Development of Low Cost Voice over Internet Protocol (VoIP) Service for Marine Fishermen and Performance Evaluation

A PROJECT REPORT

submitted by

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under the guidance of

Prof. Sethuraman N Rao

in partial fulfillment for the award of the degree of

BACHELOR OF TECHNOLOGY

in

ELECTRONICS AND COMMUNICATION ENGINEERING

AMRITA SCHOOL OF ENGINEERING
AMRITA VISHWA VIDYAPEETHAM
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AMRITA SCHOOL OF ENGINEERING

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AMRITAPURI (INDIA)

BONAFIDE CERTIFICATE

This is to certify that the project report entitled **"Development of Low Cost Voice over Internet Protocol (VoIP) Service for Marine Fishermen and Performance Evaluation"** submitted by **Nadella Harshith (AM.EN.U4ECE16504)** and **Nagamalla Charitha (AM.EN.U4ECE16136)** in partial fulfillment of the requirements for the award of the Degree Bachelor of Technology in Electronics and Communication Engineering is a bonafide record of the work carried out by them under my guidance and supervision at Amrita School of Engineering, Amritapuri.

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TO WHOMSOEVER IT MAY CONCERN

This is to certify that Mr. Harshith Nadella was working as an Internship Trainee in Amrita Center for Wireless Networks & Applications (AmritaWNA), Amrita Vishwa Vidyapeetham, Amritapuri Campus from 1st January 2020 to 30th April 2020.

As part of his internship, Mr. Harshith Nadella was working on **Low Cost VoIP Service** for the Ocean Net Project. He has knowledge of embedded systems using Raspberry Pi, BeagleBone and NodeMCU. Able to use the Python programming language and linux operating system required for the embedded systems very well.

Mr. Harshith developed a VoIP Server for this service and deployed it to the OceanNet Offshore backend servers. The test results that were captured, improved the efficiency of the system. Additionally, he developed a VoIP Client to deploy on all boats where the Ocean Net device was being used. Gained knowledge of Amazon Web Service Elastic Instance to deploy cloud servers. Mr. Harshith has presented a conference paper at WISPNET 2020 Conference, Chennai. a paper at 3MPT, Concentia 2020 organised by Indian Institute of Space Science and Technology, Trivandrum.

Apart from these technical activities, Mr Harshith participated in hackathons and other competitions. Shortlisted to participate in the semifinal of the India Innovate Design Contest (IICDC) organised by Texas Instruments and Department of Science and Technology, India , winning $200 worth TI Products in this contest during this process.

He has good technical knowledge, good analytical skills and is capable of handling the given assignments very well and eager to learn new concepts making him an asset for any organization he wishes to join.

Sincerely,

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TO WHOM IT MAY CONCERN

This is to certify that **Ms. Nagamalla Charitha**, a student of B Tech, from **Amrita Vishwa Vidyapeetham** has successfully completed 5 (Five) months (From 6th January 2020 to 29th May 2020) internship programme at TheMathCompany Pvt Limited. During the period of the internship programme with us he/she was found punctual, hardworking, and inquisitive.

We wish you every success in life.

For TheMathCompany Pvt Ltd.

Ashish Thomas Sam
Senior Partner – People & Operations

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DECLARATION

We, **Nadella Harshith (AM.EN.U4ECE16504), Nagamalla Charitha (AM. EN.U4ECE16136)** hereby declare that this project report entitled **"Development of Low Cost Voice over Internet Protocol (VoIP) Service for Marine Fishermen and Performance Evaluation"** is the record of the original work done by us under the guidance of **Prof. Sethuraman N Rao** Associate Professor, Amrita Center for Wireless Networks and Applications, Amritapuri; **Mr.Sai Shibu N B** Research Assistant III, Amrita Center for Wireless Networks and Applications, Amritapuri and **Gayathri Narayanan** Assistant Professor, Department of Electronics and Communications Engineering, Amrita School of Engineering, Amritapuri to the best of my knowledge this work has not formed the basis for the award of any degree/diploma/ associateship/fellowship/or a similar award to any candidate in any University.

Place: **Signature of the Students**
Date:

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We wish to express sincere thanks to our Chairperson, Dr. Ravishankar for giving us the support and encouragement that was necessary for the completion of this major project.

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Last but not the least, we would like to thank our Beloved Chancellor, Sri (Dr.) Mata Amritanandhamayi Devi for providing us the guidance and a friendly environment to learn and work.

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MHz	Mega Hertz
GHz	Giga Hertz

List of Abbreviations

VoIP	Voice Over Internet Protocol.
SIP	Session Initiation Protocol
PBX	Private branch exchange
CapEx	Capital Expenses
OpEx	Operating Expenses
TCP	Transmission Control Protocol
UDP	User Data-gram Protocol
VHF	Very High Frequency
Wi-Fi	Wireless Fidelity
LAN	Local Area Network
WAN	Wide Area Network
P2P	Peer to Peer
CPE	Consumer Premise Equipment
DTMF	Dual Tone Multi Frequency
IP	Internet Protocol
PC	Personal Computer
OS	Operating System
QoS	Quality of Service

Abstract

Fishermen travel far up to 50 nautical miles into the ocean for fishing. They are often stressed and worried about their families as they cannot contact them for many days. Also, it is very difficult to pass any information regarding cyclone or other natural calamities to the fishermen. As there is no scope for mobile communication offshore beyond 8-9 nautical miles due to high operational expenses(OPeX). Traditional communication systems like Marine VHF radios and satellite phones are costly and restricted to only the boats in the ocean. This paper proposes a cost-effective, full duplex communication system for the marine fishermen to communicate among the boats in the ocean as well as with their families on the shore. This paper also presents the performance evaluation results of the system setup in a lab environment. The results obtained from the tests are satisfactory and the system can be implemented on the boats.

Chapter 1

Introduction

1.1 Introduction

The transition in the way of communication between the users began in the early 19th century. The Radio technology was still in its infancy in 1912, it was limited in the Morse code for transmissions. Most of the radio transmitters at the time were called "spark" transmitters which depended on electrical energy sparks for transmitting signals. This form of transmitter could not continuously emit radio signals, making voice messages virtually impossible. Spark communications also covered a large bandwidth, making interference from outside messages inevitable. As a result, radio was considered an unreliable form of emergency communication, despite being marketed as maritime technology. The transition continued and after the radio waves were adapted into a communication system in the late 19th century, Marine VHF Radio also called as VHF Amateur Radio was invented. The use of this Amateur Radio was started by the marine fisherman for the communication among the nearby boats. According to the article [1], the Titanic sinking had some inevitably large effects on radio and broadcasting. Only

four months after Royal Mail Ship Titanic was lost, American government passed the Radio Act of 1912 to gain control over the airwaves and all operators to hold a valid federal license to use radio equipment and it also restricted amateur users to bands less than 200 meter wavelengths far below the official maritime communications would be conducted, which reduced the chances of interference with transmissions.This way of communication is not efficient for long range communications. OceanNet, developed by our research center, is a commercially available internet service for the marine fishermen, across the Arabian sea. The OceanNet system uses proprietary long range WiFi technology to provide internet service in the ocean for up to 100km.Long Range WiFi Base Stations are deployed along the shores and the boats subscribing to this service are provided with a long range consumer premises equipment and control box. Fisherman can connect their mobile phones to the WiFi on this system to connect to the internet when they are in the ocean. This project presents an efficient, cost-effective way of communication for marine fisherman through Voice over Internet Protocol (VoIP) technology through the use of long range WiFi [2], [3] and [4] services. The OceanNet system consists of a control system to automate the antenna orientation and track the boats in real time using GPS. The control system is developed using Raspberry Pi [5]
.

A fisherman community in the Azheekal village, near Kollam, Kerala, India was considered to understand the current system being used by the fisherman. A survey was conducted to understand the issues faced by them and their requirements for a better system. Figure 1.4 shows the VHF Amateur Radio system used on the boat.

Figure 1.1: Marine VHF System deployed on the boat

The Amateur radios use the spectrum of radio frequencies to exchange non-commercial messages that are used primarily for emergency broadcasts and experimentation purposes. VHF Amateur radios operate within frequency range from 156MHz-172MHz [6]. Marine VHF Radios operating at these radio frequencies are affected by changes in weather and terrain conditions, which require a power source for their operations and are not user-friendly. On account of these drawbacks of using marine VHF radios, there is a need to use VoIP service to provide a cost-effective mode of full duplex communication for the marine fisherman that can solve the disadvantages of using VHF Amateur Radios for communication.

