

**VoIP MOBILE APPLICATION FOR GROUP OF USERS USING REACTJS AND ANDROID
STUDIO**

PROJECT REPORT

at

**Sathyabama Institute of Science and Technology
(Deemed to be University)**

Submitted in partial fulfillment of the requirements for the award of
Bachelor of Engineering Degree in Computer Science and Engineering

By

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DEPARTMENT OF COMPUTER SCIENCE AND ENGINEERING

SCHOOL OF COMPUTING

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BONAFIDE CERTIFICATE

This is to certify that this Project Report is the Bonafide work of **P VINAY REDDY** (Reg no: **38110399**) who carried out the project entitled “**VoIP MOBILE APPLICATION FOR GROUP OF USERS USING REACTJS AND ANDROID STUDIO**” under my supervision from June 2021 to November 2021.

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Submitted for Viva voce Examination held on: _____

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DECLARATION

I, **P VINAY REDDY** hereby declare that the project report entitled “**VoIP MOBILE APPICATION FOR GROUP OF USERS USING REACTJS AND ANDRIOD STUDIO**” done by me under the guidance of **Dr. K. Babu M. Tech., (Ph. D)**, at “Sathyabama Institute of Science and Technology (Deemed to be University), Jeppiaar Nagar, Rajiv Gandhi Salai, Chennai – 600119” is submitted in partial fulfillment of the requirements for the award of Bachelor of Engineering/Technology degree in Information Technology.

DATE:05/12/22

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SIGNATURE OF THE CANDIDATE

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ABSTRACT

This paper discusses the application of VoIP mobile application done by using React Js and android studio. Mobile applications are tremendously increasing due to huge utilization by vast range users. More than 100 million users on average using popular games. But almost every application does not give the feature to interact by voice with their pairs. The revolution of telecommunication system with new applications and services is accomplished by the Internet. Voice over Internet Protocol (VoIP) is the most renowned telecommunication service based on the Internet Protocol (IP). In the present paper, explanation of android app which solve the problem of not having interactive medium between gamers and students.

Keywords

Voice over Internet Protocol
Internet Protocol Telephony-IP Telephony
Internet Protocol
Online Processing User Service

TABLE OF CONTENT

CHAPTER NO.	TITLE	PAGE NO.
	ABSTRACT	i
	LIST OF ABBREVIATIONS	ii
	LIST OF FIGURES	iii
1	INTRODUCTION	9
	1.1 INTRO	9
	1.2 THE IMPORTANCE OF HUMAN INTERACTION	9
	1.3 VoIP Important	10
	1.4 DATASET	10
	1.5 Minimum Requirement	11
2	BACKGROUND	12
	2.1INTRO	12
	2.2 WORKING OF VoIP	12
	2.3VoIP Protocols	12
	2.4 VoIP-Enabled Services	13
	2.5 BACKGROUND AND HISTORY OF VOIP	13

3	DIFFERENT TYPES OF VOIP COMMUNICATIONS	14
		16
4	TOOLS AND LIBRARIES USED	17
	4.1 OPENCV	17
	4.2 FIGMA	17
	4.3 VSCODE	17
	4.4 REACT JS	18
	4.5 Android Studio	18
	4.6 Google Play Services	18
5	RESULT AND CONCLUSIONS	20
	5.1 RESULT	20
	5.2 CONCLUSION	21
6	REFERENCES	22
7	APPENDIX	23
	A. Architecture Diagram	24
	B. SOURCE CODE	25

LIST OF ABBREVIATIONS: -

	ABBREVIATION	MEANING
	VoIP	Voice over Internet Protocol
	Opus IP	Online Processing User Service Internet Protocol
IPT	Internet Protocol Telephony	
SDK	software development kit	

CHAPTER 1

INTRODUCTION

1.1 INTRO

VOIP provides an alternative by channeling digitized voice signals through IP networks, such as corporate intranets or, in certain situations, the public Internet. According to the survey, India has over 300 million active players, and the mobile gaming business generated \$1.5 billion in revenue in 2020, with revenues expected to reach \$5 billion by 2025. This project aims to empower people to create space in their lives where they may feel at home. We aim to make it easier for you to communicate with the people you care about on a regular basis. We want you to form true bonds with your friends and communities, whether they are local or global. Authentic, dependable, amusing, and relatable.

1.2 The Importance of Human Interaction

As technology's capabilities improve, so does our reliance on it. Technology advancements have aided growth and triumphs in a variety of fields, including business, science, medicine, and a slew of others. Socially, it has brought people together in a variety of ways by allowing us to communicate with one another at the touch of a button. It has enabled people to work remotely, travel while working, and stay in touch with colleagues when physically located in another state or country. This is a sentiment shared by many people, not just those in the medical field. Work isn't only about cashing a paycheck at the end of the month in any industry. It's about meeting individuals who share your interests, exchanging ideas, and working toward a common goal. Being in the same room as a client or colleague, shaking hands with them, or even sharing a cup of coffee with a new possible business partner allows you to have a clear picture of a scenario without having to assume what is intended through digital correspondence. It allows us to pick up on gestures, tone, and nuance, all of which might be misunderstood when communicating by email.

1.3 VoIP Important

One of the most essential things to remember is that VoIP isn't just for audio calls.

Indeed, there have been several attempts to modify this famous marketing slogan to properly reflect the fact that VoIP stands for voice, video, and data conferencing. Although all previous attempts have failed, keep in mind that video telephony and real-time text communication (RTT), for example, are clearly within the scope of VoIP. VoIP is crucial because, for the first time in over a century, there is a chance to significantly alter people's communication habits. We now have the option of employing pure IP-based phones, including desktop and wireless phones, in addition to the telephones we have today for real-time communication. We can also utilise videophones, which are similar to the ones shown in science fiction films. A person can call home to visit his or her family rather than calling home to talk to them.

