database which are then displayed as blips on Google Maps.Then the user can pick anyone from map or a buddy already in buddy list to initiate a video call with him.

There are two main modules of VOCAL: Video Call, and Location Tracking.

To implement **Video Call**, we have used Dubango Framework. For resolving IP address NATing issue,we have used STUN Protocol.A SIP Server and SIP have also been used.SIP is an application-layer control (signalling) protocol which is meant to create, modify and terminate sessions between participants. Such sessions may include Internet telephone calls and multimedia conferences. SIP invitations, which are used for establishing sessions, carry session descriptions which allow participants to agree on a set of compatible media types. SIP usesProxy Servers to help route requests to the user's current location, authenticate and authorize users for services and implement call-routing policies. SIP also provides a registration function which allows users to upload their current locations for use by Proxy Servers. SIP runs at top of several different Transport Layer Protocols.

Location Trackingin our application is implemented using Google Maps, which are embedded in our application through Google Maps APIs. Blips are shown on the freshly captured GPS coordinates of users within a set radius, and also on the location of buddies (whose coordinates are got from a DB Server).

CERTIFICATE OF CORRECTNESSAND APPROVAL

It is certified that the work contained in this project report (prepared by IqraSabir, UmairAyub, Muhammad TehseenAsif and SharoonJiwan Mall, under the supervision of Lecturer Bilal Rauf) for partial fulfilment of Degree of Bachelors of Computer Software Engineering is correct and is approved.

APPROVED BY

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(BILAL RAUF)

DEPARTMENT OF CSE

MCS

DATED: _______________

ACKNOWLEDGEMENTS

We dedicate our work to our families who had to bear our repeated and prolonged absence and had to miss our attention during this project. We are thankful to our teacherstoo, who enabled us to reach a point where we could embark at, and then complete, this project.We should not forget to thank our colleagues alsowho have, from time to time, been great sources of help, encouragement and creative ideas. There were certainly many times when we felt stuck but at these times we found very wonderful ideas and useful help came sometimes from these friends and sometimes from people like Mr. Bilal Rauf, our Project Supervisor. His kind counsel indeed kept us on the right track.

We also thank all the staff of our department: the Lab Engineers, technicians, clerks, peons and sweepers. They all contributed towards keeping us comfortable, playing their respective parts at different times.

In the end, we apologize if we have forgotten to acknowledge someone’s efforts, and assure of our most sincere gratitude for their contribution in our project.

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CHAPTER 1

INTRODUCTION

General Description

The project aims at developing a mobile application (Android based) that will allow users to communicate through video calls and track one another's location (in a specified area, by using GPS) on Google Maps and to form a social network. This social network will have following features:-

Video calling.

Displaying blips on locations of other application users within a selected radius (keeping the default radius for display at 1km).

Location acquisition using GPS.

User profile i.e. name, age and picture.

Messages. (Extended Goal)

Settings. (Extended Goal)

Definitions, Acronyms and Abbreviations

This document uses various terminologies for referring to different technologies, techniques and network components. These are briefly described below. Readersmay like to find information in greater detail from the given references.

CODEC (Coder-Decoder): CODEC is a type of software for encoding (compressing) and decoding (decompressing) digital media files (songs / videos). Different available media players are used for creating and then playing such digital media files. Windows Media Player 12 is one popular CODEC.

EDGE (Enhanced Data Rate for GSM Evolution): EDGE is a mobile phone technology of GPRS family which is standardized by 3GPP (described below). It allows a high bit-rate data transmission (three time better capacity and performance as compared to GSM / GPRS connections). This is achieved through sophisticated methods used for data compression and transmission. A benefit of EDGE is that is conserves the available bandwidth by loading it with lighter (highly compressed) data.

GPL (General Public License): The GNU General Public License (GNU GPL or simply GPL) is the most widely used free software license.

GPRS (General Packet Radio Service): It is a standard for wireless communications which runs at speeds up to 15 kilobits per seconds (kbps). It supports a wide range of bandwidths and is suited for sending small bursts of data e.g. emails and web browsing content, as well as data of large volume. This standard is now maintained (specified) by 3GPP.

GSM (Global System for Mobile Communication): It is a standard for wireless communications which runs at speeds up to 9.6 kbps. It was developed for the 2nd generation (2G) digital cellular networks which replaced the old analog AMPS technology. Many mobile phone sets still work on this technology, although it has been succeeded by the 3G Technology.

HTTP(Hyper Text Transfer Protocol):[1] HTTP is the most widely used network protocol in the World Wide Web. It is a stateless, request response protocol in a client – server computer architecture.[2] Through HTTP request and response messages, web content is exchanged between client and server systems.[3]

LGPL (Lesser General Public License):LGPL is a free software license published by the Free Software Foundation.

Media Server:It is a dedicated machine or a software meant for storing and sharing media (video) content to client machines. It provides real time media processing functions.[4] Hence, it is used to streaming video between two nodes. Usually it is web server of a large internet website but it may be a specialized application also which enables users to remotely access media files over the internet.

NAT (Network Address Translator): NAT is a method by which IP addresses are mapped from one realm to another.[5] NAT is necessitated due to growing shortage of IP addresses and scaling of routing.[6] An advantage of NAT is that the IP addresses of systems within a particular domain, can be used by several other systems inside another domain. It is a router function so a dedicated router is configured to replace the IP address (in the headers of outgoing messages of systems inside a domain) with that of its own IP, and likewise for the incoming traffic.

RTP (Realtime Transport Protocol): **RTP defines a standardized packet format for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications and web-based push-to-talk features.**

SIP (Session Initiation Protocol):SIP is an application-layer control (signaling) protocol which is meant to create, modify and terminate sessions between participants.[8]

UML (Unified Modeling Language):UML is a visual language for specifying, constructing, and documenting the artifacts of a system. Most of the complex software designs are hard to be described textually. These can readily be conveyed through diagrams using UML.

3G (3RD Generation Mobile Telecommunication):3G is the third-generation of mobile phone technology standards. Typical services associated with 3G include wireless voice telephony and broadband wireless data, all in a mobile environment. With the capability for high-speed wireless data transfer, 3G has enhanced / enabledseveral additional applications such as mobile video, secure mobile ecommerce, location-based services and mobile gaming.

3 GPP(3RD Generation Partnership Project):3GPP is a collaboration between different groups for the purpose of preparing, approving and maintaining Technical Specifications and Technical Reports for GPRS, 3G, EDGE and the later technologies of mobile phones.[7]

Background

VOCAL is intended for the present and potential users of Android based cellular phones (in particular) and all cellular phone users (in general) who are interested in video chatting with their friends. The target market will further include users who like to broaden their friend circle by spotting other users of this application located within a specified radius. VOCAL is a mobile application for Android smartphones that will provide video call feature on more economical devices as compared to the expensive smart phones like iPhone. Moreover, the feature of location tracking will add another exciting experience. Unlike Android based version of Skype and a similar service by Apple which are available only on wifi, our solution will provide the same service over Edge. Our product is constrained in quality of video due to limitation of bandwidth over wireless medium. Due to this constraint, a lot of compression will be required, which, with the present compression techniques, is not feasible. So it will have to be achieved by reducing the frame rate and image resolution. However, this issue may be addressed with evolving efficient compression algorithms.

Problems Addressed

Although similar applications are available in market but none integrates all the features which VOCAL will provide. Our product particularly addresses the following two issues:

Quality: Main aim is to provide a quality software that will allow users to make video calls with less interruption and jitters. Our project will be the first calling software that will provide a video call over wifi / EDGE.

Location: VOCAL will provide a location of the users using Google Maps on the basis of which a user can call other users and track their location.

Business Goals and Objectives

To understand comprehensively and implement the concepts of networking and Androidoperating system, getting familiarized with programming to operate Android based mobile devices and understand the concepts of distributed computing.

VOCAL will be an attractive application that will efficiently fulfill its claimed functionalities by quick access and updates. Success will be measured by the number of users who choose to register to this application and by encountering lesser number of bugs after deployment. Since it is not intended to be kept open-source, another important metric of success will be its being purchased by cellular companies e.g. Mobilink, Ufone. If this materializes, financial benefit is expected.

Similar applications are already available in iPhones but the developers are hopeful of success because it will offer solution with added features on relatively more economical mobile phone sets. It is also expected to attract users by offering a new platform for social networking in which users will take interest in connecting due to their location. The user population is expected to be particularly interested in knowing where their chat / video-call partners are located.

The end goal of developing VOCAL is a working Android application that provides location based video chatting.

CHAPTER 2

LITERATURE REVIEW

Past work in the Domain

A major development that started in 2004 was the introduction of mass-market Voice over IP (VoIP) services that utilize existing broadband internet access, by which subscribers place and receive telephone calls in much the same manner as they would via the PSTN. Full-service VoIP phone companies provide inbound and outbound service with Direct Inbound Dialing. Many companies offer unlimited domestic calling for a flat monthly subscription fee. Sometimes this includes international calls to certain countries. Phone calls between subscribers of the same provider are usually free when flat-fee service is not available.

A VoIP phone is necessary to connect to a VoIP service provider. This can be implemented in several ways listed below:

- **Dedicated VoIP Phones:** Dedicated VoIP phones connect directly to the IP network using technologies such as wired Ethernet or wireless Wi-Fi. They are typically designed in the style of traditional digital business telephones.
- **Analogue Telephone Adapter:** An analogue telephone adapter is a device that connects to the network and implements the electronics and firmware to operate a conventional analog telephone attached through a modular phone jack. Some residential Internet gateways and cable modems have this function built in.
- **Soft Phone:** A soft phone is application software installed on a networked computer that is equipped with a microphone and speaker, or headset. The application typically presents a dial pad and display field to the user to operate the application by mouse clicks or keyboard input.

Our aim is to build an application on mobile phone that will act as a soft phone and provide video calling functionality / feature.

Existing Systems

- **Ready-Made Products:** Similar products, like VoIP Stunt, are available for different mobile devices, even if they don’t offer all the functionality which our application will provide.
- **Reusable Components:** Our team will rely on SIP Server to provide the SIP Connectivity.
- **Products that can be referred to:** Skype and similar high quality programs should provide development guidelines, since they are market leaders. Sip Droid is a popular and complete Android Sip Client from which we can get ideas.

Shortcomings / Issues in Existing Systems

Quality: Voice over IP is the integration of VoIP architecture and internet technology. One of the greatest disadvantages of Voice over IP is the quality of voice delivered. The other main drawback is the need for a reliable broadband and a fast internet connection.Without this, a Voice over IP solution is probably not good enough for anyone to use.

Power Management: Power outages or electricity failure present another imminent hurdle. Since Voice over IP requires a live Internet connection, losing electricity means losing your phone service. If you are concerned about losing power, you could install a UPS for your cable/DSL modem and Analogue Telephone Adapter (ATA). Even a small UPS would keep those devices powered for several hours. Alternatively, you could keep a single phone line with no premium services as a lifeline for those times that you might lose your power or internet connection.

Compression: The voice quality of a VoIP call is another consideration. Since the data travels across the internet, there is a potential of dropouts or jitters similar to what one might experience on a cell phone. Managed IP networks,

used by all the VoIP providers use quality of service (QOS) mechanisms built into their ATAs, and a high-quality compression algorithm (G.711) yields call quality that greatly exceeds the computational power of cell phones. Although it is getting better all the time, the quality of most VoIP services and products can't yet match that of PSTN. There are inherent challenges in sending a voice stream over a packet network.

Emergency (SOS) Calls: Emergency calls are another challenge for VoIP telephony, because unlike traditional telephone it's difficult to determine the exact location of an IP address with geographic certainty. Without knowing the exact location, it's also difficult to determine which call center should receive a particular VoIP originated SOS call (though most VoIP providers do have the SOS call routing based on the address you used when you registered).

Common Software Required: Because of VoIP's architecture, some services require both the caller and the called party to subscribe to their service, and some software programs require that both parties have the same software installed.However, there are other services/programs that allow a user to call anyone, including calling from phone to phone with packets routed over IP network.

CHAPTER 3

SYSTEM REQUIREMENTS SPECIFICATION

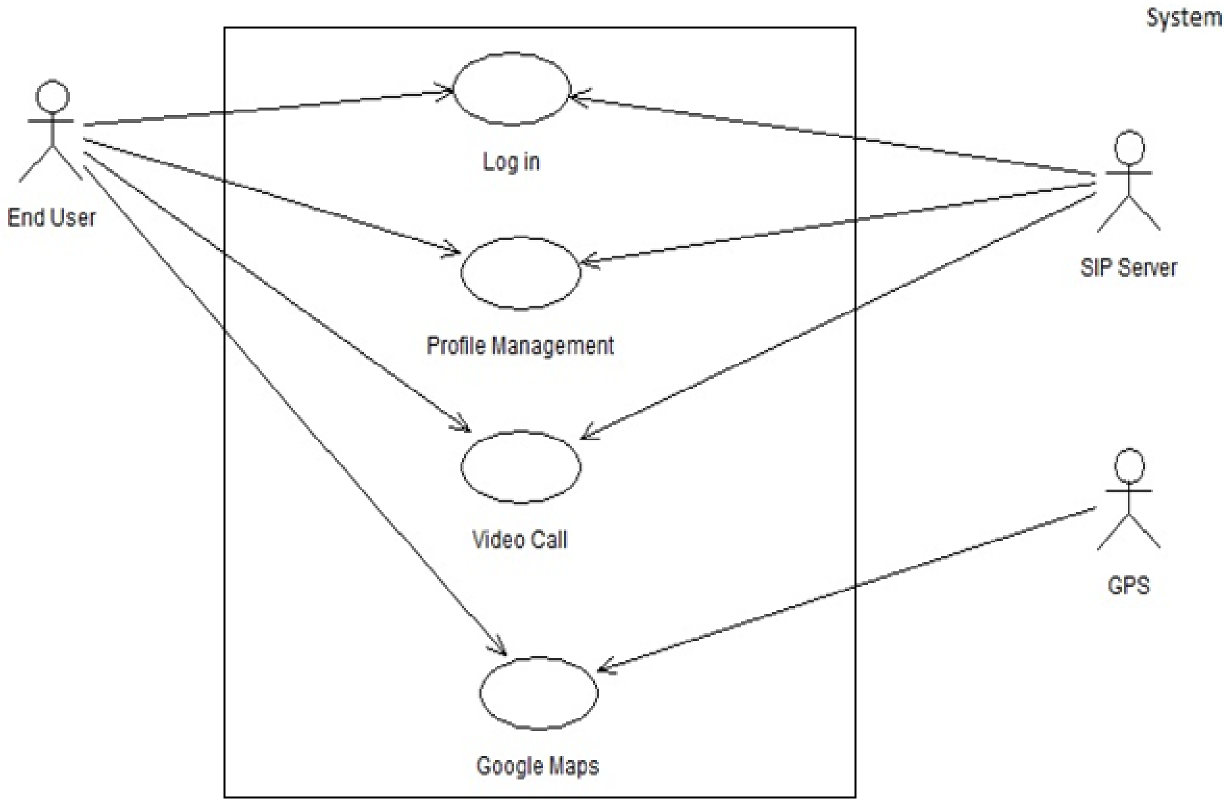
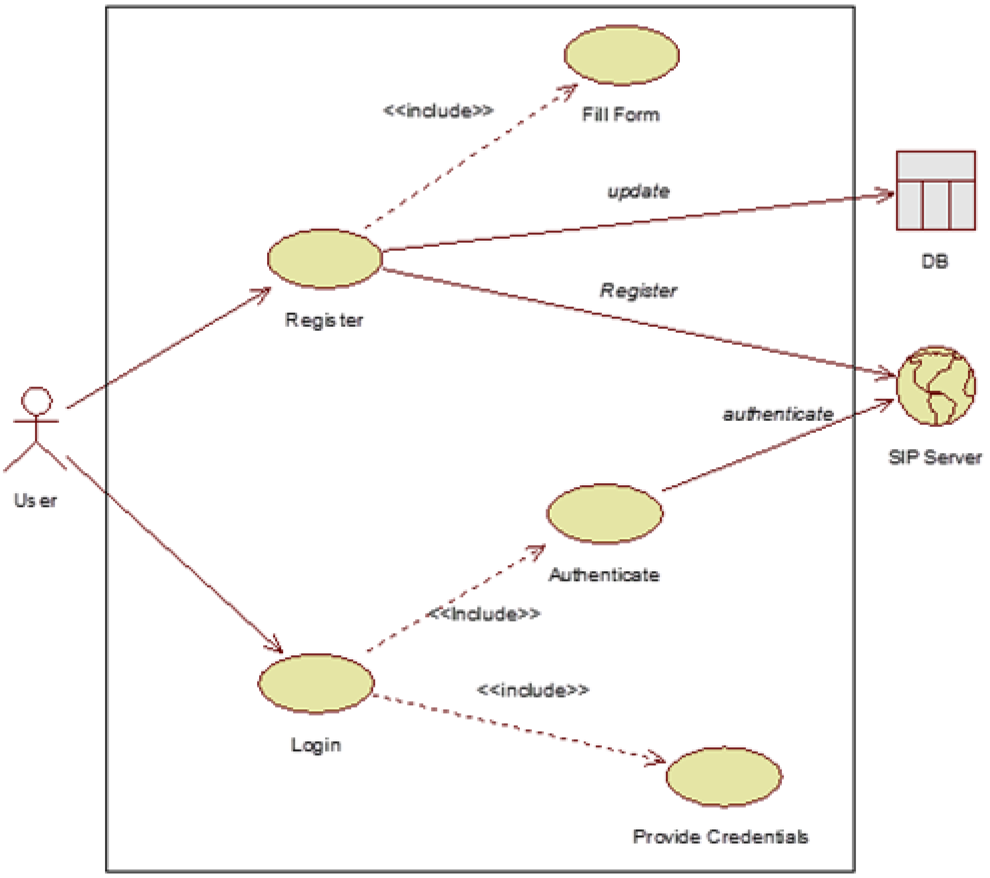