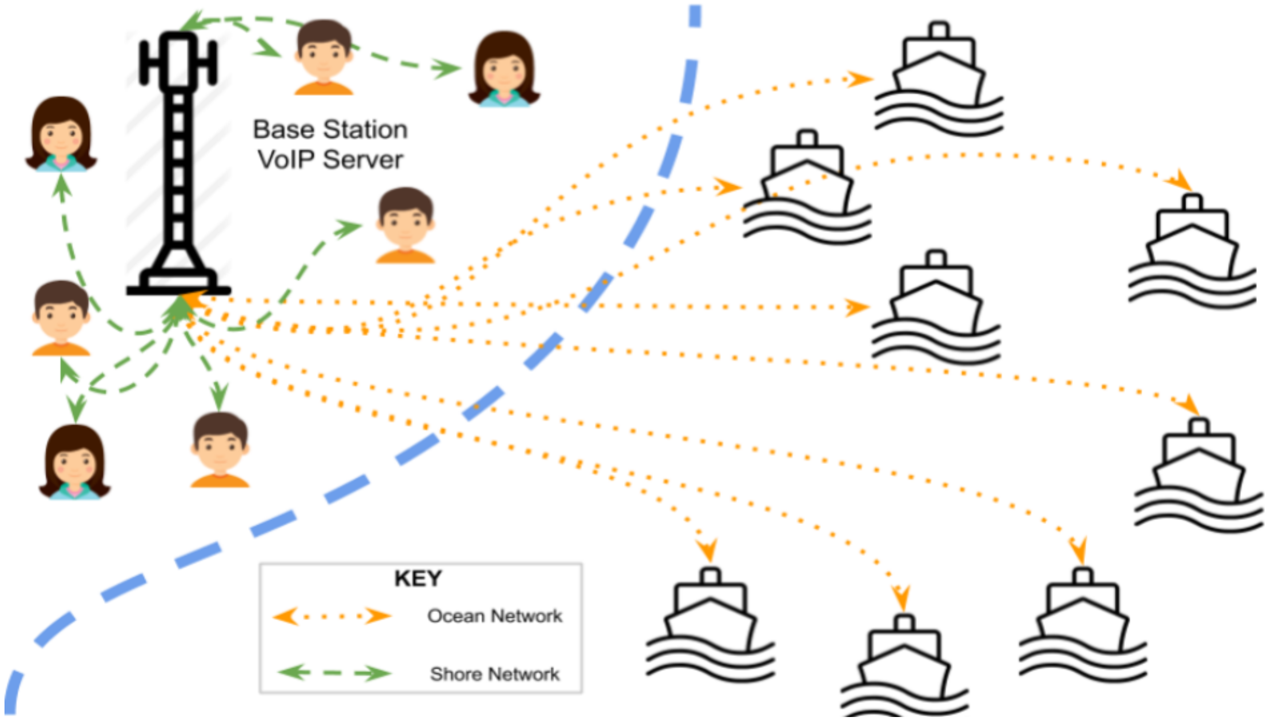


Figure 1.2: Network diagram of the VoIP Service: Version 1

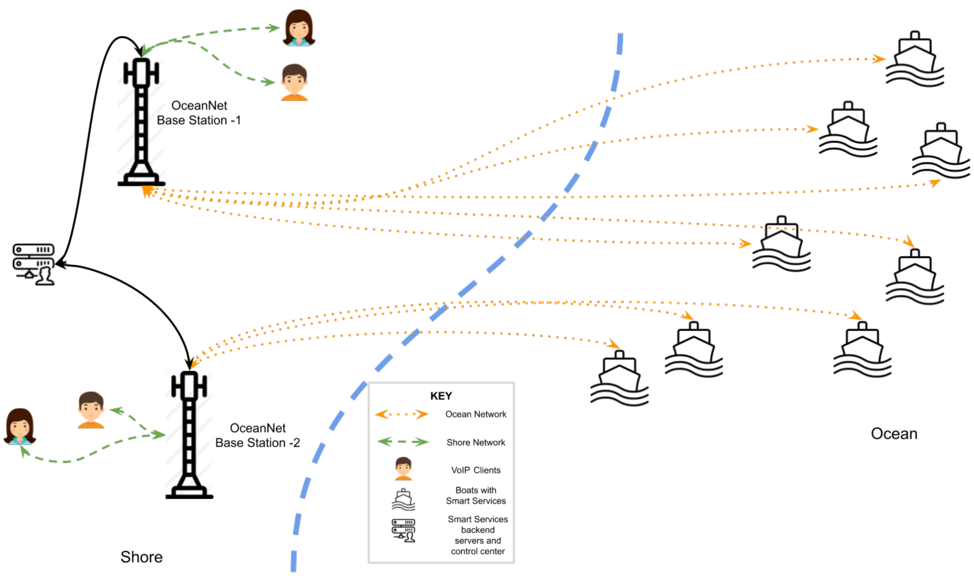


Figure 1.3: Network diagram of the VoIP Service: Version 2

1.2 Proposed System Architecture and Configurations

This section explains about the low cost VoIP system developed for the marine fisherman. The system uses the existing OceanNet system as the back-haul network. The proposed system consists of a Raspberry Pi Model 3 and a VoIP Software to route calls between the boats. The VoIP server is deployed at the base station and the client is deployed on the boat. The Raspberry Pi 3 is a single board computer with 1.2GHz ARM Processor, 1GB RAM, up to 32GB Storage, on board WiFi, Bluetooth and Ethernet. The power consumed by the Raspberry Pi is around 10 watts at maximum CPU utilization. The operating system is a lightweight Linux based Raspbian OS. Major programming languages such as Python, C, Java are supported. The VoIP technology requires an IP based private branch exchange (PBX) for routing the calls between the clients. Asterisk is an open source framework that can be configured on the Raspberry Pi and used as the VoIP Server. Asterisk uses Session Initiation Protocol (SIP) for routing the calls. SIP is an application layer protocol that works in conjunction with other application layer protocols to control multimedia communication sessions over the Internet. This is a standardized protocol which establishes either of TCP or UDP connections between the end users for the data transfer. A VoIP call, which is the exchange of data between the clients is called a session. The VoIP server is capable of handling multiple simultaneous calls, conference calls and video calls. Figure 1.2 and 1.3 shows the network diagram of the VoIP Service.The two network scenarios where the VoIP server placed at the OceanNet base station and the VoIP Server placed at the data center of the OceanNet can be observed from the Figure 1.2 and 1.4. From the network scenario Version 1 it is seen that the network coverage of the single OceanNet Nano station can have the access of VoIP Server but as there are multiple OceanNet Pilot deployments which has an extended network coverage for multiple regions, the network scenario is changed to version 2 as that the VoIP Server developed was placed at the data center of the OceanNet.This scenario can be followed for the deployment as that multiple users using the nth nano-station can also be benefited and have the access of VoIP.This can be observed from Figure 1.4

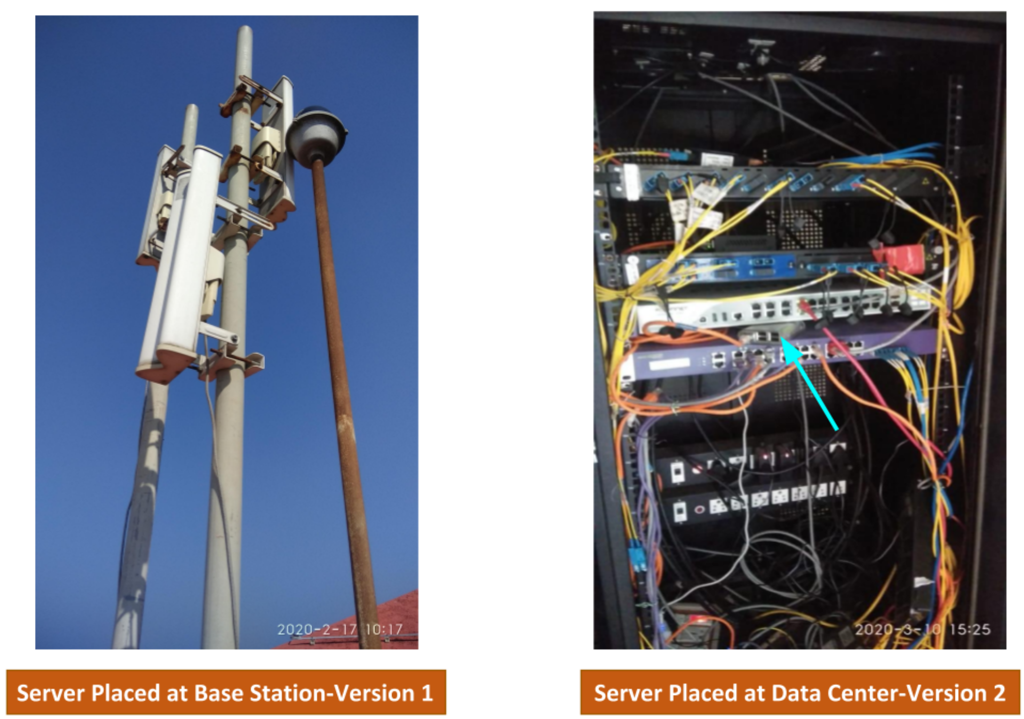


Figure 1.4: Different Scenarios of Network Operation of VoIP Server

1.3 Review of State of the Art

This section briefly explains similar work executed by other organisations and research institutes. The heterogeneous VoIP clients interconnection model was explained in [7] shows the transition in the VoIP protocols. This also discusses about the H.323 which was the first generation VoIP protocol. H.323 is an "umbrella" specification proposed at first in 1996 by ITU-T and has been updated several times. It suggests a standard for multimedia communication over packet switched networks such as LANs, WANs and internet.The performance comparison between Skype P2P VoIP and traditional server-based VoIP applications was discussed in [8] and the network topology used by the Skype P2P and Server-based VoIP applications were discussed. This paper compares the parameters such as connection setup delay, end-to-end delay and jitter between the Skype P2P VoIP and server-based VoIP applications.The authors in [9] studied the impact of G.711 and iLBC codec and the influence of packet loss on VOIP characteristics using the databases. The databases for training and testing are created using Asterisk. The paper has discussed about the framework for VOIP database generation.