One of the most intriguing features of VoIP is the ability to connect a standalone phone or videophone to a computer. One can utilise a computer only for voice and video communications (softphones), a telephone for voice and the computer for video, or just a computer with a separate voice/video phone to enable data conferencing services such as application sharing, electronic whiteboarding, and text chat.

1.4 DATASET

In a more advanced version, the dataset values can be locked until the user authenticates the dataset; in this scenario, when the user selects a dataset, the Android System launches an intent defined by the service to "unlock" the dataset.

When a data set contains credit card information (such as the number, expiration date, and verification code), for example, you may offer a dataset with the message "Tap to authenticate." When the user hits that choice, an activity will appear asking them to input their credit card code, and if they submit a valid code, the dataset will be "unlocked."

You may also leverage authorised datasets to provide the user with an interactive UI. If the auto-filled activity is an account creation screen, for example, you might utilise an authorised dataset to create a random password for the user.

For more information on the dataset authentication technique, see `Dataset.Builder#setAuthentication (IntentSender)`.

1.5 Minimum Requirement:

Windows

*64-bit Microsoft® Windows® 8/10

*x86_64 CPU architecture; 2nd generation Intel Core or newer, or AMD CPU with support for *a

Windows Hypervisor

*8 GB RAM or more

*8 GB of available disk space minimum (IDE + Android SDK + Android Emulator)

*1280 x 800 minimum screen resolution

CHAPTER 2

BACKGROUND

2.1 INTRO

You may make calls from your phone using a VoIP app. Over 3G/4G data consumption is only charged for. A local call may be charged by your mobile phone operator.

Android currently controls the vast bulk of the smartphone market. The platform's dominance is due in part to the large range of phones available and the fact that it is available from all of the major mobile carriers. Android smartphones, on the other hand, are available from every major carrier (AT&T, Verizon, Sprint, and T-Mobile) as well as a variety of manufacturers.

We are clearly not as reliant on mobile networks as we once were. A data signal can be received in the same locations as a mobile signal. People are also beginning to utilise Wi-Fi more often. Technological improvements have enabled us to communicate with others on the Internet in the same way that we would over the phone. Here's a list of some of the top applications for SIP and VoIP conversations if you're ready to swap your mobile network for something that's not only easier to use, but also cheaper and more efficient.

2.2 WORKING OF VoIP

It's quite simple to get caught up in a technical debate that most readers will find perplexing. The goal of this section is to present a high-level overview of Voice over IP (VoIP) for individuals who do not consider themselves experts in the field, with enough clarity to serve as a good introduction for most readers.

Many individuals have recorded a human voice or other sounds using a computer and a microphone. The procedure entails sampling the sound that the computer hears at a very fast rate (typically 8,000 times per second or more) and saving those "samples" in memory or a computer file. Each sample of sound is just a very tiny bit of the person's voice or other sound recorded by the computer. The computer has the wherewithal to take all of those samples and play them, so that the listener can hear what was recorded.

VoIP is based on the same idea, but the difference is that the audio samples are not stored locally. Instead, they are sent over the IP network to another computer and played there.

Of course, there is much more required in order to make VoIP work. When recording the sound samples, the computer might compress those sounds so that they require less space and will record only a limited frequency range. There are several methods for compressing audio, each of which is known as a "compressor/de-compressor," or simply CODEC. Many CODECs exist for a

range of applications (e.g., movies and sound recordings), and CODECs for VoIP are specialised for compressing voice, reducing bandwidth use dramatically as compared to an uncompressed audio stream. Voice CODECs are tuned to enhance spoken words at the expense of noises that fall outside of the human speech frequency range. When recorded music and other sounds are transmitted via a speech CODEC, they don't sound particularly well, but that's OK for the work at hand. However, there are several "wideband" CODECs that perform admirably with both speech and music.

2.3 VoIP Protocols

In order to provide VoIP communication services, a variety of protocols might be used. In this part, we'll look at the ones that are most frequent among the devices that have been deployed and are still being deployed today.

Real-Time Protocol (RTP) is a standard for exchanging audio and video packets between communicating computers that is used by almost every device on the planet. RTP is described by the Internet Engineering Task Force (IETF) in RFC 3550. RFC 3551 defines the payload format for a variety of CODECs, while payload format standards are also described in ITU papers and other IETF RFCs.

One of the areas of concern for people communicating over the Internet is the potential a person to eavesdrop on communication. To address these security concerns, RTP was improved upon with the result being called Secure RTP (defined in RFC 3711). Secure RTP provides for encryption, authentication, and integrity of the audio and video packets transmitted between communicating devices.

Various protocols must be used to locate the distant device and negotiate the manner through which media will travel between the two devices before audio or video material may flow between them. Call-signaling protocols, the most prominent of which are H.323 and Session Initiation Protocol (SIP), both rely on static provisioning, RAS (ITU-T Rec. H.225.0), DNS, TRIP (RFC 3219), ENUM (RFC 3762), and other protocols to discover other users.

Both H.323 and SIP have their roots in 1995, when academics were trying to figure out how to get

two computers to communicate in order to share audio and video media streams. Because those working on the protocol in the ITU moved rapidly to establish the first standard in early 1996, H.323 was the first commercial success. SIP, on the other hand, moved at a considerably slower pace in the IETF, with the first draught appearing in 1996 and the first acknowledged standard appearing in 1999. SIP was updated throughout time and re-published in 2002 as RFC 3261, which is now the industry standard for SIP

H.323 and SIP both allow users to perform the same thing in terms of multimedia communication (audio, video, or other data communication). H.323 and SIP, on the other hand, are designed differently, with H.323 drawing extensively from historical communication systems and being a binary protocol, whereas SIP is an ASCII-based protocol that does not inherit many of the information components prevalent in past systems. Supporters of each strategy have argued for years over which approach is superior, with varied outcomes.