Figure 3.1 – System Use-Case



Functional Requirements

Figure 3.2 – Use-Case: Log-in

Use Case - 1: Log-in

Below is the use case description.

Brief Description: *This use case describes how the user logs-in to avail the functionalities provided by VOCAL.*

Actors

- End User.
- Database / Server.

Pre-Conditions

- The user has VOCAL installed on its Android mobile.
- The user has internet connectivity available over EDGE.

Basic Flow of Events

- User starts the VOCAL application.
- System / application prompts the user to:
  - ✓ Get registered,**or**
  - ✓ Enter credentials (if already registered).
- User provides his credentials (username and password) if it has already registered for the application and chooses to log-in.
- Credentials are sent to the DB for authentication.
- User gets logged-in.

Alternate Flow

Invalid Credentials: In step b of the Basic Flow, if validation of user fails, then

- ✓ System / app resumes at Step b of **Basic Flow**.

Unregistered User

- ✓ The user can register by filling up a sign up form.
- ✓ DB checks for the availability of the username chosen by the registering user.
- ✓ System verifies acceptability of password.

Forget Password: ….

- ✓ DB is Unresponsive.
- ✓ System / app resumes at step b of the **Basic Flow**.

Quit Application. If user selects to quit the app, then:

- ✓ Use case ends successfully.
- ✓ App is closed.
- ✓ User is logged off //state what b4 logging in.

Post-Conditions

Successful completion

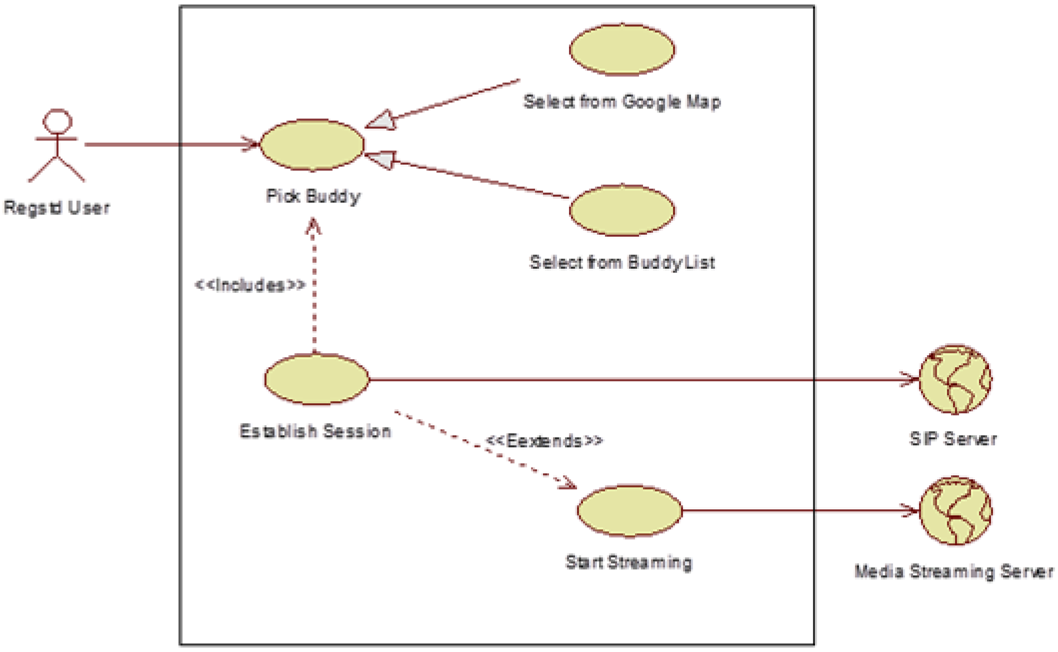
- ✓ User is logged in.
- ✓ User makes video calls, views location and manages its personal profile.

Unsuccessful completion

- ✓ User is not logged in.

Special Requirements

- // can log in from one place at a time.
- // log in attempts – 3 (max).



Use Case – 2: Video Call

Figure 3.3 – Use-Case: Video Call

Brief Description:

This use case describes how user can make a video call.

Actors

End User.

Streaming Server.

Pre-Condition

User is logged in.

Basic Flow of Events

Select a user from the displayed list.

Make call.

Successful completion of use case / call is successfully dialed.

Alternate Flow

Make call to another user

User is offline.

Call is ended.

Use case is resumed at step 1 of both I. and II.

Call is rejected (by callee)

Call is ended.

Use case is resumed at step 1 of both I. and II.

Disconnect call (by caller)

Call is ended.

Use case is resumed at step 1 of both I. and II.

Post-Conditions

Successful Call

Call is connected successfully.

Unsuccessful Call

Call is disconnected / ended / rejected.

Use Case – 3: Google Maps

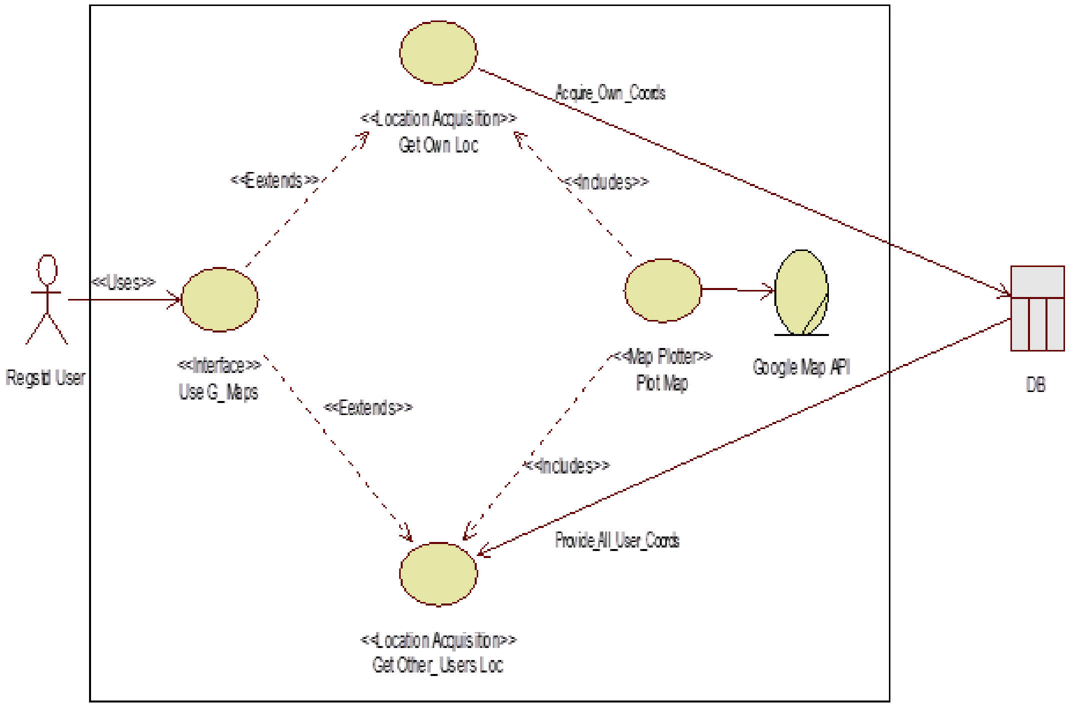


Figure 3.4 – Use-Case: Google Maps

Brief Description: *This use case describes how user can view location of other users using Google maps.*

Actors

End User.

GP.

Pre-Condition

User is logged in.

Basic Flow of Events

User views location of a friend.

User browses friend list.

User views map.

Location of selected user is displayed on the map.

User views location in his / her set radius.

User specifies the radius upto which he / shewants the map to be displayed.

User views map.

Locations of all the users lying within the specified radius are displayed.

Alternate Flow

Map / location are not displayed / GPS not responding / working.

Map is not displayed, may be because unresponsiveness of GPS or due to some other reason.

Use case is resumed at step 1 of both Basic Flows.

Post-Conditions

Map / Location are displayed

Successful completion of the use-case.

Map/ Location are not displayed

//hide location feature:

Use Case – 4: Profile Management

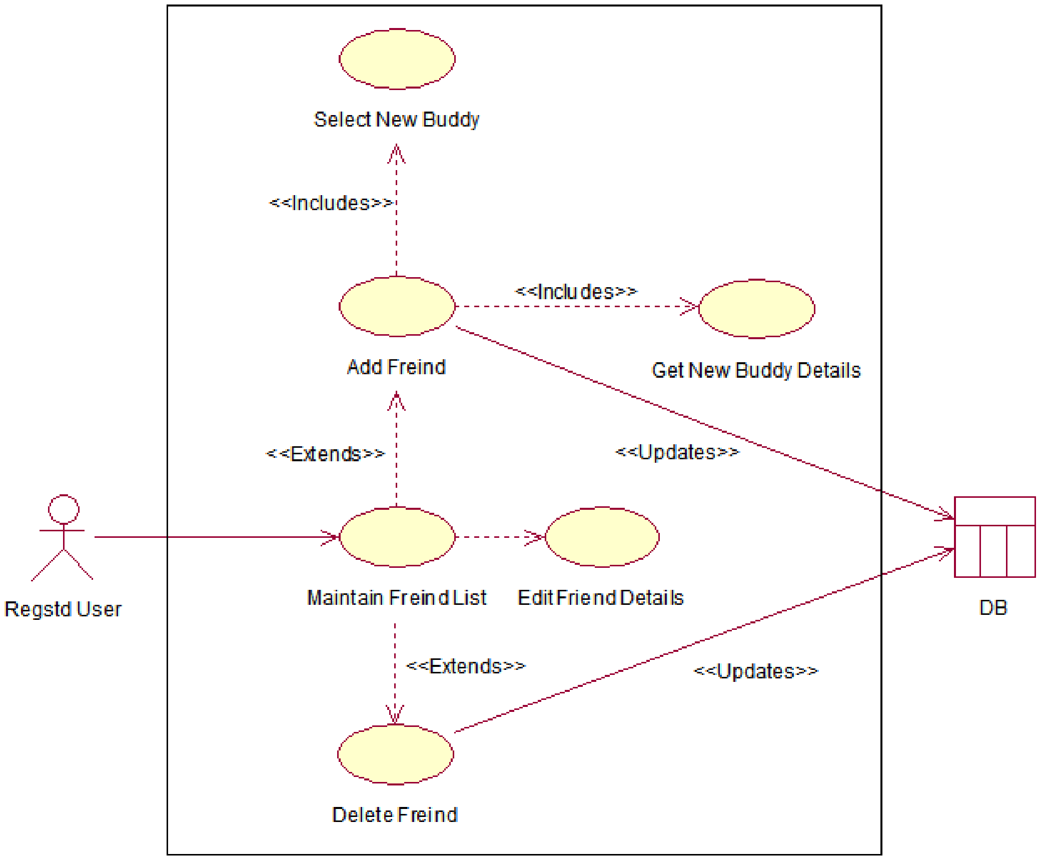


Figure 3.5 – Use-Case: Profile Management

Brief Description: *This use case describes how the user manages his / her profile.*

Actors

End user.

Database / Server.

Pre-Condition

User is logged in.

Basic Flow of Events

Fill basic information

Add a friend.

Delete friend.

Post-Condition

Changes made in profile are updated in the database.

Communication Features

Features	Priority
User can receive video call only when he / she is logged in.	High
User can make video call to one contact at a time.	High
User can quit the call anytime.	High
User can receive call from any VOCAL user.	High
User can send an instant message to one or multiple contacts.	Low(Extended goal)
User can receive instant message.	Low(Extended goal)

General Features

Features	Priority
Add a contact.	(Extended goal)
Delete an existing contact.	(Extended goal)
Block / unblock a contact.	(Extended goal)
Import contacts.	(Extended goal)

Settings Features

Features	Priority
Control audio volume.	High
Edit user profile.	Low (Extended goal)
Change video size / resolution / quality based both on user preferences and on bandwidth availability. It will be possible to change it during a video call.	Low(Extended goal)
Switch location privacy on / off - allows contacts to see the user location.	Low(Extended goal)

External Interface Requirements

User Interfaces

UIs will be same as already shown in use cases.

Hardware Interfaces

There is no hardware interface required for this product.

Software Interfaces

The software components which link VOCAL with the Android, include libraries for using Google Maps and video compression; API for interacting with the SIP Server. Mobile sets' own memory / database will be used for storing pictures and certain details of other users' profiles. Such details will include user name, last date / time of interaction.

The user will first provide its user name and password to gain access to the user interface for VOCAL. For authentication, the user name and password will be sent to the Windows Server 2003 through HTTP. After this, a message for successful / unsuccessful authentication will be sent from Windows Server to the user interface.

When a user logs in, he / she will be connected to SIP Server. Once connected to the SIP server, the user's status will be updated on Google Maps and other users will be able to see that user's location.