In [10], the authors have discussed about developing HD-VoIP android applications with G.711.1 wideband coder. The coder G.711.1 can interoperate with the G.711 coder and it is one of the wideband coders which is the most suitable to the coexistence with legacy VoIP terminals. Most of the applications and technologies discussed are intended for land use and they need a very stable internet connectivity. As of now, there exists no solution other than Marine VHF Radio and Satellite phones to communicate with the fisherman. Therefore, it is needed to have a low cost and reliable communication system that helps the fishermen to stay connected with his family, when they are away from home.

1.4 Problem under Investigation

Long range communications for the maritime communication are limited to a few nautical miles as the network operators are not able to bear the expenses.The two technical terms CapEx: Capital expenses and OpEX: Operating expenses are taken into consideration while deciding the revenue by the network operators.Unfortunately as there are less users utilising the band when compared to the onshore communication and so the revenue they get from each user is less.Hence the cellular communications are limited to 15 to 20 nautical miles and beyond that there is connectivity for the fishermen to communicate with their families.The fishermen are using some legacy systems such as VHF Amateur Radios and satellite links which provides simplex communication through which they can communicate with the people onshore. OceanNet is a commercial network which provides the connectivity for fishermen up to 50-60 nautical miles.It only provides the connectivity for the fishermen within 60 nautical mile range but does not have a system to handle calls with its individual database just like the Cellular Network operators have. The idea is to develop a VoIP Server which takes the support of OceanNet Backhaul Network and provide cost-effective mode of Full Duplex communication for the marine fisherman which can overcome disadvantages of current system so that the maritime communication can be improvised.

1.5 Performance analysis of the VoIP Server

1.5.1 Call Latency of VoIP Server

Table 1.1: Latency Range for VoIP Services

Latency Ranges (μS)	Performance
0 - 30000	Exceptional
30010 - 50000	Excellent
50010 - 80000	Good
80010 - 120000	Acceptable
More than 120000	Poor

Depending on the Codecs and the hardware configurations, VoIP calls usually have the latency of around 20ms to 150ms [11].To Calculate the Call latency of the VoIP Server developed, two ways of latency calculations were done

A. Latency Calculations using Cyclic-test Tool

In [12], a network monitoring tool, Cyclic-test was used to perform the latency test for the configured VoIP server. Table 1.1 summarises the latency range of VoIP Calls [12]. Cyclic-test tool can operate the server for the fixed duration of time and gives the real-time mean Latency Values considering the delays at different time intervals. It can be installed as a Open Source Python library in Ubuntu. The Performance was calculated after testing the Server for a duration of around 160 minutes by performing four stages of testing procedure as mentioned in Table 1.2.The server setup on Raspberry Pi in the test network with Codec G.711 (Mu-Law), had a latency of 56.82ms as observed from Figure 1.3 which falls under the acceptable category and close to excellent range.