Over the years, there have been a lot of papers debating H.323 vs. SIP, but most of the arguments have often been "religious" in nature (e.g., "ITU vs. IETF" and "binary versus ASCII"). Very few of the papers and reports have compared the protocol on the basis of functionality and what really matters: does the protocol do the job? The fact is, both can do the job, though each has certain strengths and weaknesses, but users cannot see those differences. Thus, from a practical point of view, the differences are irrelevant.

2.4 VoIP-Enabled Services

Many people have proclaimed that VoIP enables all kinds of new services that were never possible before. This is certainly true, though the hype far exceeds reality and what is practical. Even so, there are a number of new capabilities which are practical and will come forward as we continue to deploy VoIP systems.

Video telephony is probably the first new service that will come forward that helps set VoIP apart from traditional telephone systems. Service providers are already rolling out services offering video terminals to allow people to call friends and family using video-enabled phones.

The Internet allows one to potentially launch calls from the PC, determine the availability of friends and family members (called "presence"), control telephone services from the PC or tablet, etc. Some of these technologies exist in stand-alone form, while some are more tightly integrated -- often using some proprietary integration mechanism.

The one business application that VoIP, video telephony (or, videoconferencing), and instant messaging will enable is application sharing and electronic whiteboarding. The ITU has defined a suite of protocols (called T.120) to address this application and it has been used in tools like Microsoft NetMeeting. While NetMeeting met some success, it failed to gain wider market

adoption due to the fact that it was somewhat difficult to set up and use in a corporate environment. By having better integration with the phone and wider deployment of VoIP, businesses will probably find the ability to do application sharing and electronic whiteboarding very appealing in order to improve productivity. These kinds of services that are related to VoIP are most exciting.

2.5. BACKGROUND AND HISTORY OF VOIP

Most people know the VOIP through consumer of SKYPE that have found public recognition in recent years. However, SKYPE is just one example of implement of VOIP which have important technological history and close relationship whit telecommunications industry. The ideas of Voice over IP were first discussed in 1970 and was introduced in 1995 by the VOLCALTEC Israeli company. These basic systems were for connecting computers and they had to contain Sound Card – Speaker – Microphone – Modem – VOIP software. The software codifies and compresses audio signals and converts them into packets to be transmitted over the network. From 1970, telecom companies have begun to offer Actuator IP software for their telephony equipment. The human voice is an analog wave signal and historically calls had been created on the network of analog circuits which provided End to end link for every call, and had been known as switching circuits. Most of companies that provided telephone service were the public agencies that are typically part of the postal office service's country and these networks were identified as Post-Office Telephone System (POTS). Public Switched Telephone Network (PSTN) is the name generally given to the networks that were created by the phone companies. Between 1950 and 1990, analog systems were replaced to digital networks and telephone exchanges were done by high-speed leased lines. This exchange was used from digital technology computers and digital signal protocols such as ISDN but communication still established via circuit switching, and copper wires. Since 1990 the companies that have manufactured phone equipment and communications, have started to increasing use of digital data transmission ideas between exchanges through packets related to IP. From mid-1990 manufacturers of telephony equipment added IP capabilities to existing telephony switch (PBX) and recently have been developed computer software which enable consumers to install VOIP adapter on your regular telephones (in order to calls can be established on the PSTN network and through routing by GATEWAY on the internet network) [14]. Advances in VOIP technology were led to the availability to telephone software of computer by many software providers. Gateway servers and voice processing card are as the interface between PSTN network and Internet that enable users to make calls through

pc and the IP phone. Furthermore, calls can be established through handsets that are similar to phone and are connected to the IP-based network and have more features than traditional phone.

CHAPTER 3

5. DIFFERENT TYPES OF VOIP COMMUNICATIONS

There are several ways to implement this service that are as follows .

3.1 PC to PC

In this way, both the caller must be the computer or device able to execute VOIP applications commands such as PDA. And both sides have to be connected to internet at the moment of contact. In this way This IP address must be identical for both. In PC-to-PC way both directly communicate with other through a computer (Heads phone) which currently use Internet-based voice applications; see Fig. 1.

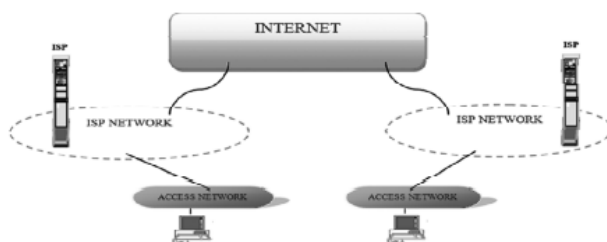


Figure 1: VOIP communications (PC to PC)

3.2 Phone to Phone (over IP Call)

In this way both are subscriber of PSTN services and use traditional phones for communication. There are two approaches to this type of communication; (i) This method uses from gateway. In this case, the caller uses the phone, but the call delivers through the gateway to management IP network and at side the recipient again converts to first state by a gateway (IP network can be fixed or wireless). (ii) In this case callers can use adapter that will have function similar to modem. In this situation the person connects by your phone that is connected to an adapter, the adapter sends the call to the PSTN but

here calls are driven via the ISP and Internet lines. At side of recipient the calls receive via the Internet and replay to adapter which adapter sends it to the phone that is connected to adapter. In this way, both sides of the conversation must be an ISP subscriber and access software must be installed on the adapter (both sides should use the same adapter; (see Fig. 2).