Location co-ordinates of the user will be acquired by using the mobile phone's built-in GPS, which in turn will update these coordinates on the server. Eventually the information will be updated on Google maps to be seen by other users.

A live video stream will be acquired for VOCAL, using mobile phone's built-in camera. Once the user is connected to the SIP Server, the user will send this live video stream through VOCAL over RTP. On start of video streaming, the other user will also start receiving the stream and a view will be available on interface of the other Google or Windows phone.

Communications Interfaces

Communication between Mobile Phone and Authentication Server

Windows Server will be used for storing and managing user names, passwords and location co-ordinates of all the users. Mobile phones will connect with Windows Server by HTTP for receiving and updating data.

Communication between Mobile Phone and SIP Server

In order to get live video stream from VOCAL, RTP/ RTSP has been used. We will initially establish an RTP connection with SIP Server and then start video streaming on the user interface. After establishment of connection, the Mobile phone will start video streaming to user interface ofthe other phone, which will continue to stream until either user selects to stop the video stream.

Other Nonfunctional Requirements

Performance Requirements

Latency: The latency should not exceed 150 ms for a video-call (one way).

Precision and Accuracy: The geo-positioning feature should be able to locate a VOCAL user with an accuracy of 15 meters, since it is a precision that most of the phones are able to achieve. Anyway a lower precision won't affect the overall functionality of the system.

Reliability and Availability: The maximum Mean-Time-To-Repair (MTTR) of the connection which is here equal to the Mean-Down-Time (MDT) should be less than 10 seconds so that if a communication fails, the user can retry or restore it and get the functionality up back within short time. Also, the Mean-Time-Between-Failures (MTBF) should absolutely not be less than 1 day. So with a MTBF of one day and a MTTR of 10 seconds, we get high availability. Maintenance should be performed during "quiet-hours" (usually at night), in order to avoid rush-hours and keep a high reliability.

Capacity Requirements: The first version of the system should support up to five pairs i.e. ten users in total, and up to 10 simultaneous users on low-definition video calling, or 4 with high-definition. The 10 user can be in action at the same time.

Scalability or Extensibility: The system will be highly scalable, i.e. as the number of clients connected and communications made will start to arise the hardware infrastructure will be expanded according to the needs.

Safety Requirements

The application should not be used for critical situations unless it has been officially certified.

Security Requirements

Access: The login connection will be implemented using the most secure Https protocol implementation available, because credentials should never be transmitted in plain-text.

Integrity: The integrity of the information stored in the system has to be preserved. Every unauthorized attempt of altering the data stored has to be blocked. This will include both the contact information and the log files. Weekly backups should be planned, in order to restore a system after eventually happened crash. The data that will be stored includes contact information and groups; they are pretty sensitive ones, but not enough to justify a frequency for backup higher than a weekly one.

Privacy:

Password won’t be stored as plain text, but as hash one. The user will have to choose a safe password of 8 characters that has to contain at least one number, one lower case character, one upper-case character and a special one.

Software Quality Attributes

Usage Ease: 90% of a test panel of non experienced users should be able to successfully achieve video-call within 5 minutes.

Ease of Remembering:95% of the test panel should be able to remember how to use all the functionalities he has experienced within two hours of use. With remembering it is meant that the user will be able to locate the functionality he wishes to use in 1 minute or less.After having used the application once, 95% of users are able to locate the experienced functionality within 1 minute.

Error Rates: After two week use, the user should achieve an error rate of less than 0.5%. Errors include: use the wrong functionality, call the wrong contact, login failure (wrong credentials), delete the wrong entry (or more than wanted); this measure can be achieved by online anonymous questionnaire.

Overall Satisfaction: After conducting a survey, 80% of the users should keep using the application after a two week exploitation period.

Trust: After having used all the features three times, 90% of the users should feel confident about the reliability, robustness and be convinced that the product does what it is expected to do.

Quick to Set-up: Within 5 minutes, the user should be able to set-up the application and create and account.

Compatibility: Both Windows and Android client applications must be able to communicate with the rest of the system, and to handle all of the functionalities (unless indicated in this document).

Learning: Any user without computer skills should be able to make a call, set up a conference, and use Push-to-Talk within the first 5 minutes of usage without referring to the user manual.

Understandability and Politeness: VOCAL application should use only symbols and words that are immediately understandable by users. All the technical details should be hidden from the user.

Maintenance: The first version of the system is open source and is its maintenance is not provided. Future commercial versions of the software may offer maintenance.

Supportability: The required support level needed by the system should be low, even lower than the one required by analogous product as Skype, up to the point it could be handled by the service provider’s helpdesk.

Adaptability Requirements: The client application should be portable on Android version higher than.

Issue Tracking: Tools should be used to bring up and track bugs/issues found by users. This is not intended to be a help desk, but just a bug reporting tool in order to achieve the best quality software achievable.

Other Requirements

Cultural and Legal Requirements: VOCAL is expected to be developed in accordance with the laws regulating communication, privacy and confidentiality. The license that has been chosen for this product is LGPL. Hence, no libraries licensed under GPL can be used for our product.

Design and Implementation Constraints

Dimension	Constraint	Factor / Reason for Constraint	Degree of Freedom
Features	Video call will not be possible under all circumstances if signal strength is poor or bandwidth is choked.	Shared bandwidth offered by cellular phone companies and distances from BTS limit the actual bandwidth to be utilized for video streaming.	30%
	The application will not be usable on Apple and other mobile phone sets which do not work on Android.	The application is being developed only for Android based mobile phone sets.	70%. Since Android phones are already much popular in public.
Quality	Video may have jitters and low quality.	Bandwidth offered by cellular companies is less and video streaming entails greater compression.	50 - 55%
Service	Application will be usable subject to availability of adequate bandwidth.	EDGE coverage is not nationwide. Shared bandwidth is being offered presently and it limits a single user in crowd of other users.	Indigenous compression will solve the issue of required bandwidth to some extent.
Hardware	Comprehensive testing on all types of Windows / Android Phone sets may not be feasible due to a vast variety of models available in the market.	Different models have different design features e.g. screen size / resolution; absence of a secondary camera, which may limit a user from making a video call while watching the screen.	Testing will have to be done on more popular mobile phone sets / other phone sets with such specifications.
Cost	Android mobile phones are generally expensive. At least three sets of different models / specifications (camera resolution; EDGE / 3G support; processing power) will have to be procured.	For a large user population, mobile set prices need to lie within affordable range.	For the moment, project may be started with only two Windows and two Android phone sets.

CHAPTER 4

SYSTEM DESIGN SPECIFICATIONS

Introduction

This is the System Design Document for the Location Based Video Calling Application for Smart Phones (Android and IPhone).

Purpose

The Software Design Document captures the design constraints and assumptions as well as the detailed design of the subsystems and components of the application.

Scope

This Design Document pertains to the Video Calling Application of this project along with location acquisition through Google Maps, with a specific focus on the features and capabilities being implemented.

Overview

This Design Document gives an overview of the application and its primary functionality, identifies the assumptions and constraints followed during the design of the software, documents the overall system architecture and provides the detailed design information for each subsystem and component in the current delivery.

System Overview

The aim of our project is to develop a mobile application on the Android which can be used to make Video Calls as well as to locate other users through Google Maps. The application allows the user to stay in touch with other users of this application by making a Video Call.Users can also add each other in their buddy list so that they can connect with each other at any later time. Along with Video Calling Users can also locate each other through Google Maps.

This project focuses on developing a Java application which is to be installed on the Android phones. Once the user starts the application he has an option for log-in or register and once the user is logged-in he can view users in his area whose locations are retrieved from a central database and are then displayed as blips on Google Maps, then the user can also pick anyone from map or any buddy from buddy list to initiate a video call.

Design Considerations

This section contains all of the assumptions, dependencies, and constraints that were considered during the design of each subsystem and component.

Assumptions and Dependencies

The default settings on the mobile application will reflect on the Google maps.

Constraints

- **User of the mobile should have the internet access.**
- **Huge data cannot be saved on the mobile.**

System Architecture

This section provides a detailed and comprehensive architectural overview of the system.MVC Architectural Pattern has been used for this projectbecause the project nature requires a view i.e. the application interface, needs to be separated from the back-end application logic so that the back-end complex logic is transparent to the user and he finds the application easy to use.

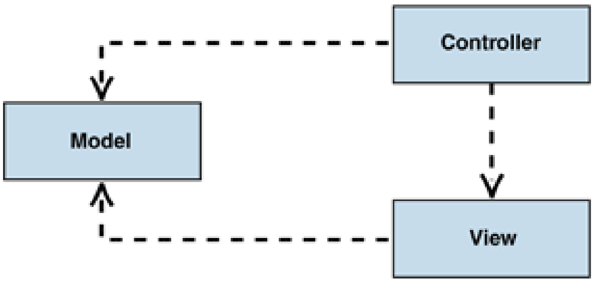


Figure 4.1 - MVC Architectural Pattern

Model

The model manages the behavior and data of the application domain, responds to requests for information about its state (usually from the view), and responds to instructions to change state (usually from the controller) i.e. **The Data Sources** and their behavior.

View

The view manages the display of information i.e. **The App Interface**.

Controller

The controller interprets the mouse and keyboard inputs from the user, informing the model and/or the view to change as appropriate i.e. The App Logic.

Design Details

Below is the Architectural Design.

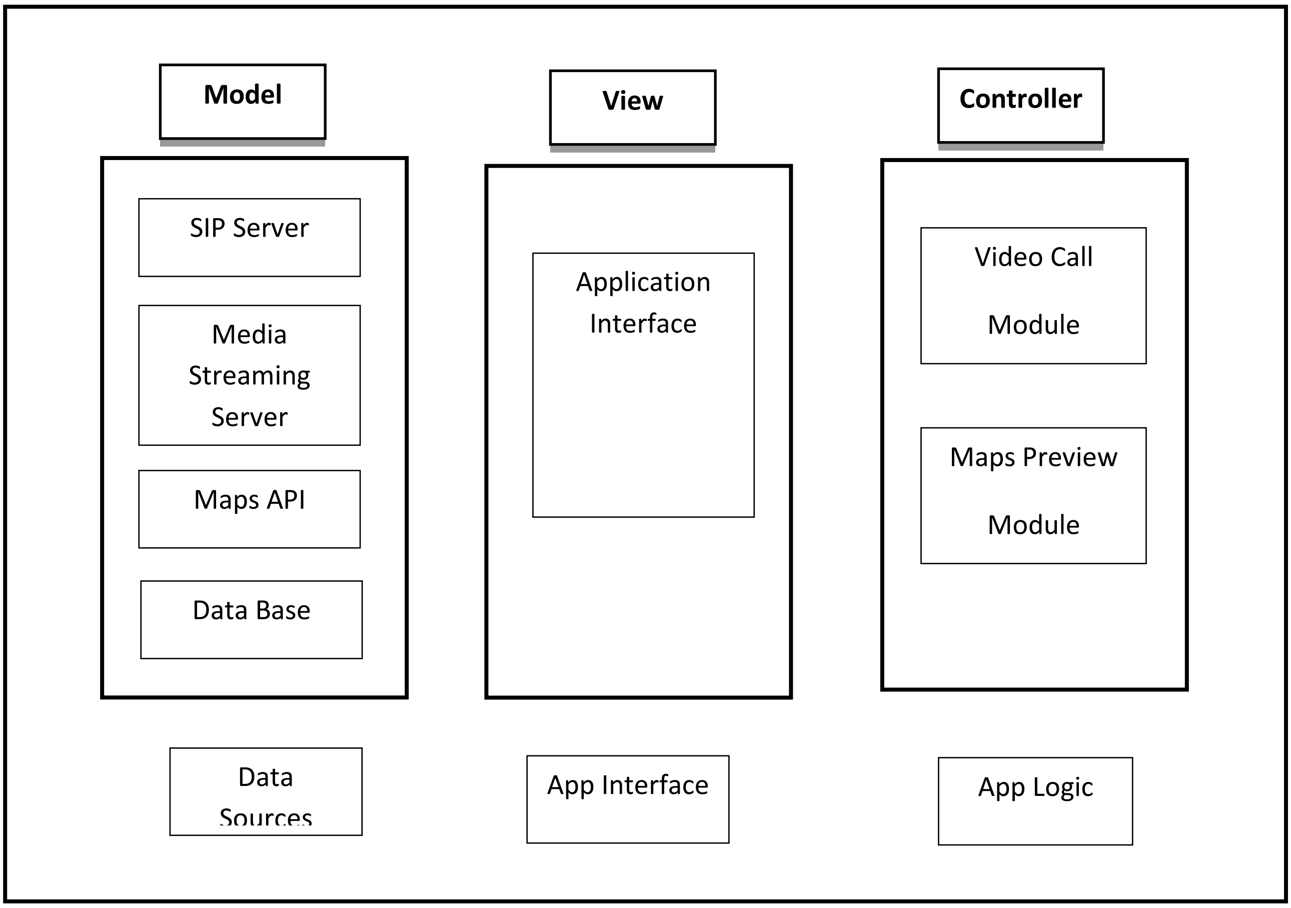


Figure 4.2 – System Architecture

Design Pattern:

"The design pattern used isBridge Pattern."

Bridge Pattern is used because our project needs to decouple an abstraction from its implementation so that the two can vary independently. The front-end needs to be transparent from the back-end complexities, so that the user can use VOCAL without having knowledge of the interactions between modules. So interface is published in an inheritance hierarchy, and implementation is buried in its own inheritance hierarchy.