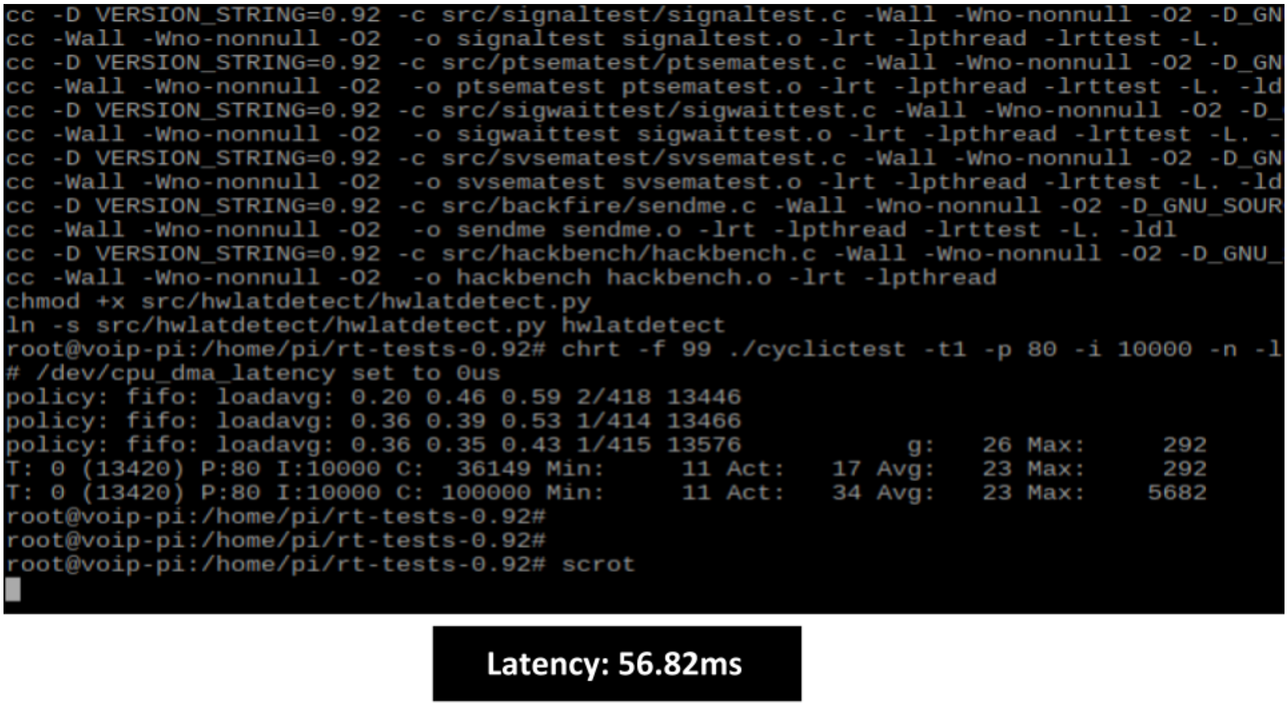


Figure 1.5: Latency Calculations Using Cyclic-test tool

B. Latency Calculations by capturing the VoIP Call data in Wireshark

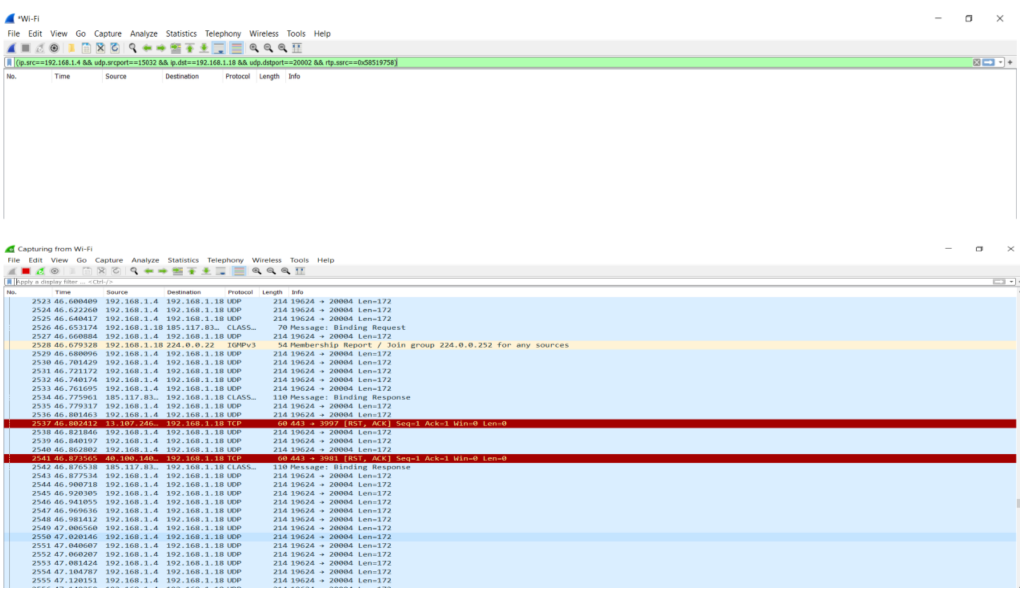


Figure 1.6: Packet Capture for VoIP Calls in Wireshark

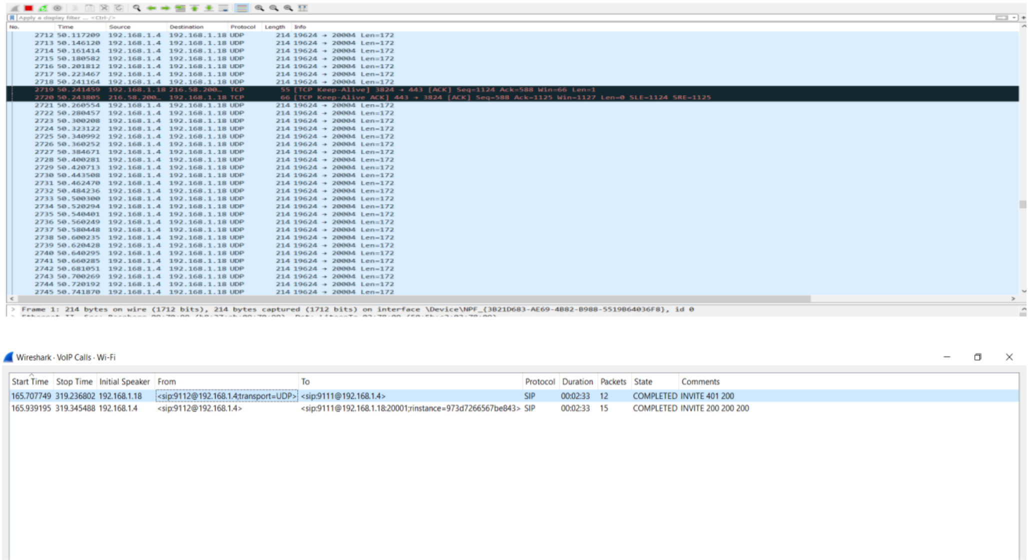


Figure 1.7: VoIP Call Logs observed in Wireshark

Figure 1.8: Audio Signal with Delay Variations in a VoIP Call:1

Wireshark is an application or tool through which we can monitor the network packet flow for a particular network.The tool has an advantage of filtering out the packets based on the protocol which can be observed from Figure 1.4 and the call logs of the VoIP calls made can also monitored in wireshark (Refer Figure 1.5) VoIP Calls uses the support of SIP Protocol and as the VoIP call is made through UDP as it is live audio or video transmission.Filtering out the network packet flow for the SIP,UDP and RTP protocols will give the data corresponding to the VoIP call made. Wireshark can generate the voice signal of the VoIP Calls with the feature of live VoIP Call in Wireshark and through the generated voice signal given by the tool,the delay variations can be observed as seen in Figure 1.6 and 1.7 for the two VoIP Calls as a part of observation whose duration was about 150 seconds.

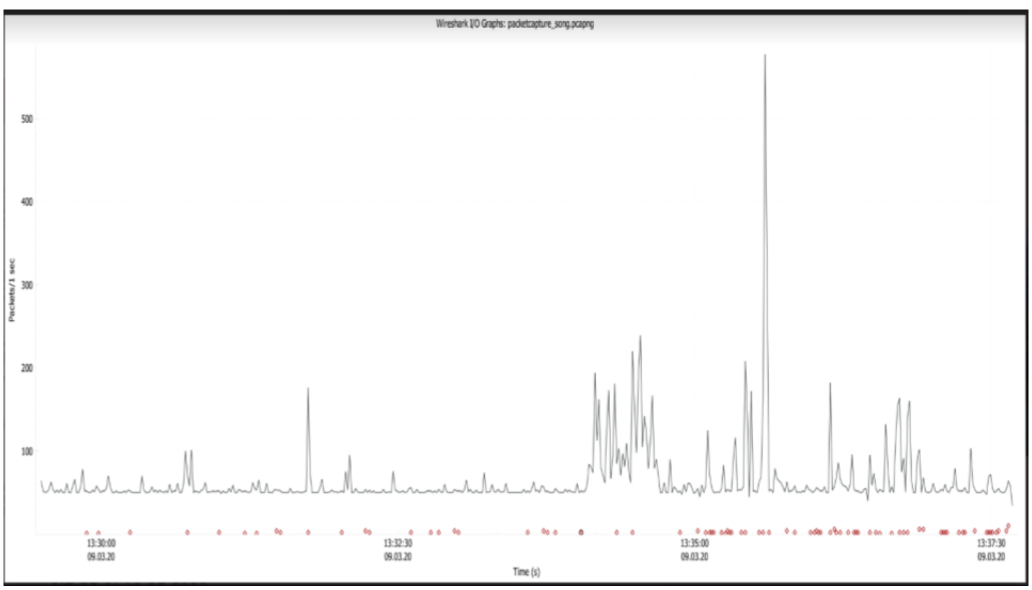


Figure 1.9: Audio Signal with Delay Variations in a VoIP Call:2

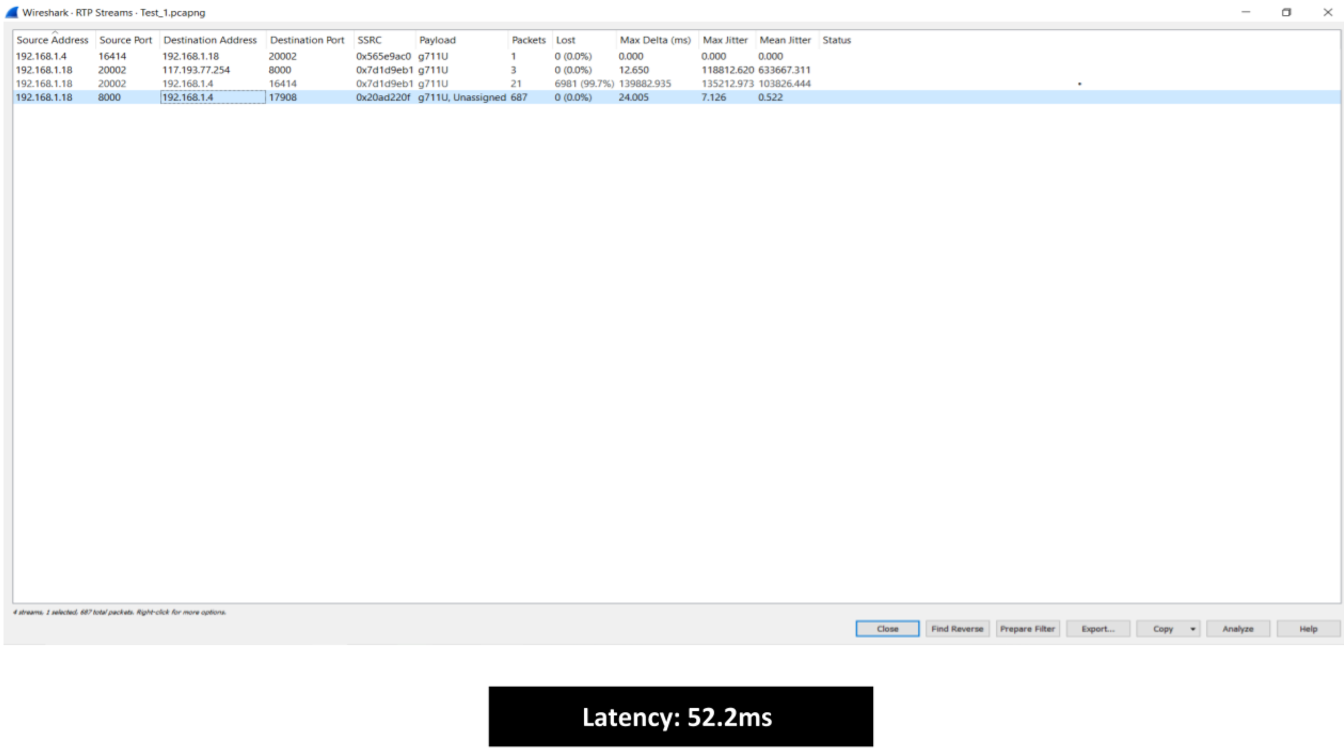


Figure 1.10: Latency Calculations observed in Wireshark

It is observed from the Figure 1.9,the delay drops at a particular second which are often called as jitter drops can be seen as Red color dots.These dots indicates that the delay was seen at that particular second over a duration of time.Similarly,the orange dots in the Figure 1.7 are known to be the jitter drops of the audio signal of a VoIP Call as indicated in the graph.The mean jitter buffer or the latency of the VoIP Call is found to be "52.2ms" similar to the latency calcualtion done using the Cyclic test it is seen that there is some variation in the delay but considering the Standard latency values the delay measured using the Wireshark is much more accurate and the performance was good and close to excellent.