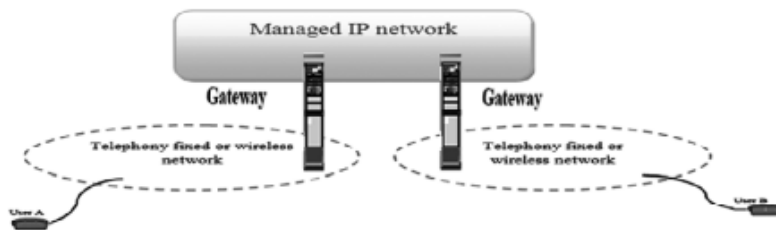


Figure 2: VOIP Communications (Phone to Phone)

3.3 IP Phone to IP

Phone In this way callers use from IP Phone (VOIP Phone) in order to the call will be transferred via IP networks and will not need to install the software or use the adapter or Media gateway. The Fig. 3 show that how to make calls using IP Phone and Media gateway.

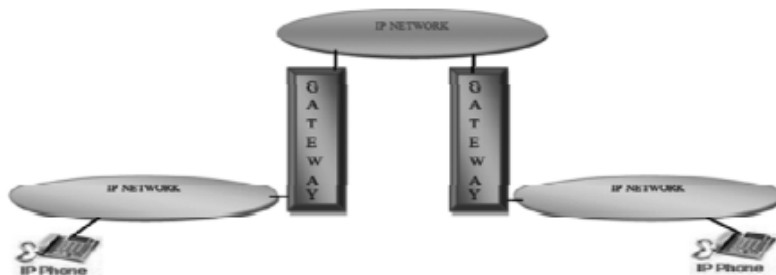


Figure 3: VOIP Communications (IP Phone to IP Phone)

3.4 PC to Phone

This method is actually a combination of the two previous methods. So, when someone uses the computer (headphone) to call, calls are transmitted via the Internet. Thus an (ITSP) receives call and using the gateway deliver call to the nearest point on the PSTN network and to phone devices (Fig. 4).

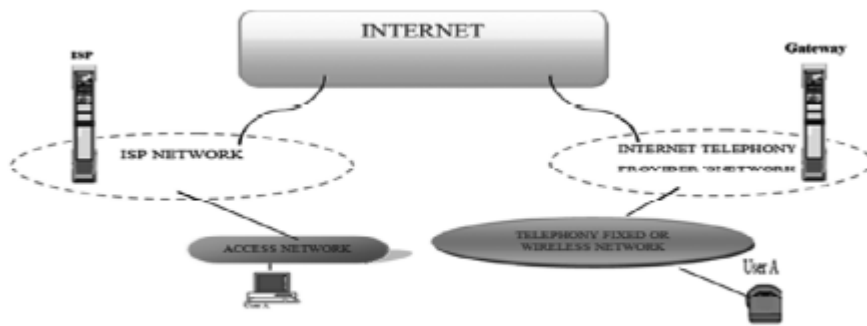


Figure 4: VOIP Communications (PC to Phone)

3.5 IP Phone to Phone

In this way, a caller uses the IP phone and calls would be transferred by gateway from the IP network to a PSTN telephone network, which it is in fact the other side of conversation and use the regular phone.

CHAPTER 4

TOOLS AND LIBRARIES USED

4.1 OPEN CV

OpenCV is the library we will be using for image transformation functions such as converting the image to grayscale. It is an open-source library and can be used for many image functions and has a wide variety of algorithm implementations. C++ and Python are the languages supported by OpenCV. It is a complete package which can be used with other libraries to form a pipeline for any image extraction or detection framework. The range of functions it supports is enormous, and it also includes algorithms to extract feature descriptors.

4.2 FIGMA

Figma is a vector graphics editor and prototyping tool which is primarily web-based, with additional offline features enabled by desktop applications for macOS and Windows. The Figma mobile app for Android and iOS allows viewing and interacting with Figma prototypes in real-time on mobile devices. The feature set of Figma focuses on use in user interface and user experience design, with an emphasis on real-time collaboration.

4.3 VSCODE

Visual Studio Code is a source-code editor made by Microsoft for Windows, Linux and macOS. Features include support for debugging, syntax highlighting, intelligent code completion, snippets, code refactoring, and embedded Git. Users can change the theme, keyboard shortcuts, preferences, and install extensions that add additional functionality. Instead of a project system, it allows users to open one or more directories, which can then be saved in workspaces for future reuse. This allows it to operate as a language-agnostic code editor for any language. It supports a number of programming languages and a set of features that differs per language. Unwanted files and folders can be excluded from the project tree via the settings. Many Visual Studio Code features are not exposed through menus or the user interface but can be accessed via the command palette.

4.4 REACT JS

React (also known as React.js or ReactJS) is a free and open-source front-end JavaScript library [for building user interfaces based on UI components. It is maintained by Meta (formerly Facebook) and a community of individual developers and companies. React can be used as a base in the development of single-page or mobile applications. However, react is only concerned with state management and rendering that state to the DOM, so creating React applications usually requires the use of additional libraries for routing, as well as certain client-side functionality.

4.5 Android Studio

Android Studio is the official integrated development environment (IDE) for Google's Android operating system, built on JetBrains' IntelliJ IDEA software and designed specifically for Android development.[8] It is available for download on Windows, macOS and Linux based operating systems or as a subscription-based service in 2020.[9][10] It is a replacement for the Eclipse Android Development Tools (E-ADT) as the primary IDE for native Android application development.

Android Studio was announced on May 16, 2013 at the Google I/O conference. It was in early access preview stage starting from version 0.1 in May 2013, then entered beta stage starting from version 0.8 which was released in June 2014.[11] The first stable build was released in December 2014, starting from version 1.0.

4.6 Google Play Services

Google Play Services is a proprietary background service and API package for Android devices from Google. When it was introduced in 2012, it provided access to the Google+ APIs and OAuth 2.0. It expanded to cover a variety of Google services, allowing applications to communicate with the services through common means.

CHAPTER 5

RESULT AND CONCLUSIONS

5.1 Result

This project is successfully connected people with a simple user-friendly UI and one touch call out feature to engage in productive time with individuals that they want to connected with.