Static View

System Use-Case Diagram

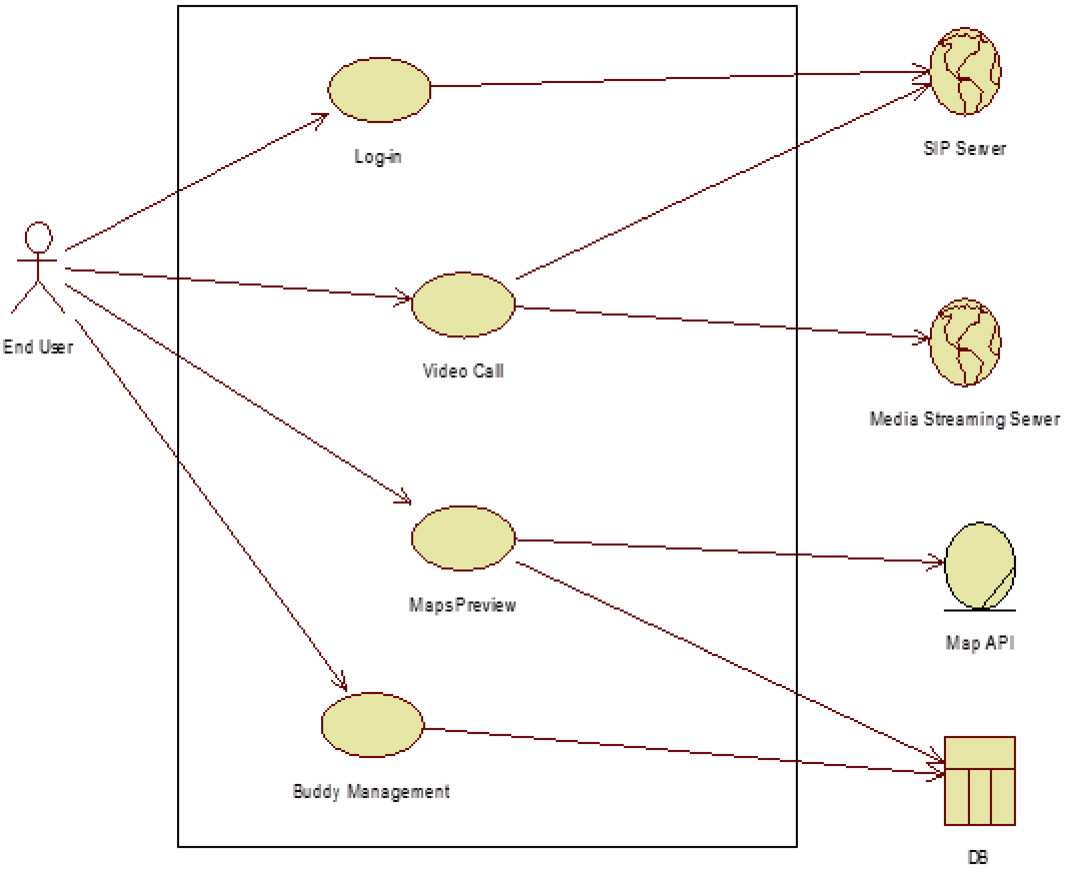


Figure 4.3– System Use-Case Diagram

Log-in UC Diagram

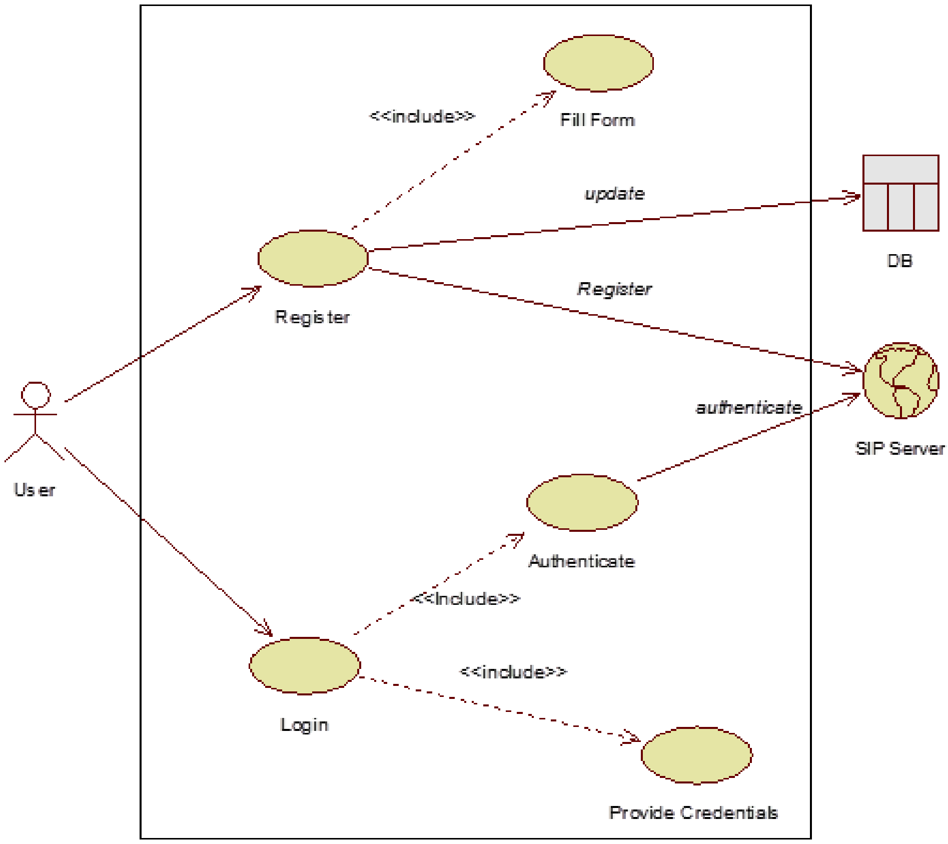


Figure 1.4 - Log inUse-Case Diagram

Brief Description: *This use case describes how the user logs-in to avail the functionalitiesprovided by VOCAL.*

Actors

End user.

SIP Server.

DB.

Pre-Conditions

User must have VOCAL application installed in his/her phone.

User must have internet connectivity available.

Basic Flow of Events

User starts the VOCAL application.

System / application prompts the user to get registered or to enter credentials (if already registered).

User provides his credentials (username and password) if he/she has already registered for the application and chooses to log-in or if the user is not registered then he/she can register by filling in the registration form.

Credentials are sent to the SIP for authentication or registration (in case if user chooses to register). DB is also updated when a new user get registered.

User gets logged-in.

Alternate Flow

Invalid Credentials: use case diagram gets resumed at step 2 if the user authentication gets failed due to invalid credentials.

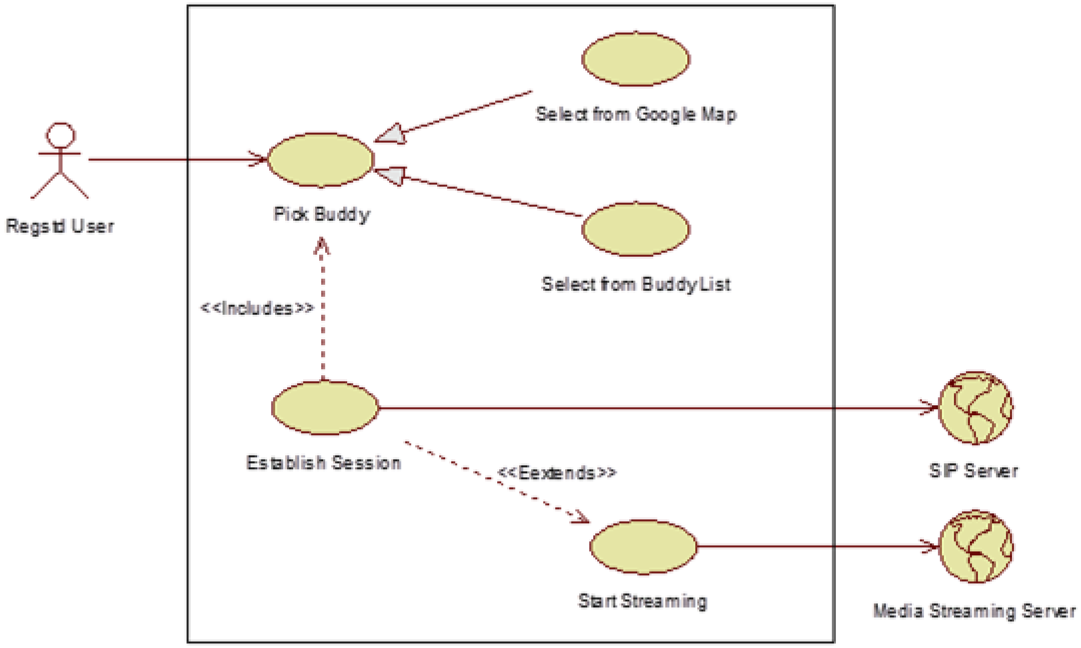
Post-Conditions

Successful Completion

User is logged-in. Now user can use all features provided by VOCAL.

Unsuccessful Completion

User is not logged-in.



Video Call Use-Case Diagram

Figure 4.5 - Video CallUse-Case Diagram

Brief Description: *This use case describes how user can make a video call.*

Actors

End User

Sip Server

Media Streaming Server

Pre-Condition

User is logged in.

Basic Flow of Events

Select a user from either the Buddy List or from the Map.

Establish session between the Caller and the Callee through SIP Server.

After successfully establishing a session, start video and audio streaming.

Successful completion of use case / call is successfully dialed.

Alternate Flow

Session is not established, use case diagram terminates at step 2.

Post-Conditions

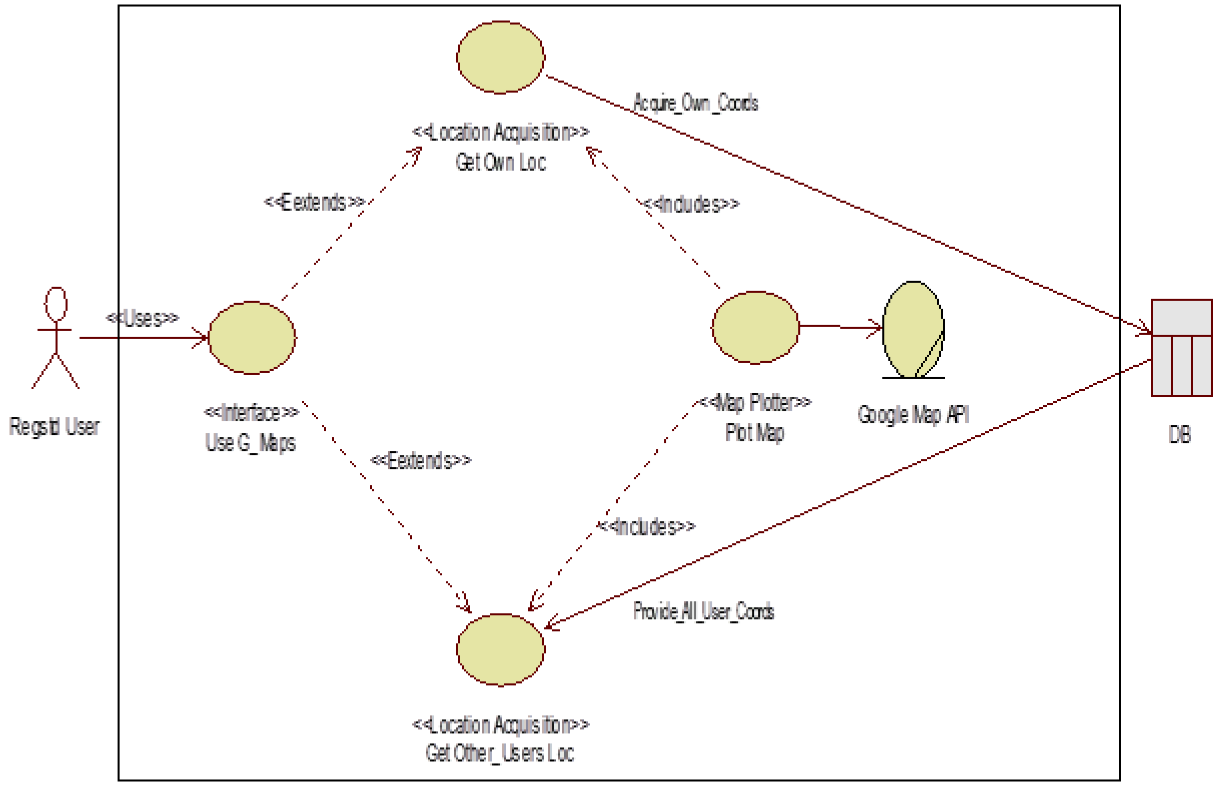
Successful Completion

Call is connected successfully.

UnsuccessfulCompletion

Session is not established.

Call log will be updated.



Maps Preview Use-Case Diagram

Figure 4.6 - Maps PreviewUse-Case Diagram

Brief Description: *This use case describes how user can view location of other users using maps API.*

Actors

End User.

Database.

Maps API.

Pre-Condition

User is logged in.

Basic Flow of Events

Retrieve own and other users location from DB.

Send locations and radius to maps API.

Map is plotted by the API and displayed to the user.

Post-Condition

Map is displayed to the user.

Class Diagram

Below is class diagram.

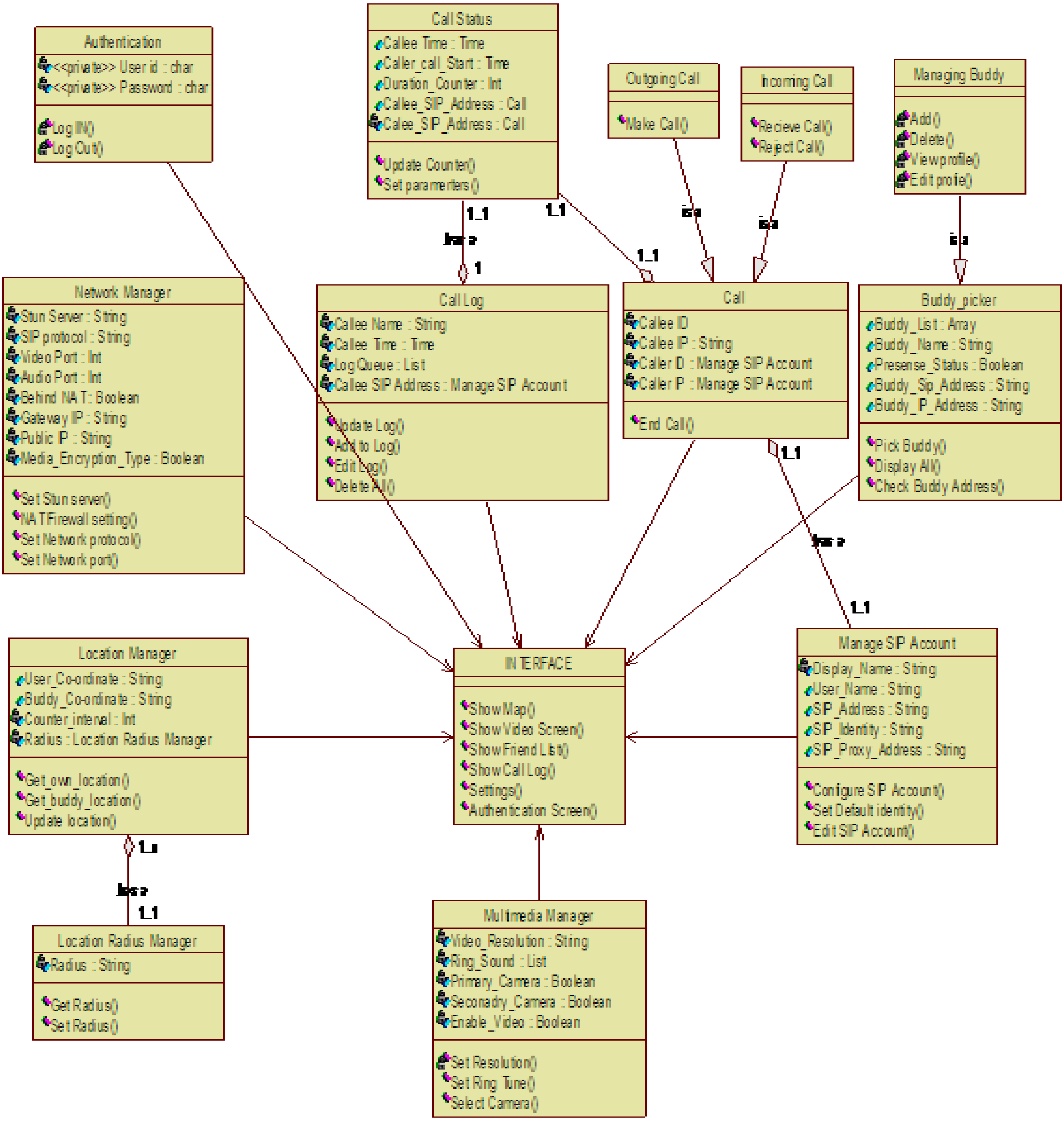
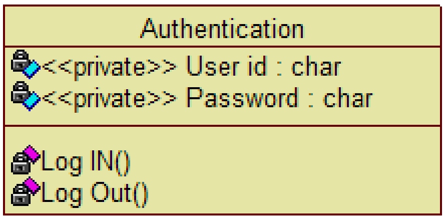


Figure 4.7 - Class Diagram

Class Descriptions

Below is class description.