In addition to the two ways discussed above there is another possibility of calculating latency of server which is through Third party Soft-Phone Applications which gives the live delay in the active call which comes with a disadvantage that the net delay of the call made cannot be found.It gives the quality of the call graded out of five as observed in Figure 2.9, which is often called as QoS grading of VoIP call.

1.5.2 CPU Load on VoIP Server

This section explains the performance test setup and discusses the results. For the test setup, the Raspberry Pi is configured as a VoIP Server and connected to a WiFi Router as shown in Figure 1.10

Table 1.2: VoIP Server logs

Stage	Sessions Handled	Sessions Handled Previously	Users
Stage - 0	0	14	0
Stage - 1	2	14	4
Stage - 2	1	15	2
Stage - 3	0	16	0

The WiFi Router provide connectivity to mobile phones and computers. Each user is assigned with a unique VoIP ID and password which is provided by the VoIP Server. The VoIP client application is setup on the mobile phone and computer. The VoIP Server address has to be manually configured on the client applications. To initiate calls, the end user must know the assigned VoIP ID of neighbouring users to whom he needs to communicate and thus a session can be created and maintained thereafter. The VoIP Server logs the number of calls initiated, processed and completed in a particular time. The active call status and also initialization states such as media type (audio or video call) can also be monitored. Table 1.2 shows the summary of call sessions. In stage - 0, which is the idle state, the VoIP server had handled 14 sessions earlier. In the stage - 1, two parallel sessions were initiated through which four users were communicating to each other. In the stage - 2 one of the sessions was closed and the other session was active. In stage - 3 all sessions were closed. Figure 1.11 shows the screenshot of VoIP Client on Android Smartphone. A python program was developed to

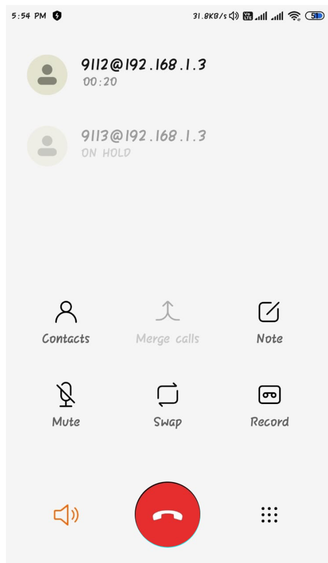
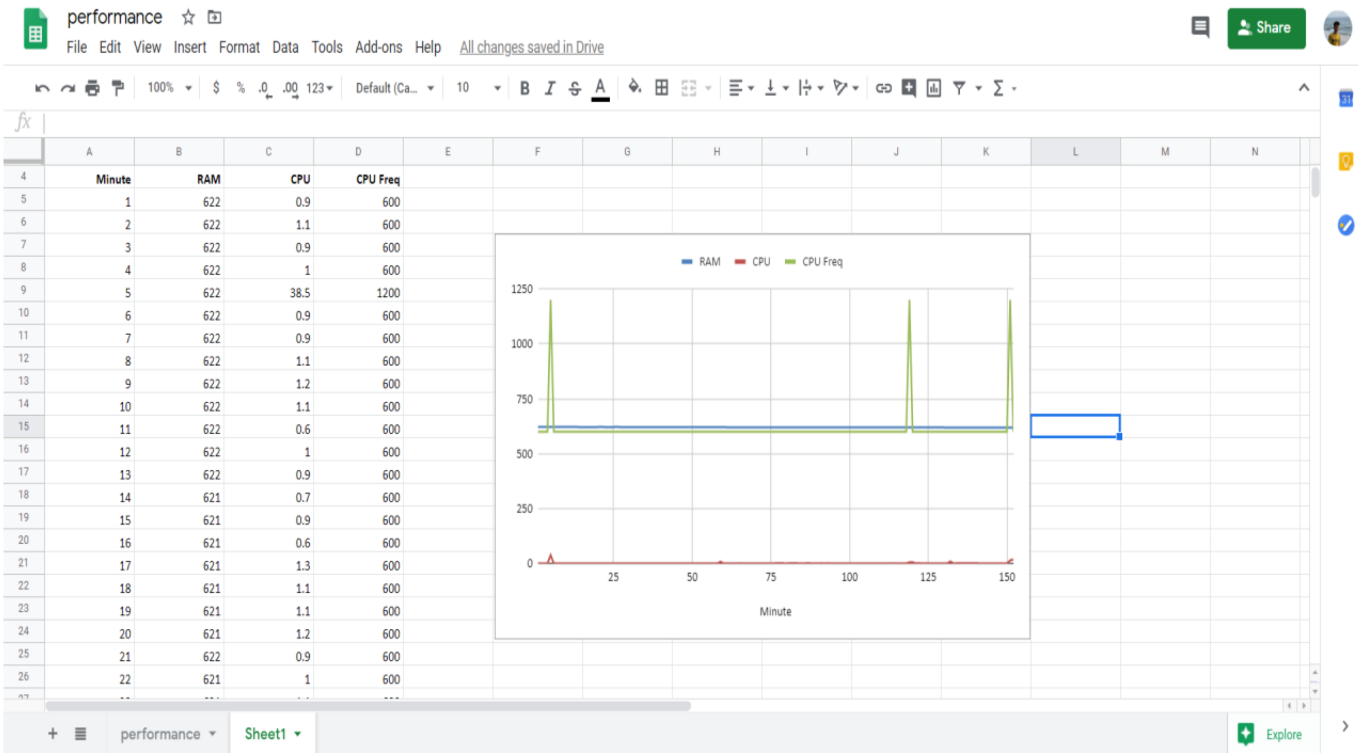


Figure 1.11: VoIP Client System in Android Smartphone

monitor the performance of the Raspberry Pi. The program captures the RAM, CPU Utilisation and CPU Frequency of the Raspberry Pi under different stages explained in Table 1. The data generated from this program is stored in a My SQL Database and later retrieved to plot the graph. The performance graph is shown in figure 1.13. The test result shows three spikes in CPU Utilisation and CPU Frequency. This spike is during the different stages. The RAM The CPU Frequency hit the maximum frequency, 1200MHz of the Raspberry Pi and the CPU Utilisation was 38.5% for Stage 1, 8.5% for



Minute	RAM	CPU	CPU Freq
1	622	0.9	600
2	622	1.1	600
3	622	0.9	600
4	622	1	600
5	622	38.5	1200
6	622	0.9	600
7	622	0.9	600
8	622	1.1	600
9	622	1.2	600
10	622	1.1	600
11	622	0.6	600
12	622	1	600
13	622	0.9	600
14	621	0.7	600
15	621	0.9	600
16	621	0.6	600
17	621	1.3	600
18	621	1.1	600
19	621	1.1	600
20	621	1.2	600
21	622	0.9	600
22	621	1	600

Figure 1.12: Data set Obtained for the CPU Load Analysis made on VoIP Server

stage 2 and 14.7% for stage 3. Rest of the time the CPU Frequency is maintained at 600MHz and the CPU Utilisation was under 1%. From the test results it is seen that the Raspberry Pi is capable of handling multiple calls and the CPU utilisation is normal.