VoIP offers advantages over traditional phone systems with its multi-party calling capabilities. Organizations large and small often need a separate conference call provider that enables many people to join the same call. Business VoIP providers include robust conference calling features built right into their service.

there are several advantages of using Design Thinking, we can summarize some of them. Safe process for confronting problems. A series of tools and techniques are designed to be used to provide solutions to a problem, both in companies and in other contexts.

The friendly platform interface provides better comfort and relaxation at the time of the class. And thus, be able to generate a better study environment.

5.2 Conclusion

Many institutions are already using of VOIP within their overall telecoms and data networking infrastructures and policies. Although many of these institutions are developing their voice services in an independent manner. In this paper we studied the basic VOIP features, including the Implementation Issues, Implementation and the various protocols required in the implementation of VoIP. We then discuss the Some factors that should be considered in implementing VOIP. The future work could be a detailed study on the Protocol Architecture of VOIP. Before the advent of VoIP technology, Internet users were expecting the risks associated with sending data over the Internet, but they were expecting a confidential network for their voice calls.

After the appearance of VoIP technology, and with the convergence of the voice and data networks, the data security issues became voice security issues as well. The fact that VoIP technology is using IP does not mean that it cannot be secured. To defeat VoIP security threats, a well-structured plan needs to be devised. The plan should include voice encryption, authentication, voice-specific firewalls, and separation of data and voice traffic. Redundancy in case of power loss should be considered in this plan as well [46]. It is also important that the voice servers and the other components of VoIP networks stay

physically secure from intruders. To protect VoIP networks, these general recommendations should be considered: Using IDSs and intrusion prevention systems (IPSs), using VoIP specific firewalls, changing default passwords on all different components of the VoIP system, patching components against known threats, and also following vendor's checklists for securing all components after installation. We have summarized the basic concepts on VoIP, and outlined some of its most important threats. As VoIP gains terrain in the world of telecommunications, it seems that soon it will become one of the dominant technologies in telephony. VoIP has inherited a number of Internet vulnerabilities exploited by malicious code, DoS, DDoS, and pharming, as well as flash crowds, all of which pose a latent threat to network infrastructures. It is evident that security policies have to be put in place to avoid disasters.