Authentication Class: This class is used to authenticate user from sip server. User provides his / her credentials which are sent to the SIP Server.

Figure 4.8- Authentication Class

Log_In() Takes two arguments**:** SIP Address and Password and sends them to SIP Server.

Log_Out() Terminate**s** the session.

Call Class: This is the main class that provides the Video Call functionality. It has two child classes Incoming and Outgoing call. It has aggregation with SIP manager class when a call comes or going outside SIP address is detected by SIP manager class.

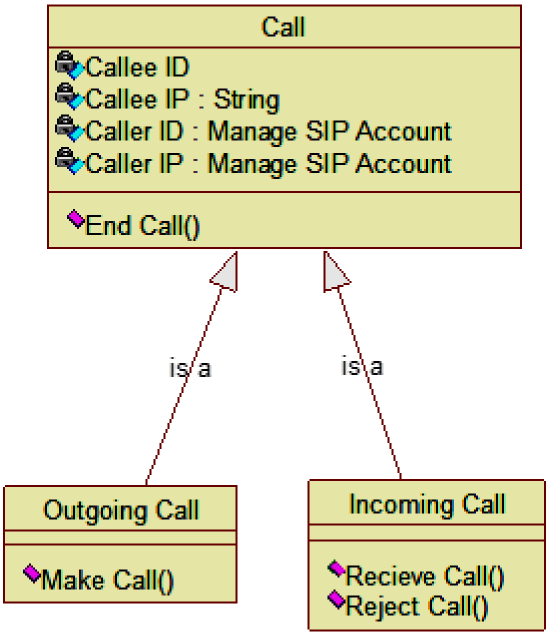


Figure 4.9- Call Class

Call Status Class: During incoming and outgoing calls, this class records time and SIP address of the callee.

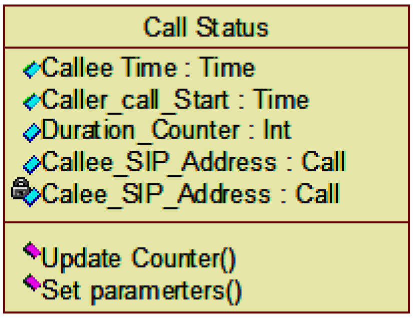


Figure 4.10 - Call Status Class

Update Counter() Continuously updates itself because to measure the call time with a string return type.

Call Log Class: Allthe call history is maintained here. Logging of outgoing as well as incoming calls, with their callee and caller addresses, and call durations is managed by this class. It has following functions. It has a queue when a call status comes it will entered in that queue.

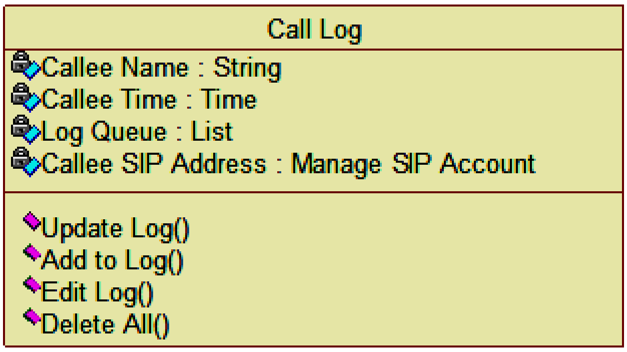


Figure 4.11 - Call Log Class

Edit()

ADD()

Delete ALL() Clears all the list.

Network Manager Class: This is an important class which contains all the information regarding network. It contains protocol, ports, STUN Server and encryption types. This enables users to make a video class if they are behind NAT Routers and proxies. STUN Server is used for resolving IP addresses of users who are behind NAT.

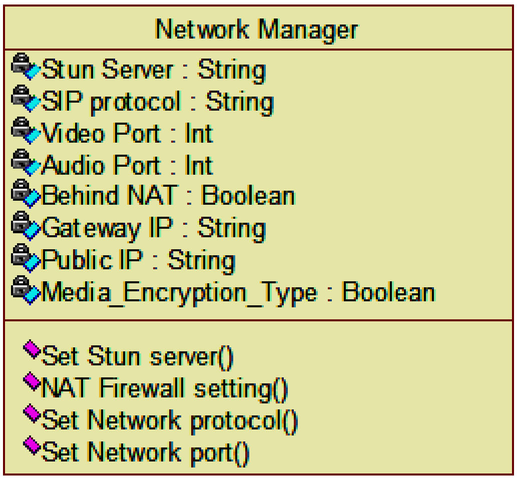


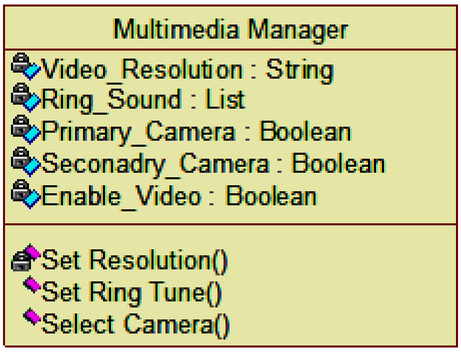
Figure 4.12 - Network Manager Class

Set StunServer() Sets up the server for resolution of IP.

Set Network Protocol() Sets the protocol for communicatione.g. TCP,UDP.

Set Firewall() Tells whether the firewall is enabled or not.It retuns aBoolean.

Multimedia Manager Class: All the settings of multimedia are controlled by this class e.g. video resolution, ring tone used, Camera etc. Actually it deals with the call quality. Video resolution controls the number of frames per second.



Multimedia Manager

Video_Resolution : String
Ring_Sound : List
Primary_Camera : Boolean
Seconadry_Camera : Boolean
Enable_Video : Boolean

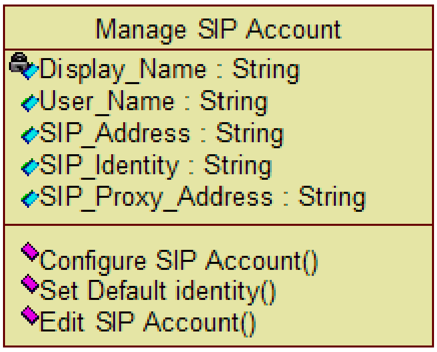
Set Resolution()
Set Ring Tune()
Select Camera()

Figure 4.13 – Multimedia Manager Class

It has following functions:

Set resolution()

Select Camera() Whether a primary or secondary camera is used for call.



Manager SIP Account Class: User display nameand the information related to SIP is entered here.SIP user ID and SIP Server Name are entered here.SIP proxy is optional.Configure SIP account () is used to Manage all information about sip account. When a user wants to make a call all the information needed it takes it from Sip manager.Edit SIP() simply edit all the information of account.

Figure 2.14 - Manager SIP Account Class

Buddy Picker Class: All the friend information is containedhereand it is connected to a SQL Server.When a user logs-in its friend list, it fetched by his SIP address from database with friend names and their SIP IDs.

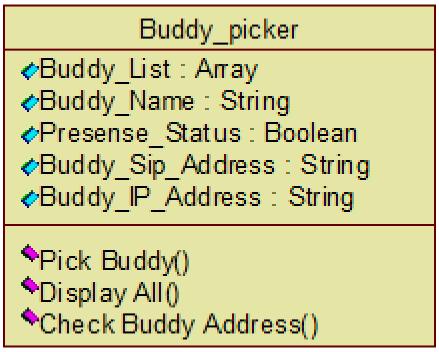


Figure 4.15 - Buddy Picker Class

Display all() Shows all the friends of that user.

Managing Buddy Class: Simply manages buddy information and provides Add, Delete and Edit functions.

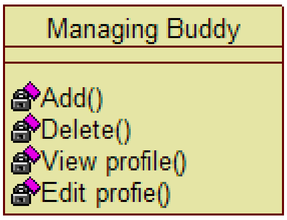


Figure 4.16 - Managing Buddy Class

Maps Manager Class: Locations of user anduser'sfriends are managed by this class. This class has aggregatesthe Radius Class. Radius is define by the Location Radius Manager Class which has get**radius_method()** with return type string. This function specifies a radius in which other users should be displayed on Google map.

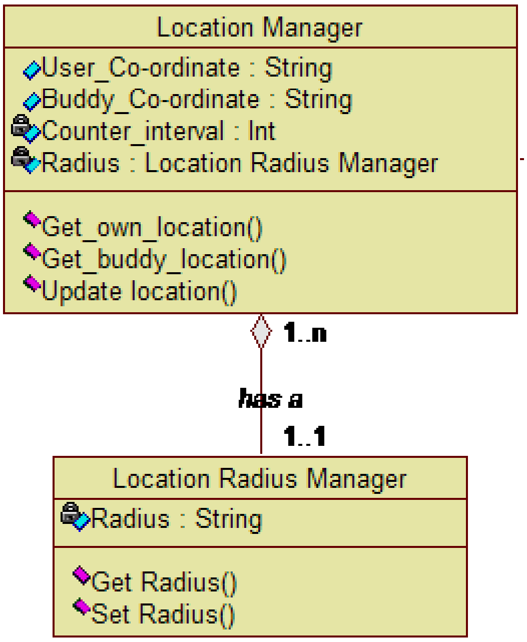


Figure 4.17 - Maps Manager Class

Get Location() Gets user's /friend's location and displays it on map.

Update Location() Updates the user location after a counter to database so that location is continuously updated.

Interface Class: This class relates all the classes with each other. The VOCAL design is separate from logic.So, all the classes have their interfaces which are connected with each otherby this class.

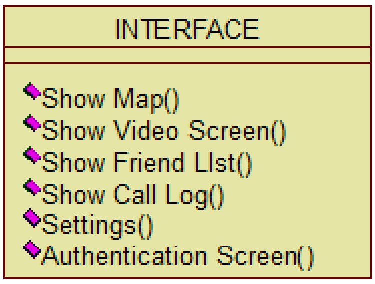


Figure 4.18 - Interface Class

Dynamic View

Dynamic view has sequence, activity and state machine diagrams.

Sequence Diagrams

Below are sequence diagrams of the system.