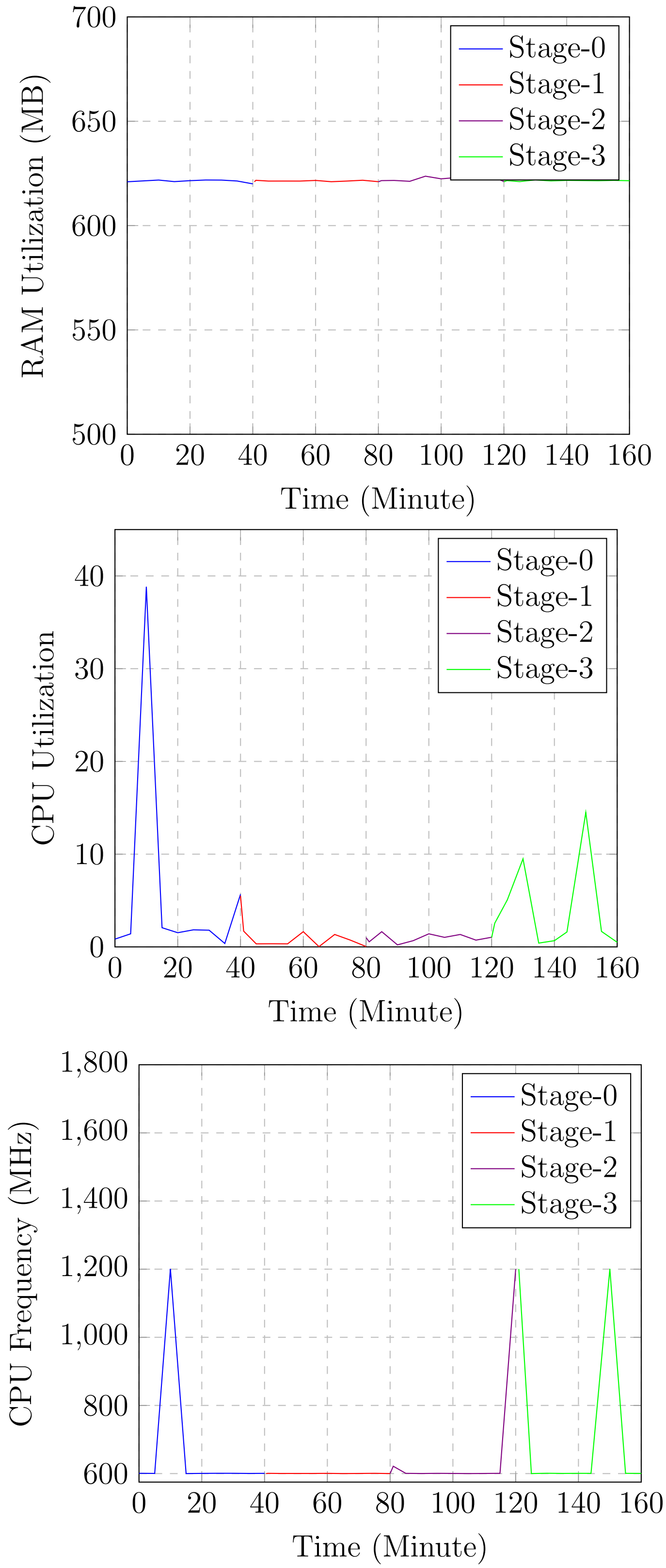


Figure 1.13: VoIP Server Performance Test Results

Chapter 2

Theory

2.1 Introduction

The legacy systems used for the Maritime communication, brief introduction on VoIP and the protocol Architecture, Open source to build the VoIP Server and the hardware tool required for developing a VoIP Server are discussed.

2.1.1 VHF amateur radio

Amateur radios use the radio frequency spectrum for the exchange of non-commercial exchange of messages which are mainly used for emergency broadcasts, experimentation purposes. These operates in the frequency range that goes from above the AM radio band 1.6 MHz to just above the citizens band 27 MHz. Ham Radios which are operating at these radio frequencies are affected due to weather changes and terrain conditions, these require a power source for its working, and these are not user-friendly. Considering these drawbacks of using Ham Radios, the main motto is to provide a cost-effective mode of full duplex communication for the marine fisherman which can overcome the drawbacks of using Ham Radios for communication, there comes the necessity of using

VoIP services.

2.1.2 Raspberry pi

A Raspberry Pi is an inexpensive, small, portable computer that functions like a desktop computer. The main features of Raspberry Pi 3 are its processor is a quad-core 1.2GHz Broadcom BCM2837 64bit CPU with 1GB RAM. In Raspberry Pi 3, there are BCM43438 wireless LAN and Bluetooth Low Energy on board. It has 40pin extended GPIO and 4 USB 2.0 ports. Raspberry Pi 3 has a micro SD port where the micro SD card with the operating system is plugged in. Raspberry Pi is an open source hardware, with Broadcom SoC (System on a Chip), which runs a considerable lot of the principal parts of the board– CPU, designs, memory, the USB controller, and so on.

2.1.3 Asterisk

Asterisk is an open source framework that can make a computer, such as a Raspberry Pi, into a communication server. This allows you to build your own business PBX phone system. Asterisk uses FreePBX as a Graphical User Interface (GUI) that controls and manages Asterisk where you can configure extensions, users etc. Asterisk is a software implementation of a private branch exchange. Asterisk is used to establish and control telephone calls between telecommunication end points and services on Voice over Internet Protocol. The name Asterisk came from the symbol “*” which is used to represent a signal used in Dual Tone Multi Frequency (DTMF) which are widely used for telecommunication signaling between telephone handsets and switching centers over analog telephone lines in the voice-frequency bands.

2.1.4 VoIP

Voice over Internet Protocol (VoIP) is a method for transmitting voice communication and multimedia sessions over the internet. Asterisk is an open software which consists of Private Branch exchange (PBX). PBX is a private telephone network used within a company or within an organization. VoIP supports Session Initiation Protocol (SIP), an application layer protocol which creates, maintains and terminates session between two VoIP systems. People travel a long distance in the seas especially fishermen and have possibilities of staying in their ships. In these times, they would not be able to communicate with their families. Initially amateur radios were used which operates using different radio frequencies based on applications and geographical locations. Using these ham radios people were able to communicate. But its operation can be affected due to weather and climatic changes. Also it requires skilled operators. So as an alternative to this came IP phones or softphones which are cheaper and more efficient than the former.

2.1.5 SIP protocol

The acronym SIP stands for Session Initiation Protocol. It is an application layer protocol that works in conjunction with other application layer protocols to control multimedia communication sessions over the Internet. SIP is often used in Voice-over-IP telephony to establish the connection for telephone calls. The Users or subscribers who can be able to send or receive data through this protocol are referred as SIP-USERS. This is a standardized protocol which establishes either of TCP or UDP connections between the END Users for the data transfer. This exchange of data between the end users is called a session.

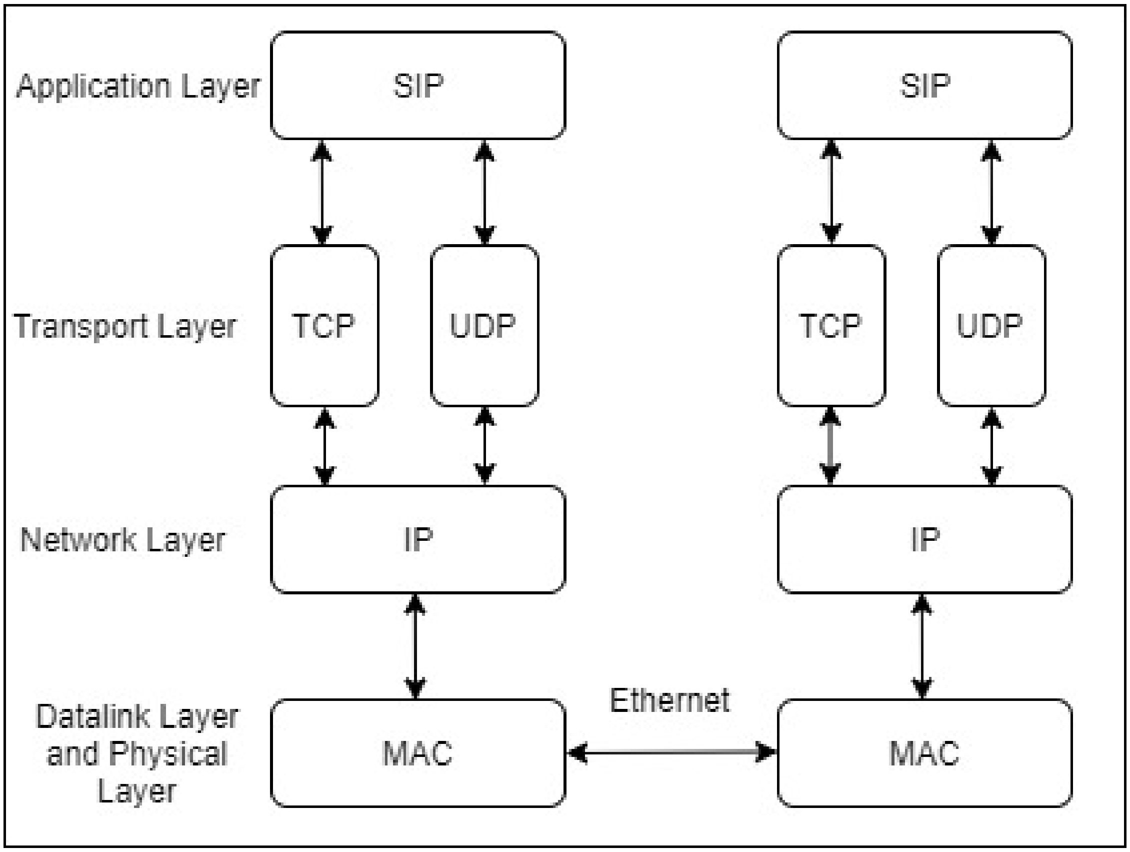


Figure 2.1: SIP Protocol Architecture

2.2 Different types of VoIP Clients systems available

To make a VoIP Call, an external client support was needed which can access the VoIP Server configured on Raspberry pi.There are three different VoIP Clients Systems which mentioned in the can be used for performing a VoIP call making use of the database created on a VoIP Server.The three different client systems are:

2.2.1 Android based SIP Clients

Many of the current versions of Android comes with in-built SIP Protocol support through which we can make VoIP Calls similar to how we can make calls through a network operators network such as JIO,Airtel etc.The Step wise approach of how a SIP User can be registered and configured using the support of an Android Operating system can be seen from the following figures.