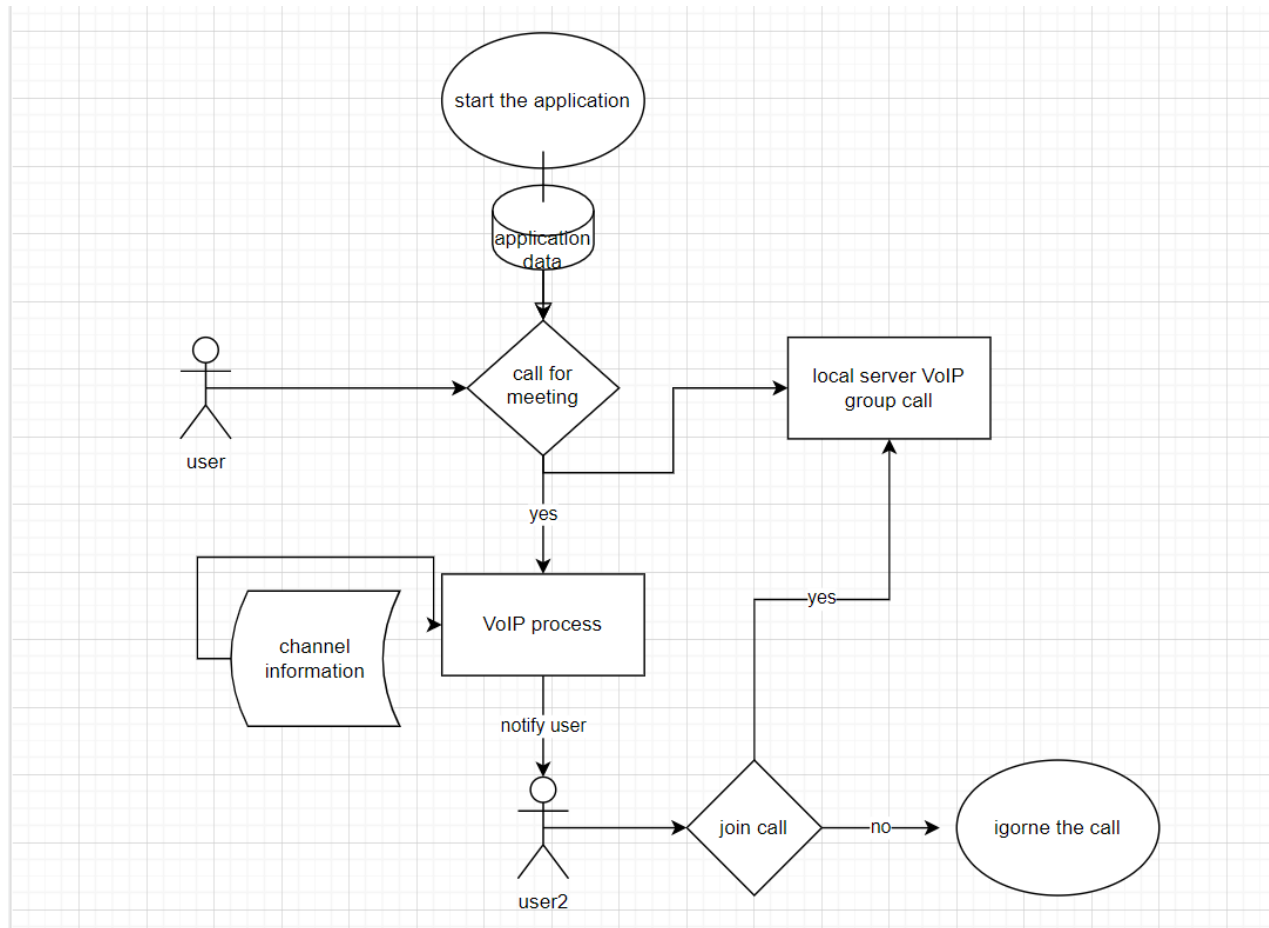
CHAPTER 6

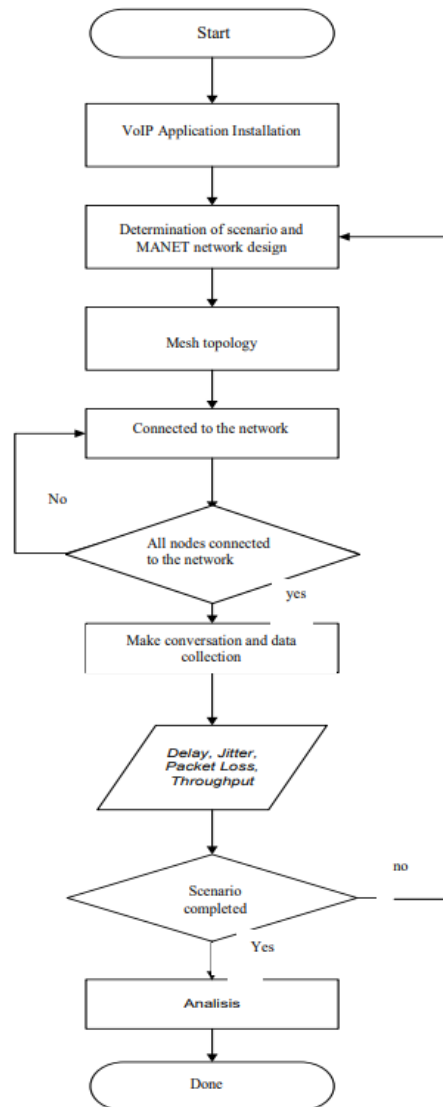
REFERENCES

[1] Barcel, J., Bellalta, B., Cano, C., & Sfairopoulou, A. (2008). VOIP Packet Delay in Single-Hop Ad-Hoc IEEE 802.11 networks. In *Wireless on Demand Network Systems and Services* (pp.

- 77–80). IEEE. [2] Chang, L. H., Sung, C. H., Chiu, S. Y., & Lin, Y. W. (2010). Design and realization of ad-hoc VoIP with embedded p-SIP server. *Journal of Systems and Software*, 83(12), 2536–2555. doi: 10.1016/j.jss.2010.07.053
- El Brak, S., Bouhorma, M., & Boudhir, a. a. (2012). [3] VoIP over VANETs (VoVAN): A QoS measurements analysis of inter-vehicular voice communication in urban scenario. 2012 5th International Conference on New Technologies, Mobility and Security
- [4]. Holzer and J. Ondrus, "Trends in mobile application development", *Proc. Mobile Wireless Middleware Operating Systems and Applications Workshops*, pp. 55-64, Oct., 2009.
- [5] M.P. Papazoglou, "Service-oriented computing: concepts characteristics and directions", *Proc. Fourth International Conference on Web Information Systems Engineering*, pp. 3-12, Dec., 2003.
- [6] J. Baliga, R. W. A. Ayre, K. Hinton and R. S. Tucker, "Green cloud computing: Balancing energy in processing storage and transport", *Proc. IEEE*, vol. 99, no. 1, pp. 149-167, 2011.
- [7] Barcel, J., Bellalta, B., Cano, C., & Sfairopoulou, A. (2008). VOIP Packet Delay in Single-Hop Ad-Hoc IEEE 802.11 networks. In *Wireless on Demand Network Systems and Services* (pp. 77–80). IEEE.
- [8] Chang, L. H., Sung, C. H., Chiu, S. Y., & Lin, Y. W. (2010). Design and realization of ad-hoc VoIP with embedded p-SIP server. *Journal of Systems and Software*, 83(12), 2536–2555. doi: 10.1016/j.jss.2010.07.053
- [9] El Brak, S., Bouhorma, M., & Boudhir, a. a. (2012). VoIP over VANETs (VoVAN): A QoS measurements analysis of inter-vehicular voice communication in urban scenario. 2012 5th International Conference on New Technologies, Mobility and Security - Proceedings of NTMS 2012 Conference and Workshops. doi:10.1109/NTMS.2012.6208691
- [10] Fahdi, D., Patih, J., Fitriawan, H., & Yuniati, Y. (2012). Analisa Perancangan Server Voip (Voice Internet Protocol) Dengan Opensource Asterisk Dan Vpn (Virtual Private Network) Sebagai Pengaman Jaringan Antar Client, 1(1), 42–48.