Log-in Sequence Diagram

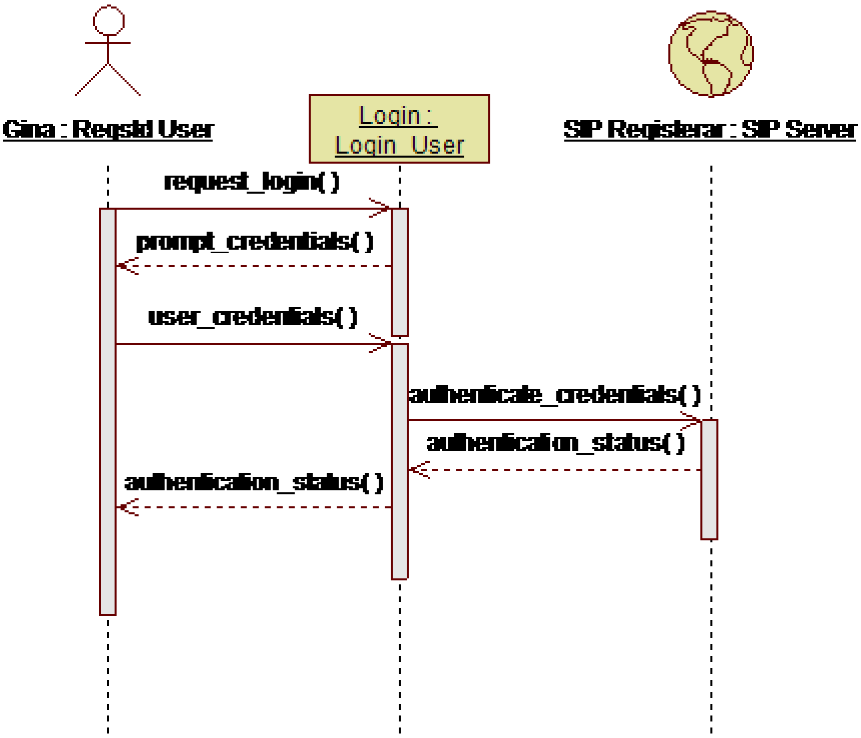


Figure 4.19 -Sequence Diagram: Log-in

Video Call Sequence Diagram

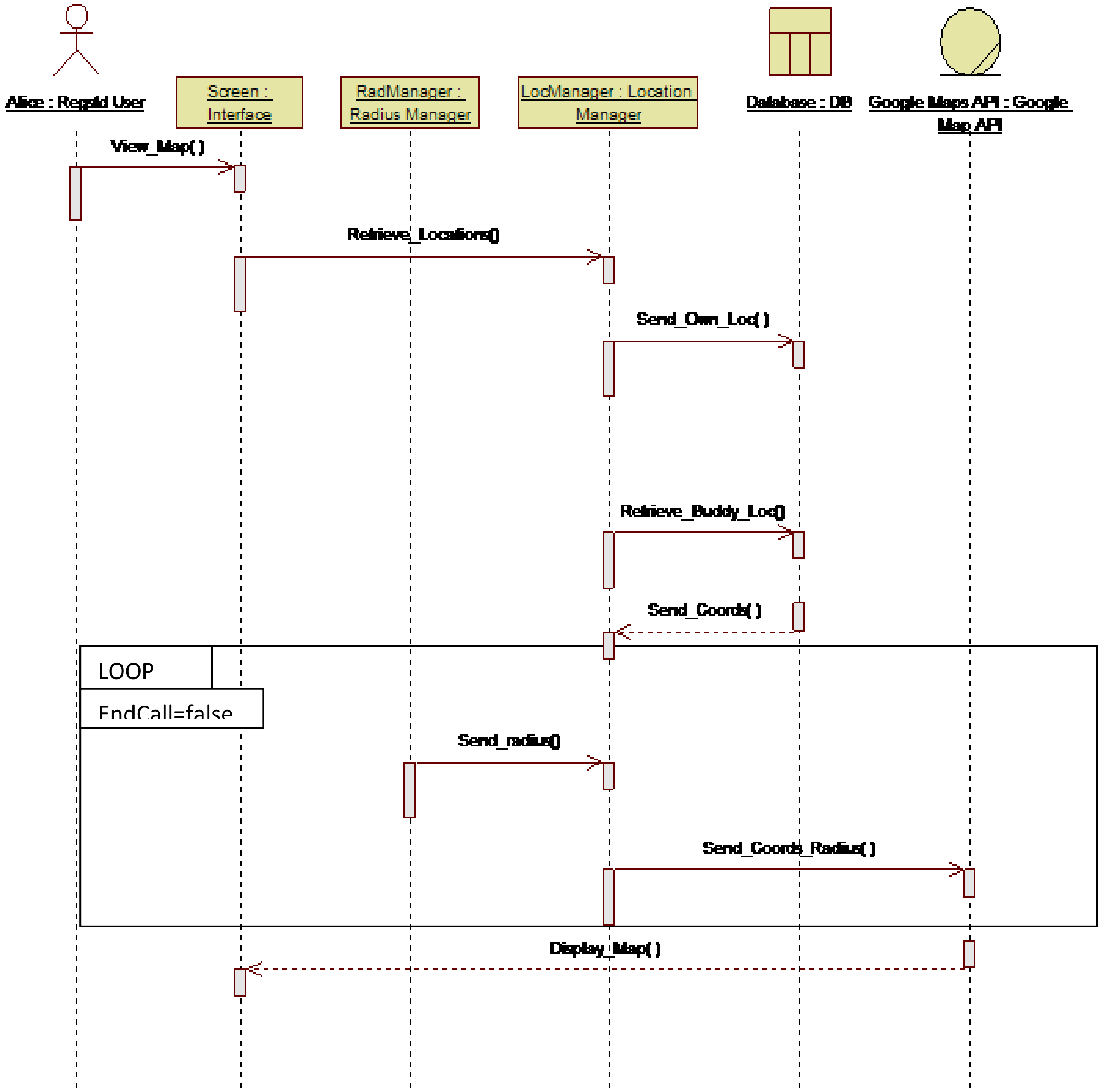


Figure 4.20 -Sequence Diagram: Video Call

Maps Preview Sequence Diagram

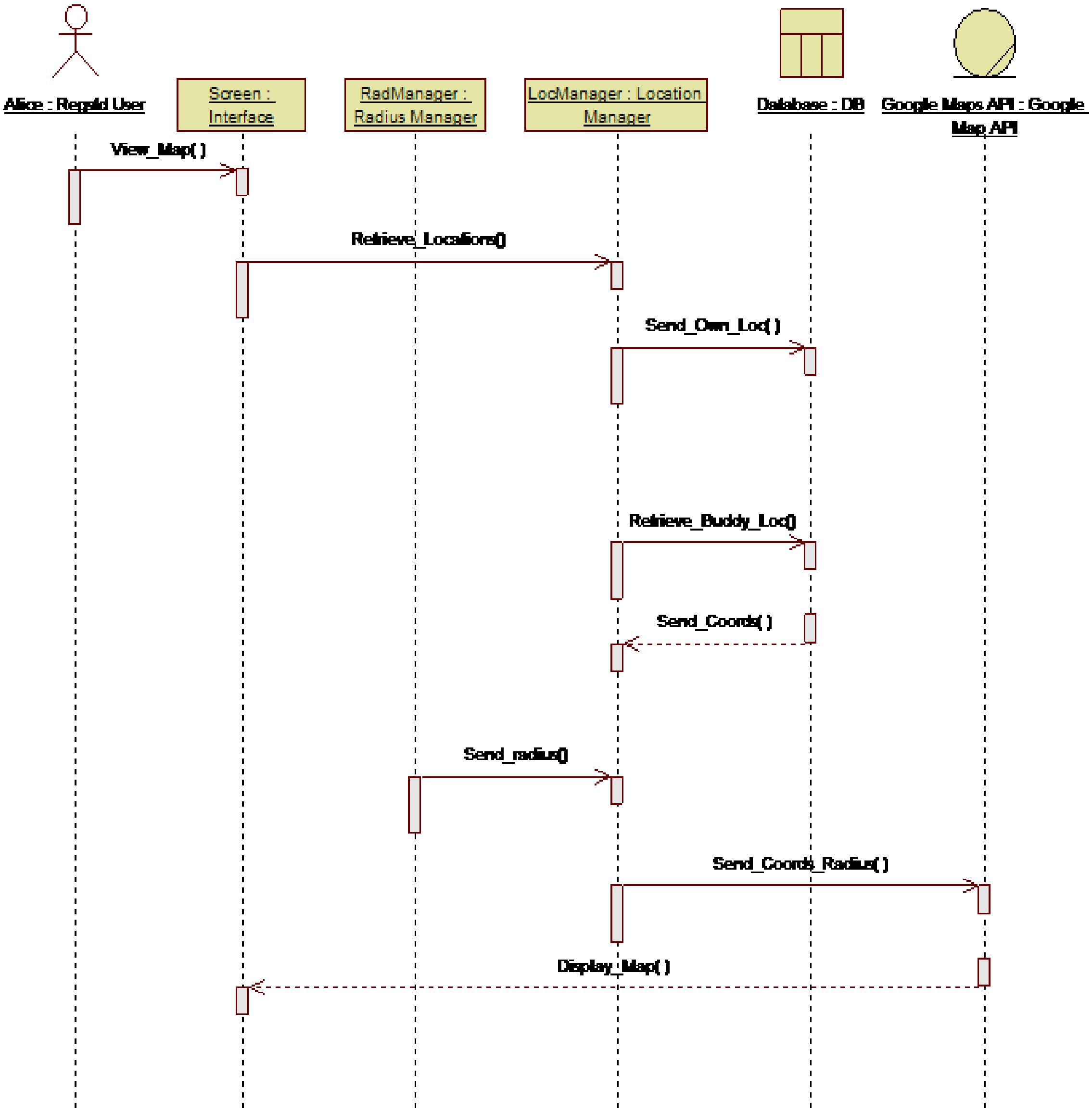


Figure 4.21 -Sequence Diagram:Maps Preview

Manage Buddies Sequence Diagram

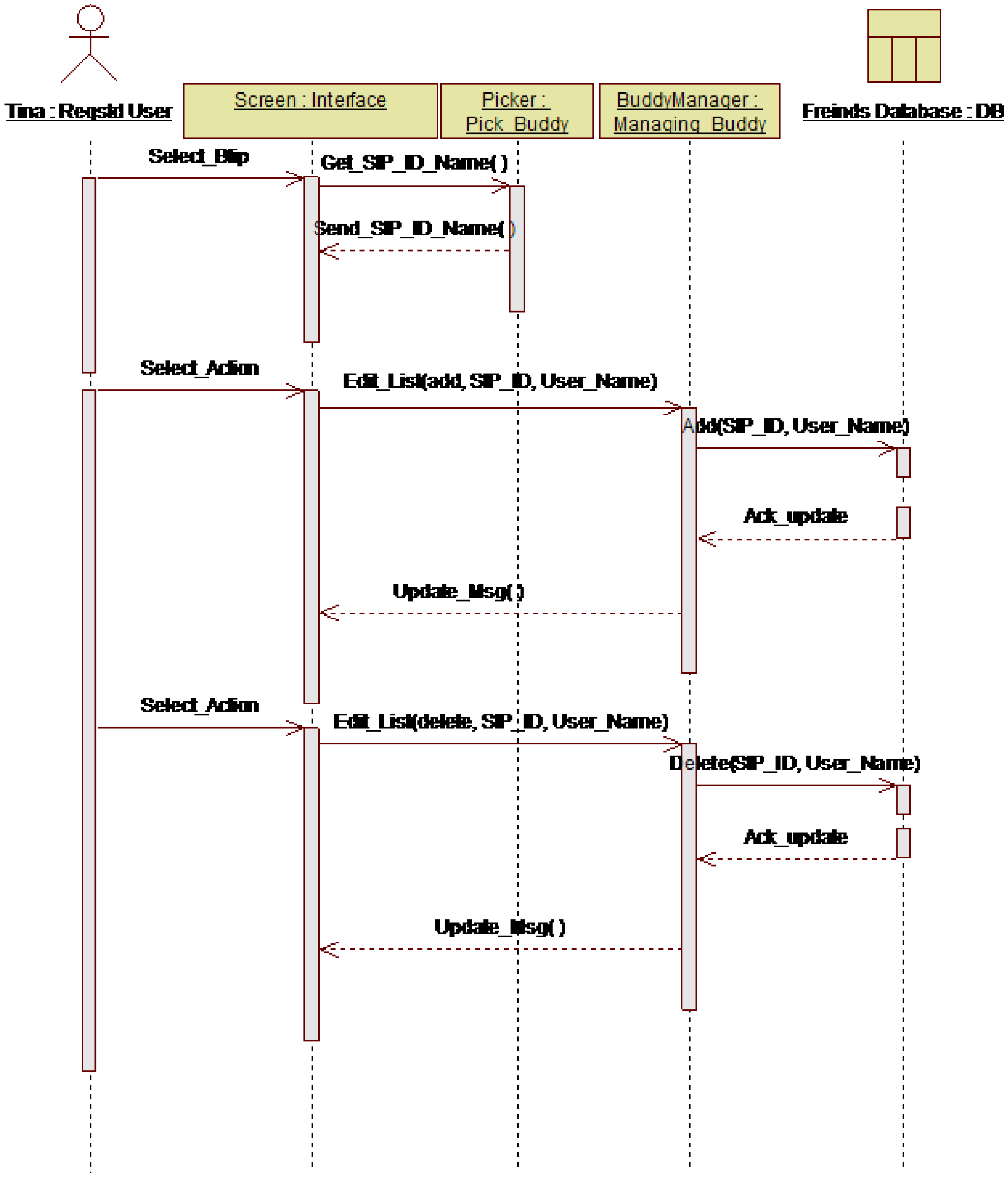
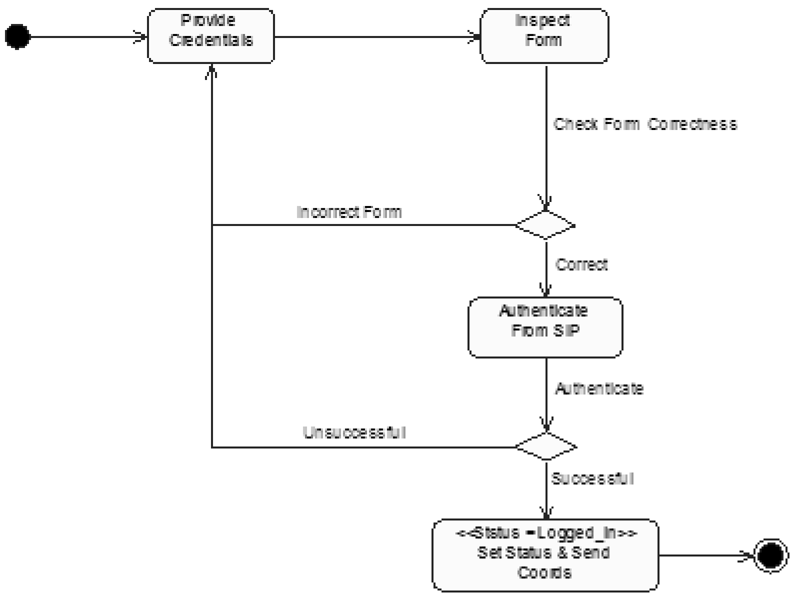


Figure 4.22 -Sequence Diagram: Manage Buddies



Activity Diagrams

Figure 4.23 - Activity Diagram: Log-in

Log-in Activity Diagram

Video Call Activity Diagram

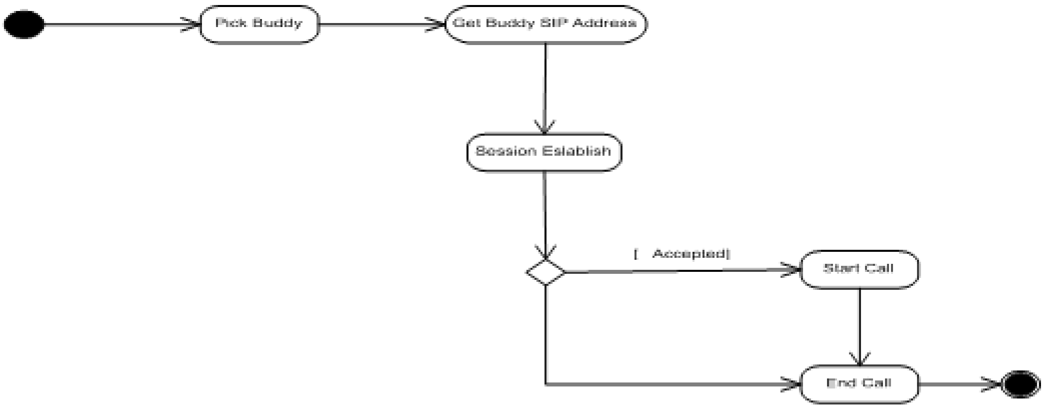


Figure 4.24 - Activity Diagram: Video Call

Maps Preview Activity Diagram

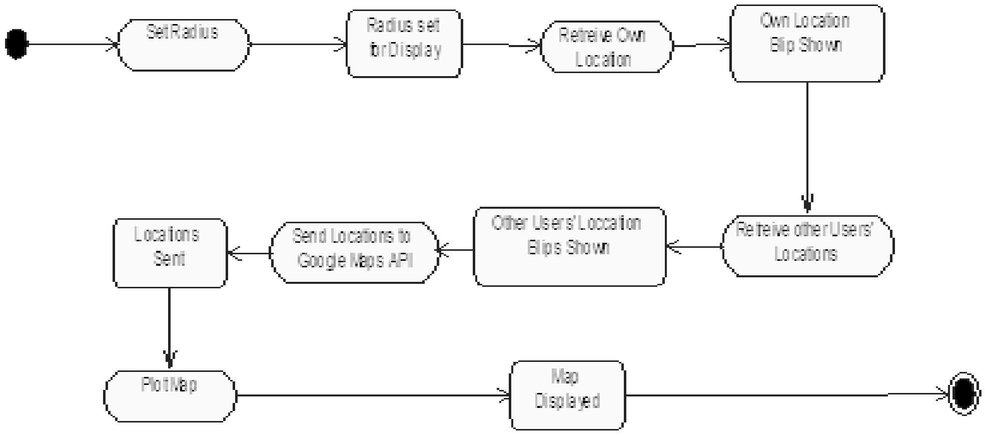
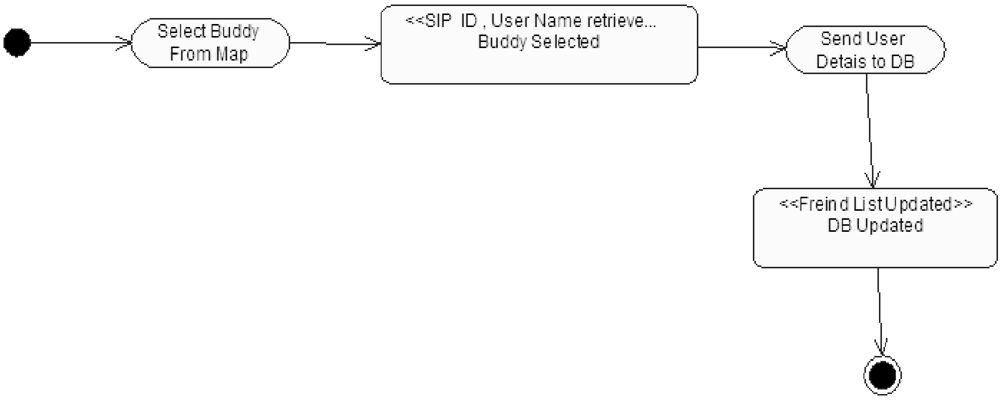