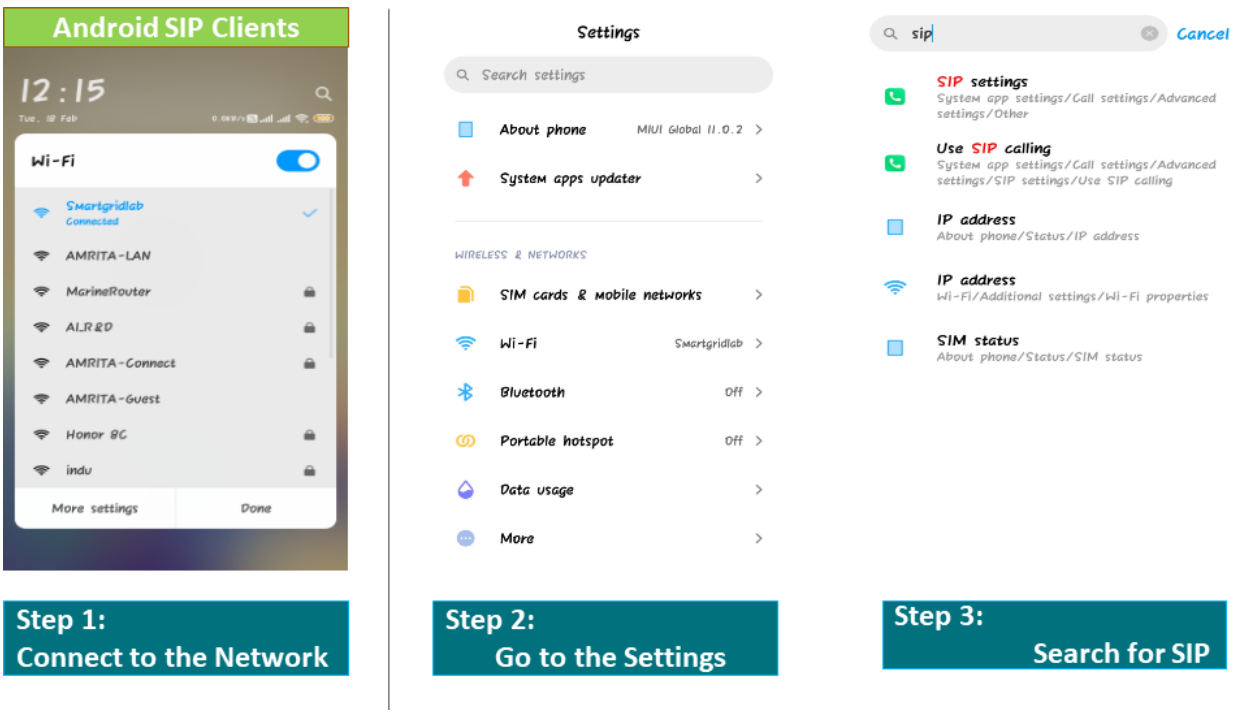


Figure 2.2: Step wise approach of Registering a SIP User using Android OS:1

It is observed that to make use of VoIP Server Database, we need to connect the client system to the network in which the VoIP Server is deployed.