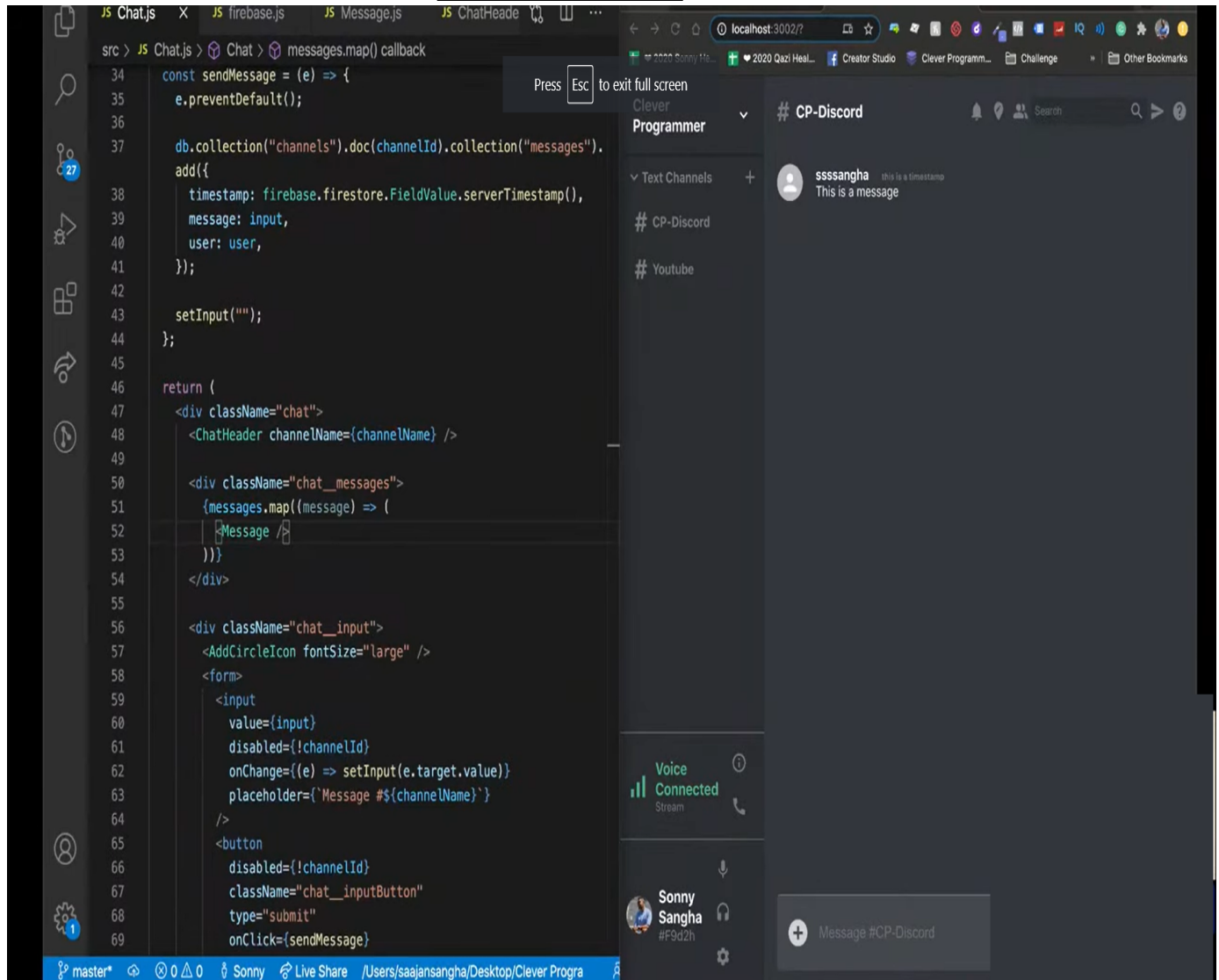
7.A Architecture Diagram





CHAPTER 7

B. SOURCE CODE



```

// Top-level build file where you can add configuration options common to all
sub-projects/modules.

buildscript {
    repositories {
        jcenter()
    }
    dependencies {
        classpath 'com. android.tools. build: gradle:2.2.3'

        // NOTE: Do not place your application dependencies here; they belong
        // in the individual module build.gradle files
    }
}

allprojects {
    repositories {
        mavenLocal()
        jcenter()
        maven {
            // All of React Native (JS, Obj-C sources, Android binaries) is installed from npm
            url "$rootDir/../node_modules/react-native/android"
        }
    }
}

import
React,
{Component
} from
"react";

import { AppRegistry, StyleSheet, Text, View} from "react-native";
//import { StackNavigator } from "react-navigation";
export default class Boiler extends Component {
    static navigationOptions = {
        headerStyle: {
            backgroundColor: "#16a085",
            elevation: null
        },
        headerLeft: null
    };
    render () {
        return <Text>OLA</Text>;
    }
}

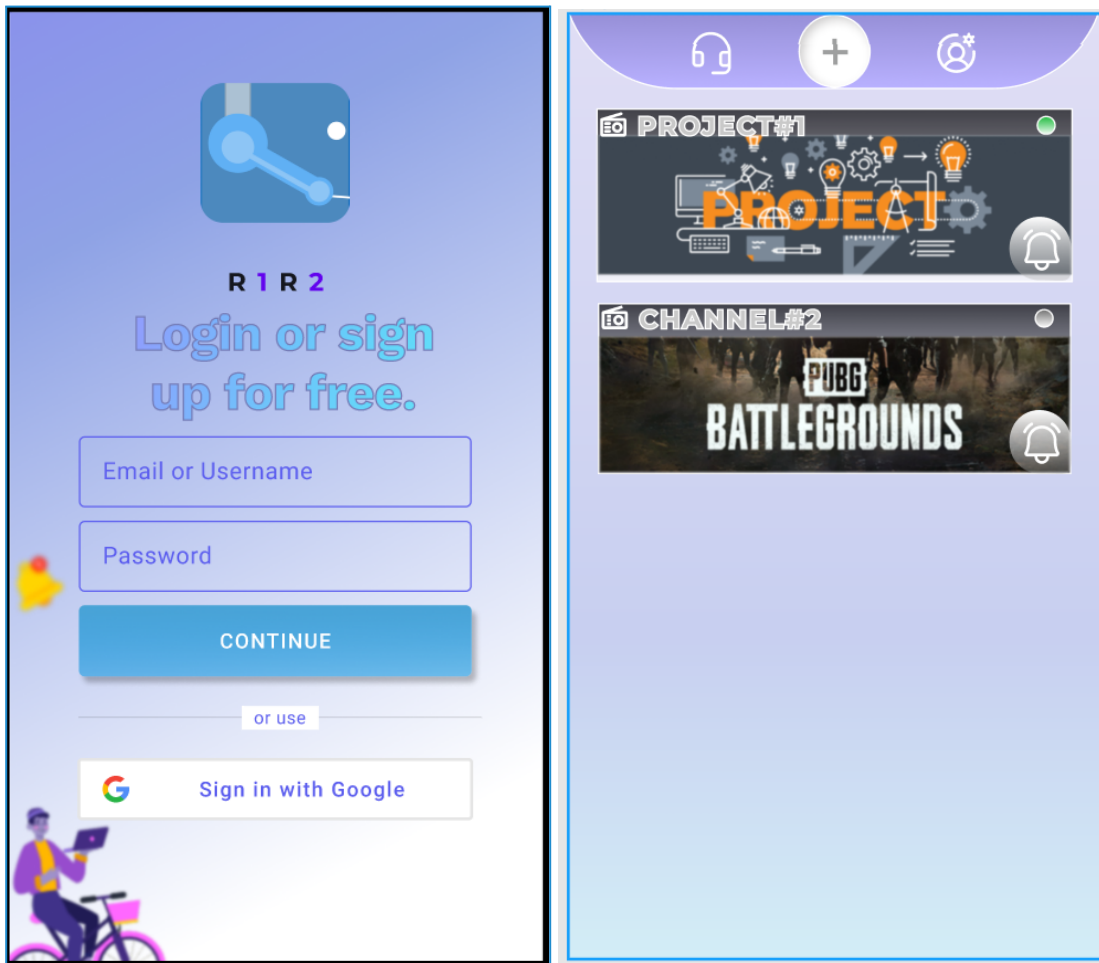