Figure 4.25 - Activity Diagram: Maps Preview



Add Buddy Activity Diagram

Figure 4.26- Activity Diagram: Add Buddy

Delete Buddy Activity Diagram

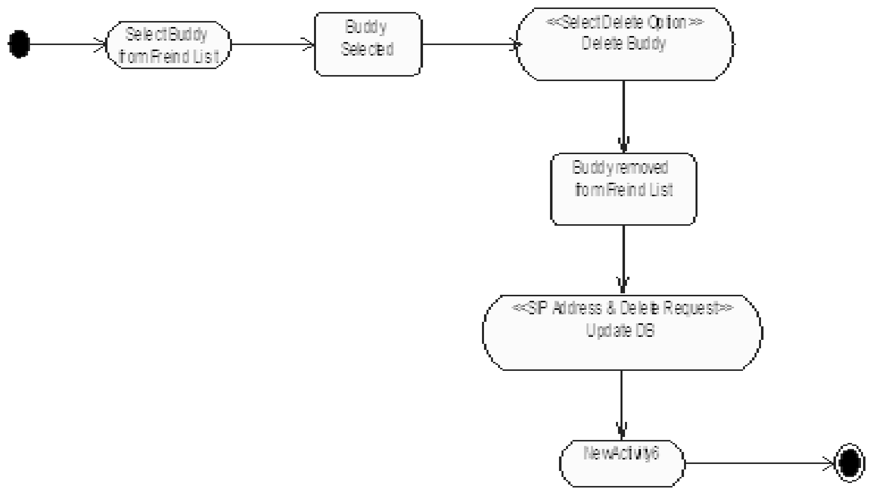

Select Buddy from Freind List
Buddy Selected
<<Select Delete Option>> Delete Buddy
Buddy removed from Freind List
<<SIP Address & Delete Request>> Update DB
NewActivity6

Figure 4.27 - Activity Diagram: Delete Buddy

State Machine Diagrams

Log-in

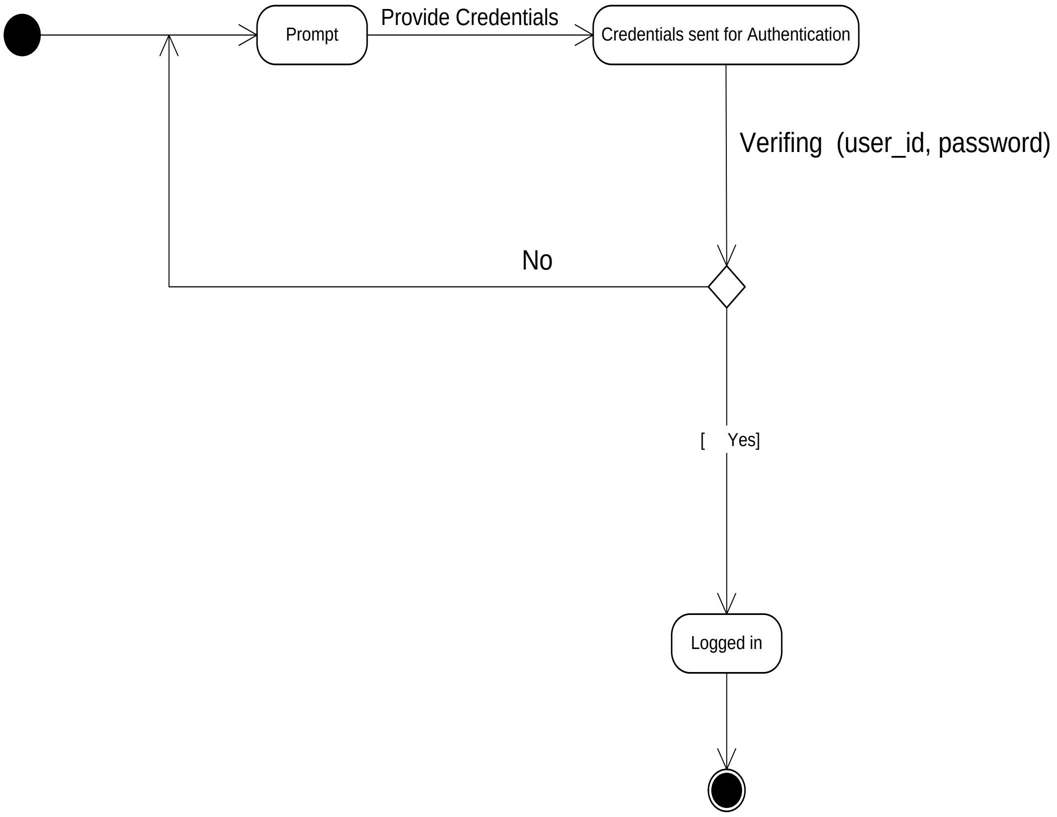

Prompt
Provide Credentials
Credentials sent for Authentication
Verifing (user_id, password)
No
[Yes]
Logged in

Figure 4.28 - State Machine Diagram: Log-in

Video Call

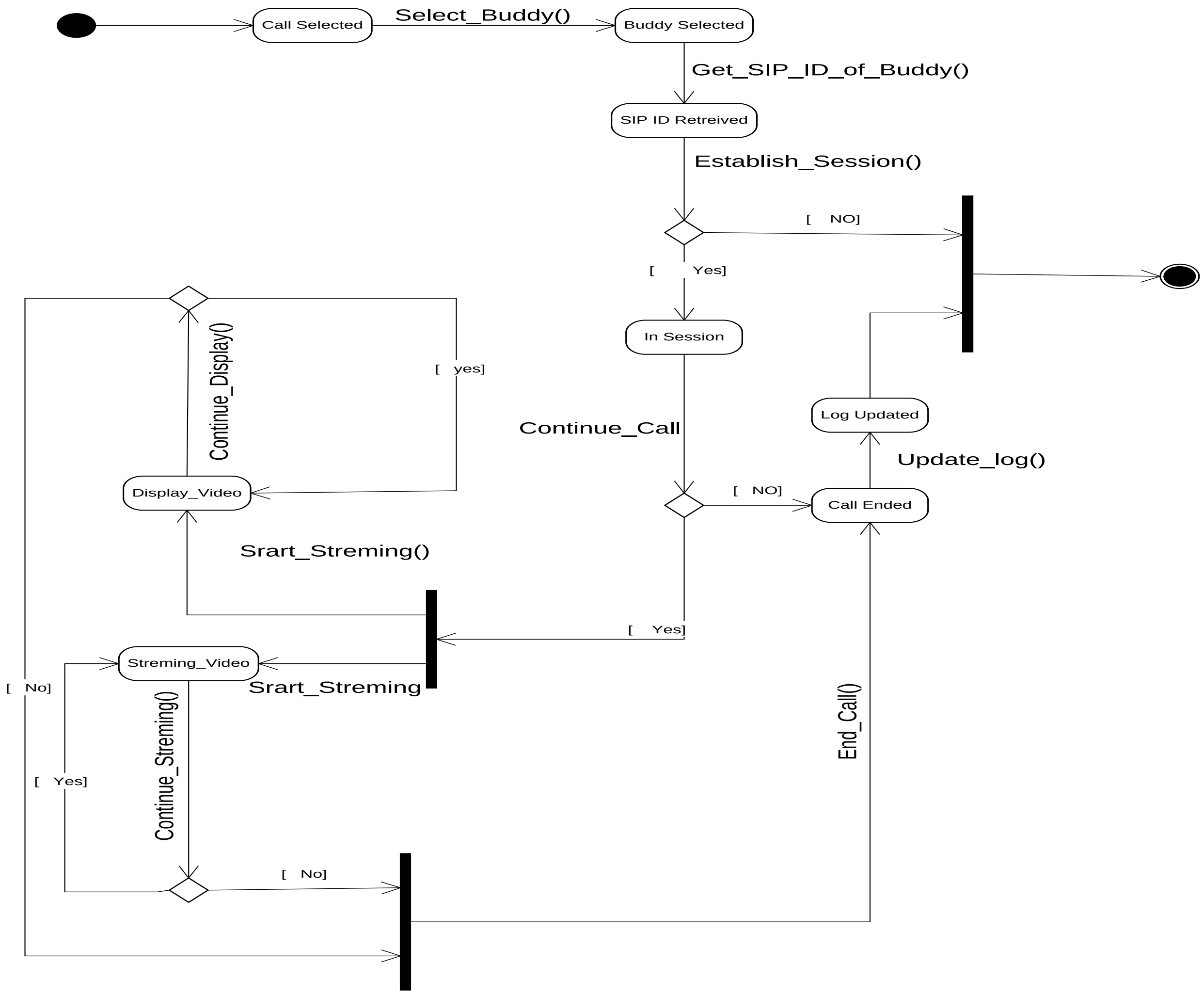


Figure 4.29- State Machine Diagram: Video Call

Data Design - ERD

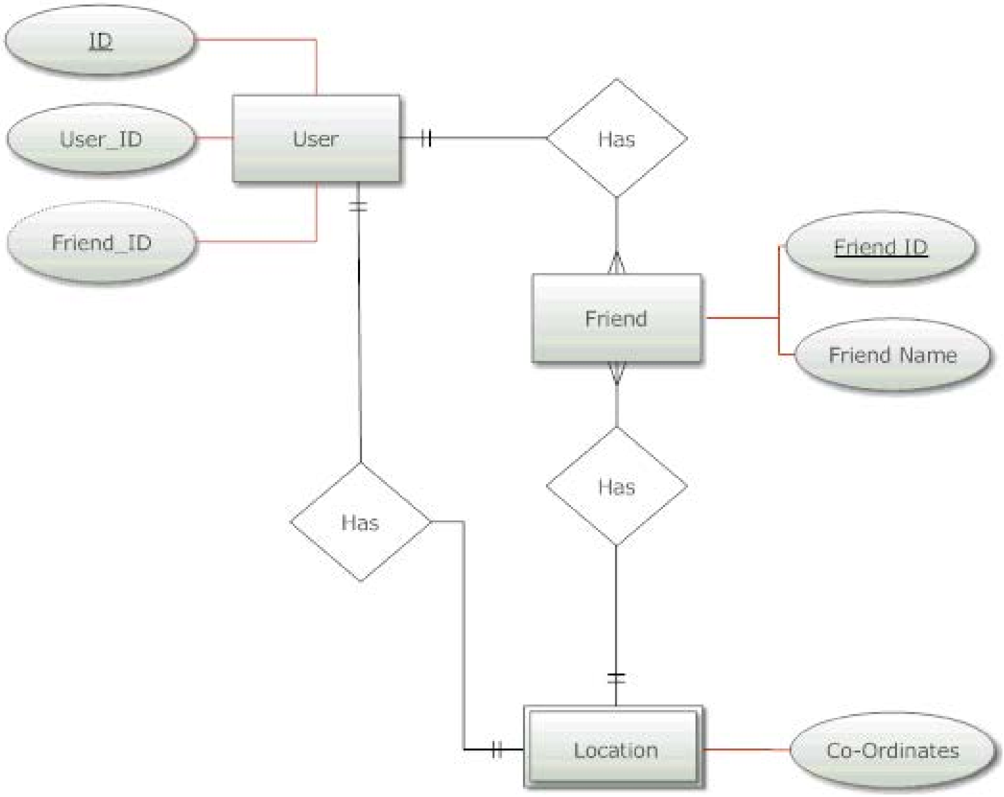
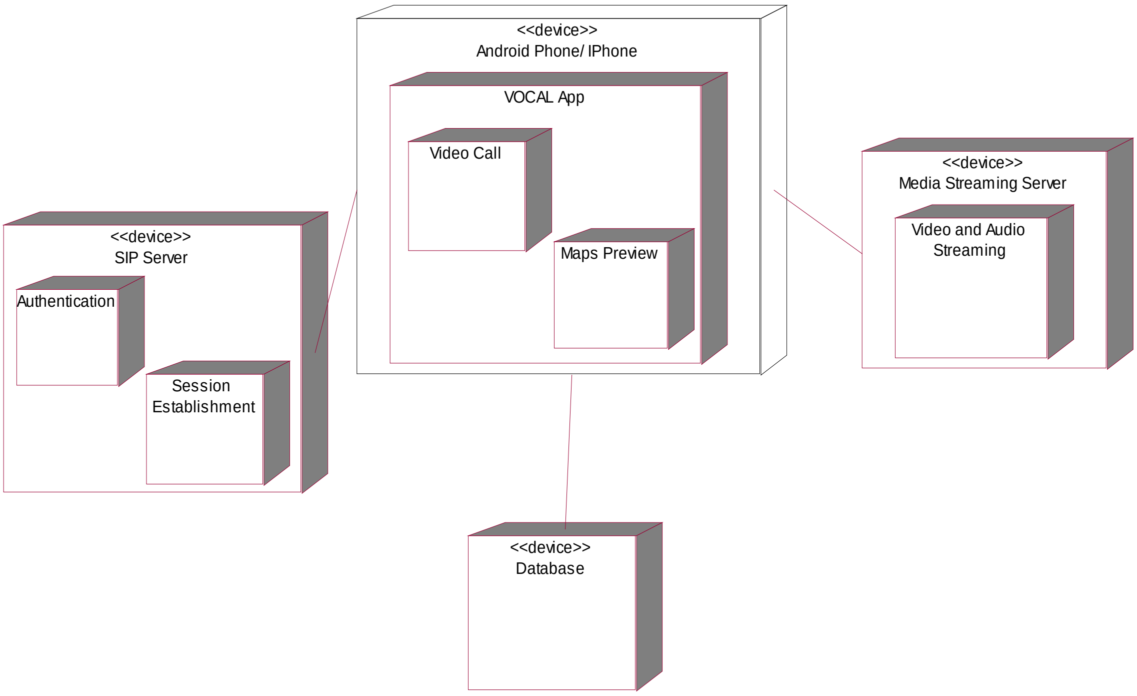


Figure 4.30 - Entity Relationship Diagram



Implementation View - Deployment Diagram

Figure 4.31 - Deployment Diagram

User Interfaces

Main Screen

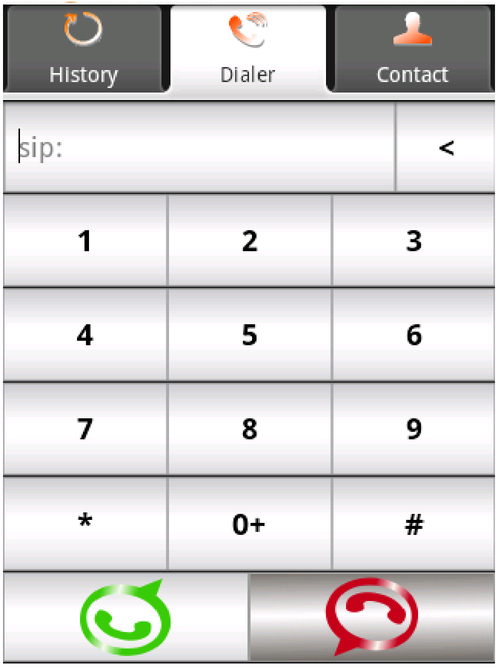


Figure 4.32 - Main Screen

Log-in Screen

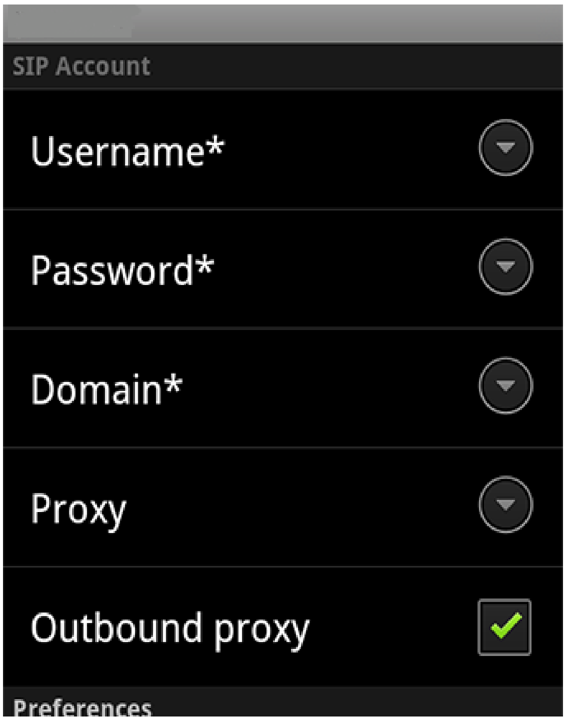
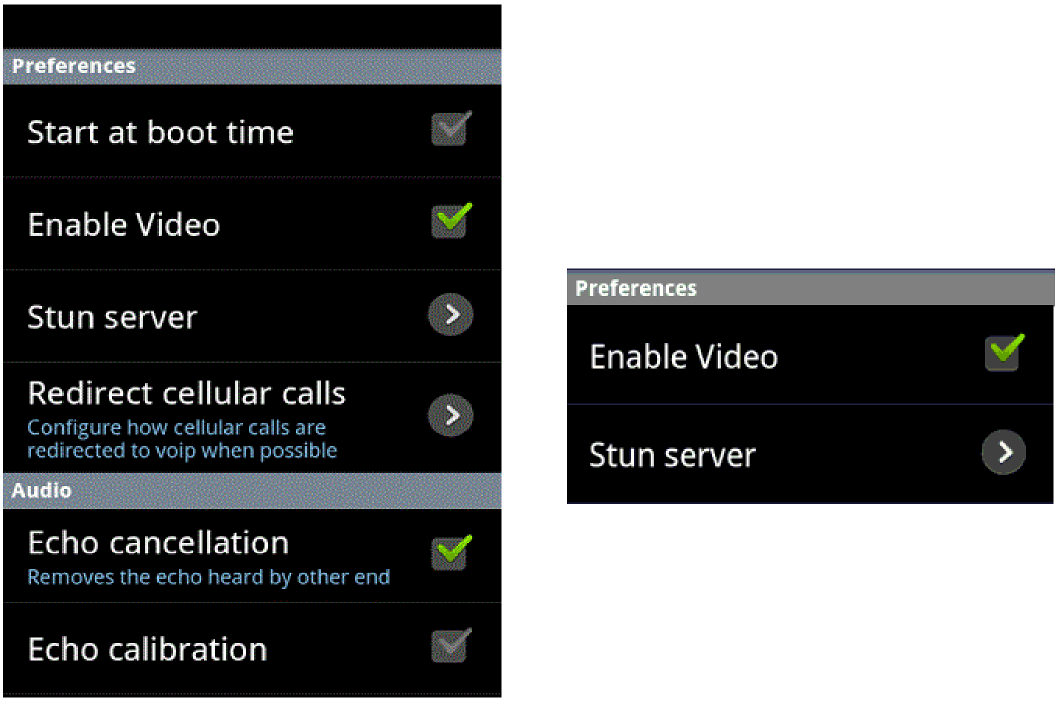


Figure 4.33 – Log-in Screen

Settings Page



Preferences

Start at boot time ☐

Enable Video ☑

Stun server ›

Redirect cellular calls ›
Configure how cellular calls are redirected to voip when possible

Audio

Echo cancellation ☑
Removes the echo heard by other end

Echo calibration ☐

Preferences

Enable Video ☑

Stun server ›

Figure 4.34 - Settings Screen

Map Display

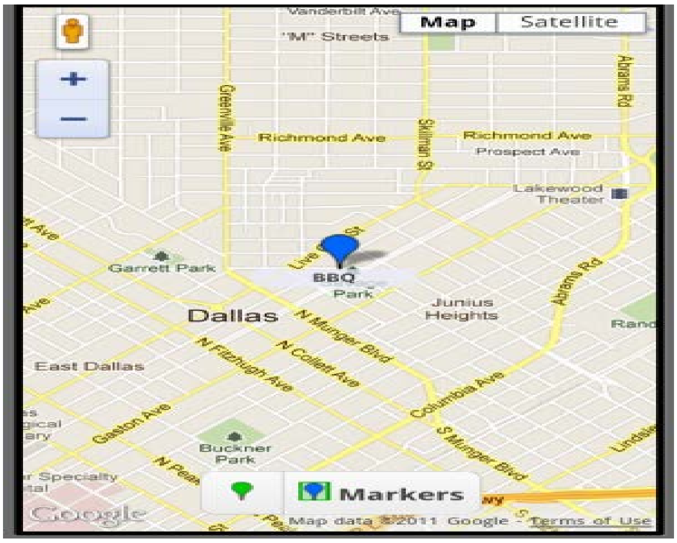


Figure 4.35 – Map Display

CHAPTER 5

SYSTEM IMPLEMENTATION

Implementation

The previous chapter discussed the design; inner details of the class attributes and methods, Use case diagram etc. Based on that design, this chapter will concentrate on the implementation details of the system.

We have categorized implementation details into Application Front End (User Interface) designing;and implementation of Video Calling Module, Location Tracking Module and DB Server. Therefore, in the remaining of this section implementation details regarding these modules have been presented.

User Interface Designing

This phase of the implementation involves developing the user end interface. Designing of an attractive user interface holds a preferred value because acceptance of the application by the Android users is dependent on it. A good and easily interactive interface can attract users to the application besides presence of other video calling or location tracking applications.

Video Calling Module Implementation

To implement video calling wehave used DubangoFramework. Stun protocol is used to resolve IP address NATing issue. SIP Server and SIP Protocol are used.

Location Tracking Module Implementation

Location tracking in our application is implemented using Google Maps, which are embedded in our application through Google Maps API. Blips are shown on the freshly captured GPS coordinates of the user of the application, and also on the location of its buddies whose coordinates are got from a DB Server (implementation details below).

DB Server Implementation

A DB server is implemented and maintained by us (the group syndicates). In this DB server GPS coordinates of all the users and their relevant data is saved so that their location could be tracked and shown on Google Maps.

CHAPTER 6

TESTING AND RESULTS ANALYSIS

Introduction

In this section of the document, system testing and load balancing test has been performed using different test cases. Software testing can also be stated as the process of validating and verifying that:

- A software program/application/product; meets the business and technical requirements that guided its design and development;works as expected; and can be implemented with the same characteristics.

- Software testing, depending on the testing method employed, can be implemented at any time in the development process. However, most of the test effort occurs after the requirements have been defined and the coding process has been completed. As such, the methodology of the test is governed by the software development methodology adopted.

Performance Testing

Test-Case – Video Call: Test Case ID: 1

Test-Case Description	Test Steps / Results			Test-Case Status	Test Status (Pass / Fail)	Test Priority	Defect Severity
	Step	Result					
		Expected	Actual				
This Test-Case is built to check that the Video Call Module is functioning properly. A video calling group of 10 people call one another 20 times. 8 users are on Wi-Fi and 2 users are on EDGE.	Check by entering SIPaddress in settings.	Successful Login	Successful Login	Designed	Pass	High	No Defect
	Check without entering SIPaddress in settings.	Error: SIP Stack is empty	Error: SIP Stack is empty	Designed	Pass	High	High
	Make a Video Call	Video Call should be established	Video Call is Established	Executed	Pass	High	No Defect
	Send Video	Video should be sent to another mobile	Video is received	Executed	Pass	Very High	No Defect
	End Call	Call should end properly	Call does not end properly	Executed	Fail	Low	Normal

Result

All 20 times call successfully using Android Mobile and statistics was each call setup took approx. 4 sec to establish call.

Load Distribution Testing

The software also tested with 2 Android mobiles which substantially reduced the time required establishes call between two users.

Summary

It has reflected the salient key points of performance and Load distribution test scenario which has shown the slight increase in time delay of video sending due to routing from GSM network to IP network.

Results and Analysis

Introduction

In this section of the document, we analyze our application and then compare it with applications having similar attributes. This does not include testing techniques rather it just gives the analysis.

VOCAL Salient Features: **VOCALhas achieved the following desired goals and outcomes;**

- Reduced cost of international calls up to 0 times.
- Integration of Google maps, SIP Server, and video calling.
- Users can track their own and their buddies' location.
- User friendly interface.
- Reduced Call Setup time.
- Implementation of new technologies like 3GPP.
- A Step towards 4G application development.
- VOCAL Comparison Analysis with State-of-the-art applications.

VOCAL Comparison Analysis

No	Features and Attributes	VOCAL	Skype, gtalk and MSN
1	Mobility	Yes	Yes
2	Flexibility	Yes	No
3	Combination of All	Yes	No
4	Location Tracking	Yes	No
5	Ease of Communication	Yes	No
6	Call Setup time reduce	Yes	No
7	Buddies Location	Yes	No

Location Tracking

Location tracking is the powerful feature of Vocal. Integration of Google Maps with video chatting application is great need. Now everyone can locate their own location and their friends. Location of other users provides the user to increase their interaction.

Summary

Vocal was compared with pre-existed state-of-the-art Video chatting and Instant messenger in terms of functionality and limitations in this chapter. It has reflected the salient features of Vocal and evaluated it with respect to its commercial applications.

CHAPTER 7

CONCLUSION AND FUTURE WORK

Introduction

This chapter describes the future scope of the project and the overall conclusion of the project. The project can be extended and few ideas are given in the chapter for the up gradation of the concept.

Conclusion

VOCAL has been developed, for the moment, to operate under GSM coverage only. There is certainly a need for using better techniques for compression in order to provide a better video and voice quality. Moreover, there is a requirement for preparing reusable components so that similar application for other platforms can also be launched. There is also need to address the buddy tracking during move. In all, it can prove an exciting application if greater bandwidth is available, which will vary from place to place and from time to time.

Future Work

Following extensions would be quite helpful in extending usability of the system and make it more acceptable.

Issue of Sky View

GPS coordinates can be acquired only if the mobile user is in open air and has sky view. This introduces a limitation for indoor uses. Some technique will have to be used to enable users to use VOCAL inside buildings.

Obtaining other Buddies and Buddy Locations during move

Acquisition of location coordinates of other users, while a user himself / herself is on move, is an issue. If it is desired that new other users are quickly shown as soon as they fall within range, then quicker algorithms will have to be used.

Reusability

Reusability is a basic requirement of software components. It is required that various modules are designed in a manner that they are reusable for developing VOCAL for Windows eytc.

APPENDIX A

GLOSSARY

CODEC (Coder-Decoder). CODEC is a type of software for encoding (compressing) and decoding (decompressing) digital media files i.e. songs or videos. Different available media players are used for creating and then playing such digital media files. Windows Media Player 12 is one popular CODEC.

EDGE (Enhanced Data Rate for GSM Evolution). EDGE is a mobile phone technology of GPRS family which is standardized by 3GPP (described below). It allows a high bit-rate data transmission (three time better capacity and performance as compared to GSM / GPRS connections). This is achieved through sophisticated methods used for data compression and transmission. A benefit of EDGE is that is conserves the available bandwidth by loading it with lighter (highly compressed) data.

GPRS (General Packet Radio Service). It is a standard for wireless communications which runs at speeds up to 15 kilobits per seconds (kbps). It supports a wide range of bandwidths and is suited for sending small bursts of data e.g. emails and web browsing content, as well as data of large volume. This standard is now maintained (specified) by 3GPP.

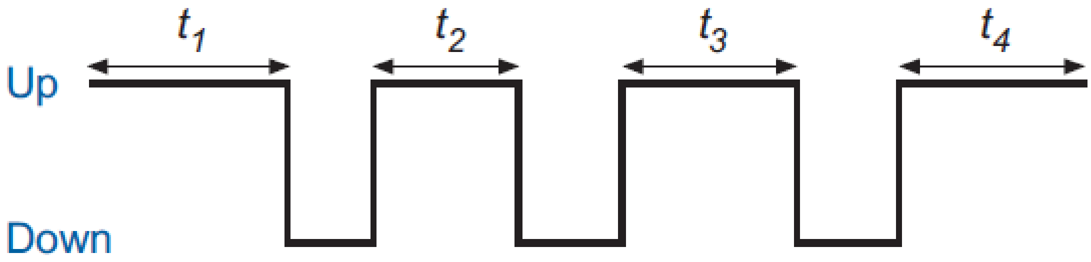
GSM (Global System for Mobile Communication). It is a standard for wireless communication which runs at speeds up to 9.6 kbps. It was developed for the 2ndGeneration (2G) Digital Cellular Networks which replaced the old analog AMPS technology. Many mobile phone sets still work on this technology, although it has been succeeded by the 3G Technology.

HTTP. Hyper Text Transfer Protocol.[1] It is the most widely used network protocol in the World Wide Web. It is a stateless, request response protocol in a client – server computer architecture. Through HTTP request and response messages, web content is exchanged between client and server systems.

Media Server. It is a dedicated machine or a software meant for storing and sharing media (video) content to client machines. It provides real time media processing functions. Hence, it is used to streaming video between two nodes. Usually it is web server of a large internet website but it may even be a specialized application which enable users to remotely access media files over the internet.

MDT (Mean Down Time) It is the mean of the total down time which is needed to restore a system to its full operational capabilities. It includes the time from reporting of a system / component's failure to the time it is made operational and returned.

MTBF (Mean Time Between Failures) MBTF is the mean Operating Time (up-time) between failures ofa specified item of equipment or a system. It is used for expressing the overall reliability of items of equipment and systems. In the following diagram it is the average value of t over the operating life of a particular system.



MTTR (Mean Time to Repair) It is the time needed to repair a failed hardware (or software) module. For operational systems, by repair, we usually mean replacing a failed part. Hence, MTTR can also be taken as the mean time to replace a failed module.

NAT (Network Address Translator). NAT is a method by which IP addresses are mapped from one realm to another. NAT is necessitated due to growing shortage of IP addresses and scaling of routing.An advantage of NAT is that the IP addresses of systems within a particular domain can be used by several other systems inside another domain. It is a router function so a dedicated router is configured to replace the IP address (in the headers of outgoing messages of systems inside a domain) with that of its own IP, and likewise for the incoming traffic.

PSTN (Public Switched Telephone Network)RTP (Realtime Transport Protocol): **RTP defines a standardized packet format for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications and web-based push-to-talk features.**

SIP (Session Initiation Protocol):SIP is an application-layer control (signaling) protocol

which is meant to create, modify and terminate sessions between participants. [8]

UML (Unified Modeling Language):UML is a visual language for specifying, constructing, and documenting the artifacts of a system. Most of the complex software designs are hard to be described textually. These can readily be conveyed through diagrams using UML.

3G (3^RD Generation Mobile Telecommunication):3G is the third-generation of mobile phone technology standards. Typical services associated with 3G include wireless voice telephony and broadband wireless data, all in a mobile environment. With the capability for high-speed wireless data transfer, 3G has enhanced / enabled several additional applications such as mobile video, secure mobile ecommerce, location-based services and mobile gaming.

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3 GPP(3RD Generation Partnership Project) 3GPP is acollaboration of different groups for the purpose of preparing, approving and maintaining Technical Specifications and Technical Reports for GPRS, 3G, EDGE and the later technologies of mobile phones.

APPENDIX B

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APPENDIX C

DISTRIBUTION LIST

This SRS will be distributed to the following people for future reference:

1. Department of CSE, Military College of Signals, NUST
2. Project Supervisor
3. Project Coordinator
4. Development Team (Team Lead)