const styles = StyleSheet.create({});

AppRegistry.registerComponent("Boiler", () => Boiler);

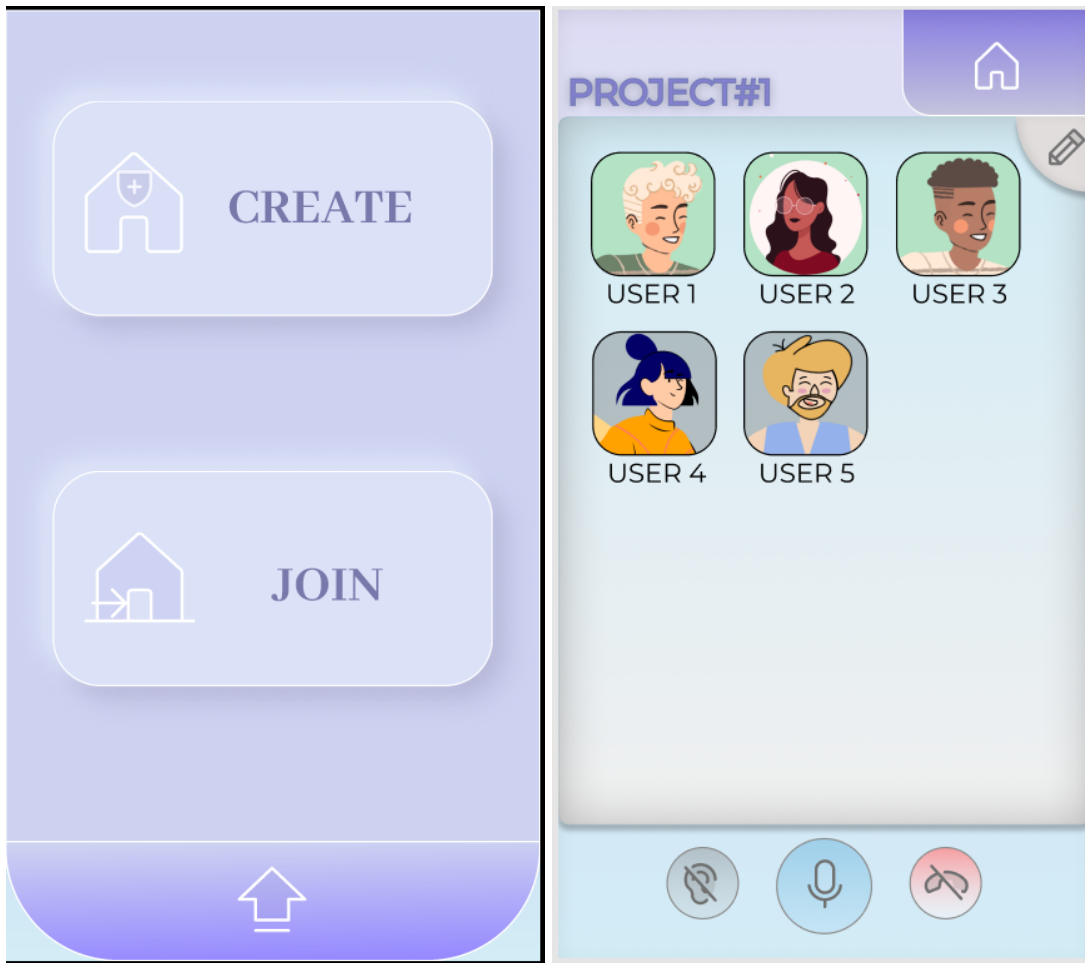
```



OUTPUT SCREEN SHOTS



OUTPUT



OUTPUT SCREENSHOTS

The image displays two side-by-side mobile application screens with a purple-to-blue gradient background. Each screen has a close button (a square with an 'x') in the top-left corner.

Left Screen (CREATE):

- Icon: A house with a plus sign inside.
- Title: **CREATE**
- Text: "Create the channel for your group"
- Form: "channel name" with a text input field containing "EX:NAME#1234".
- Form: "Passcode" with a text input field containing "EX:1234".
- Form: "members limit" with a numeric input field containing "5".
- Form: "About" with a text area containing "optional description about this voice channel ...".
- Button: A large purple button labeled **CREATE** at the bottom.

Right Screen (JOIN):

- Icon: A house with an arrow pointing inside.
- Title: **JOIN**
- Text: "JOIN the channel that has been already create"
- Form: "channel name" with a text input field containing "EX:NAME#1234".
- Form: "Passcode" with a text input field containing "EX:1234".
- Button: A large purple button labeled **JOIN** at the bottom.

SCREENSHOTS