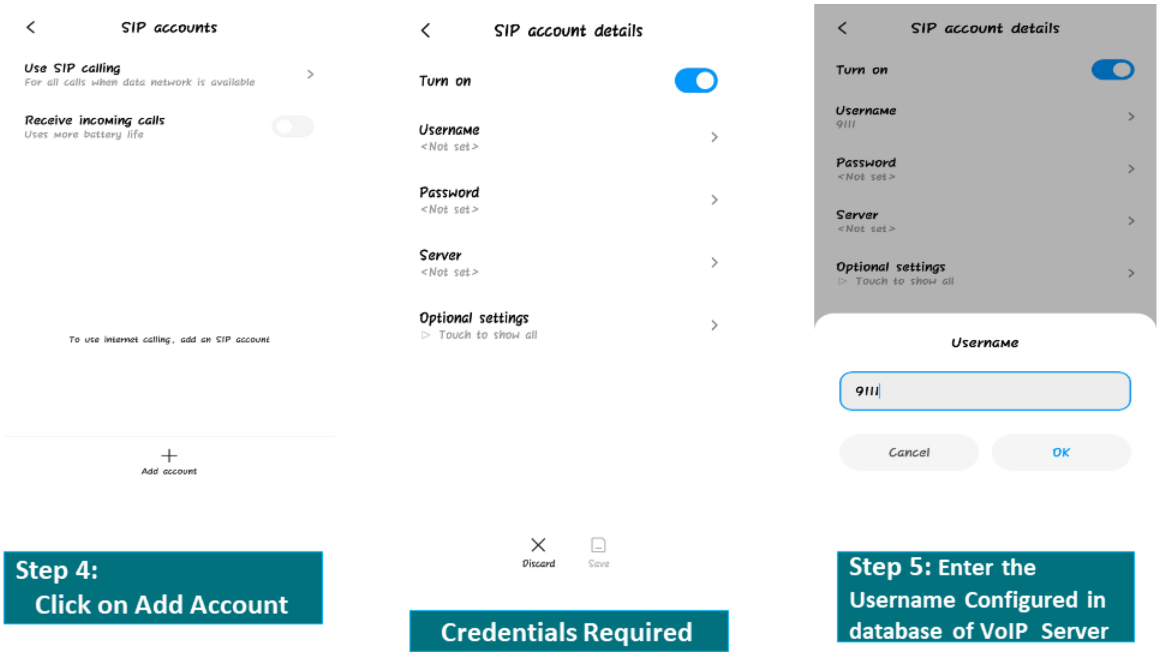


Figure 2.3: Step wise approach of Registering a SIP User on Android OS:2

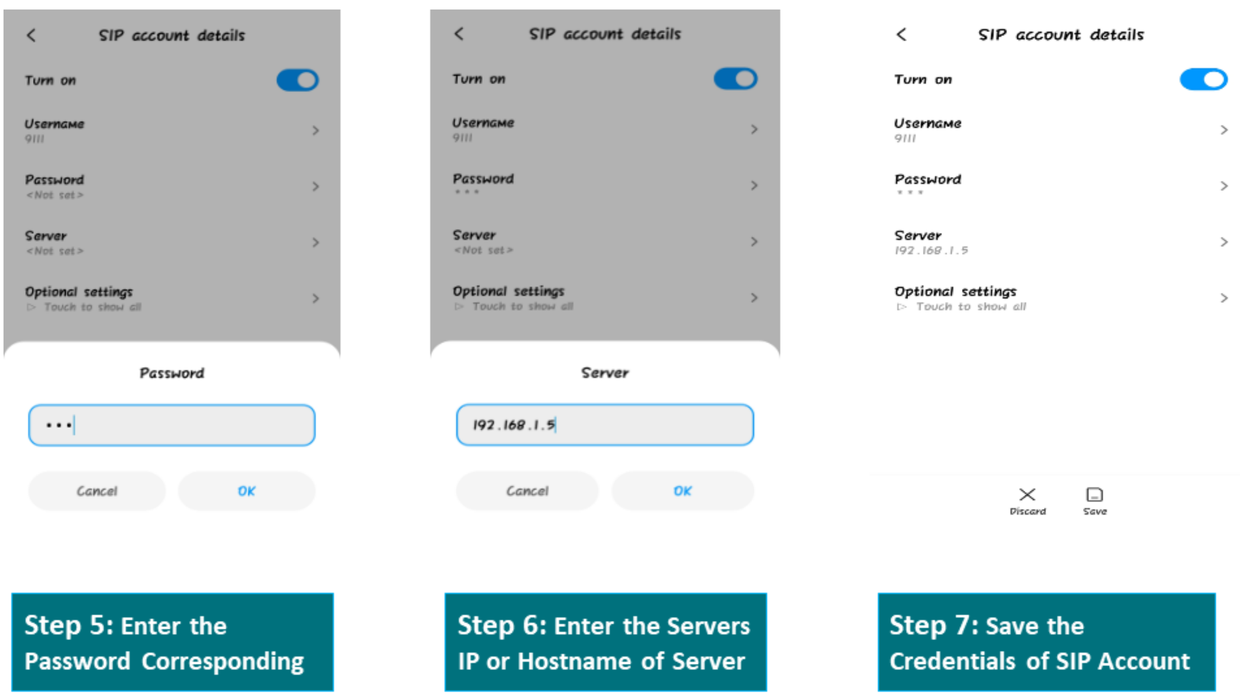


Figure 2.4: Step wise approach of Registering a SIP User on Android OS:3

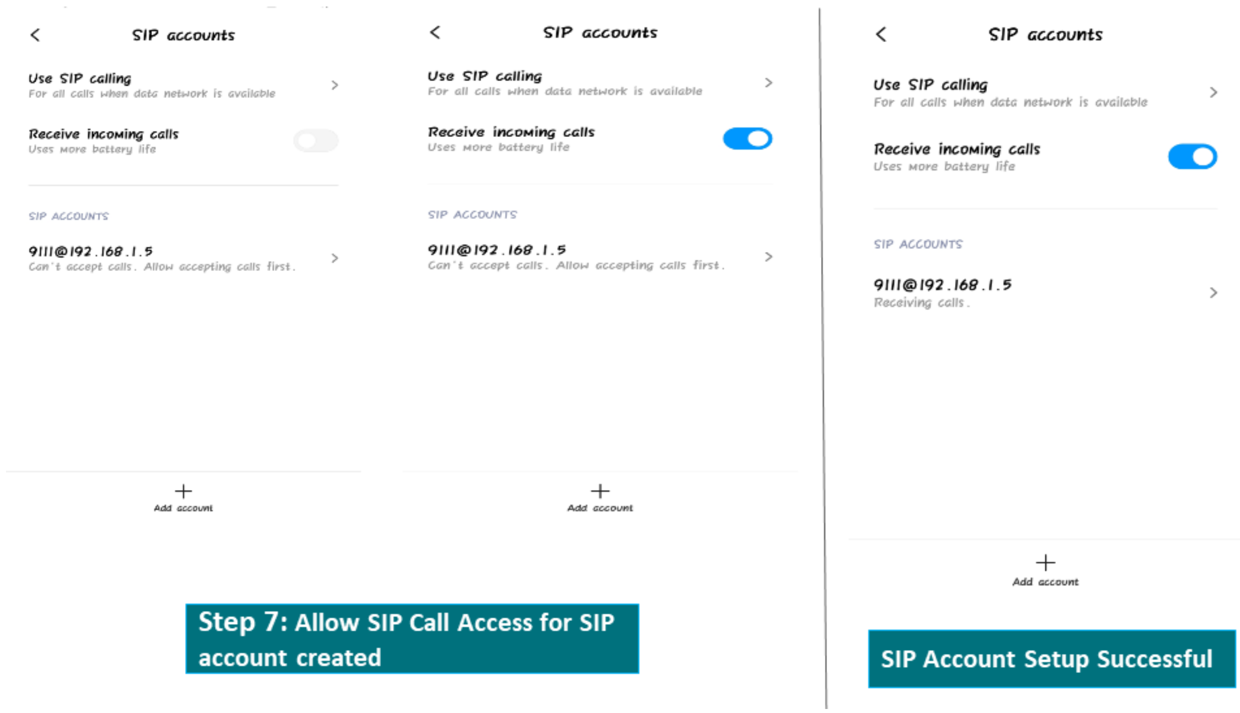


Figure 2.5: Step wise approach of Registering a SIP User on Android OS:4

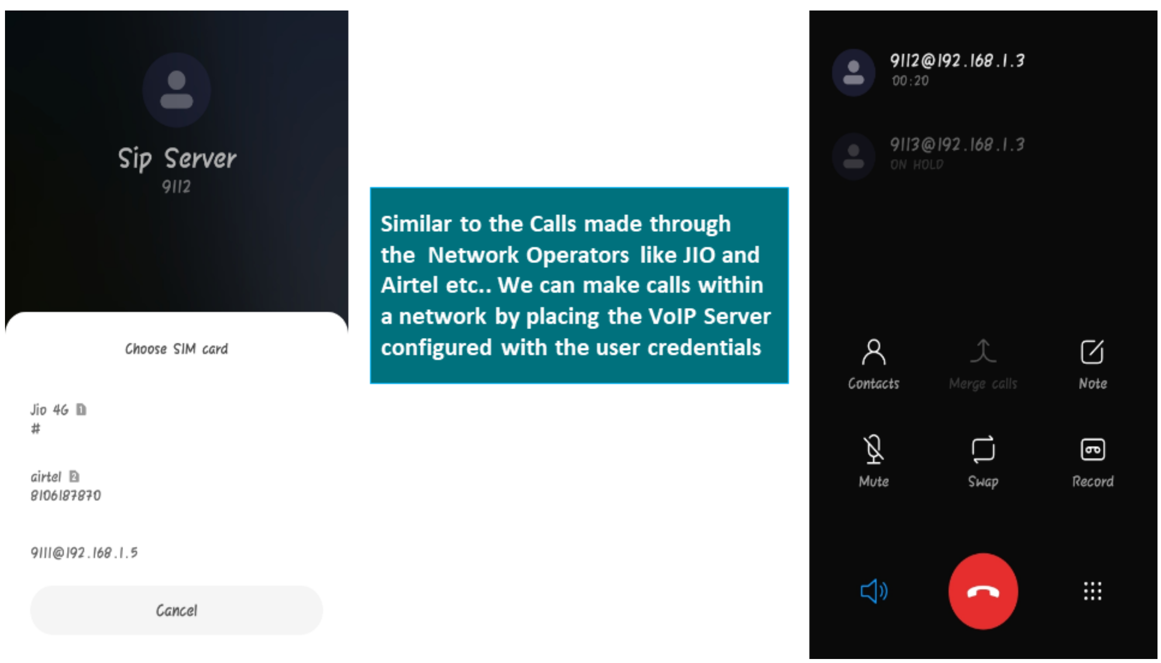


Figure 2.6: Step wise approach of Registering a SIP User on Android OS:5

2.2.2 Third Party Soft-phone Applications as SIP Clients

In addition to the SIP Protocol support in the Android OS. We can even use of some of the soft-phone applications such as Zoiper,Ozeki etc..The steps involved in registering a SIP-User in Soft-Phone Application Zoiper is shown in the following figures. The Server

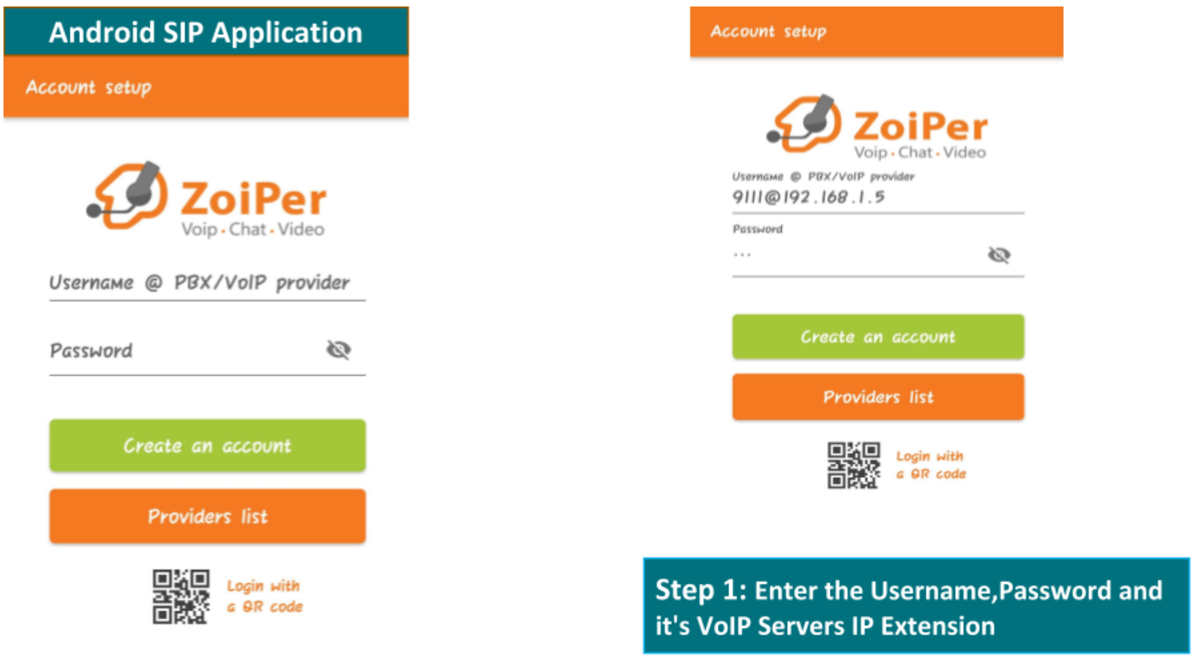


Figure 2.7: Step wise approach of Registering a SIP User on Zoiper:1

extension can be either IP Address of the VoIP Server or the IP resolved as the host-name for the server.In the above scenario it is observed that the IP Address of the VoIP Server is "192.168.1.5" which is further resolved to "voipbeyondthebounds.com". Considering the example of Gmail Server with the mail address "nharshith27@gmail.com" in its database where the "nharshith27" corresponds to the username of the user in the gmail server database,where as the extension gmail.com corresponds to the hostname of the gmail server which has its own unique IP.Comparing the Gmail Server with the VoIP Server developed,it is observed from Figure 2.7 that the username is ”9111” and the extension ”192.168.1.5” which is the IP Address of VoIP Server.The IP Address of the VoIP Server is resolved to a hostname ”voipbeyondthebounds.com”.

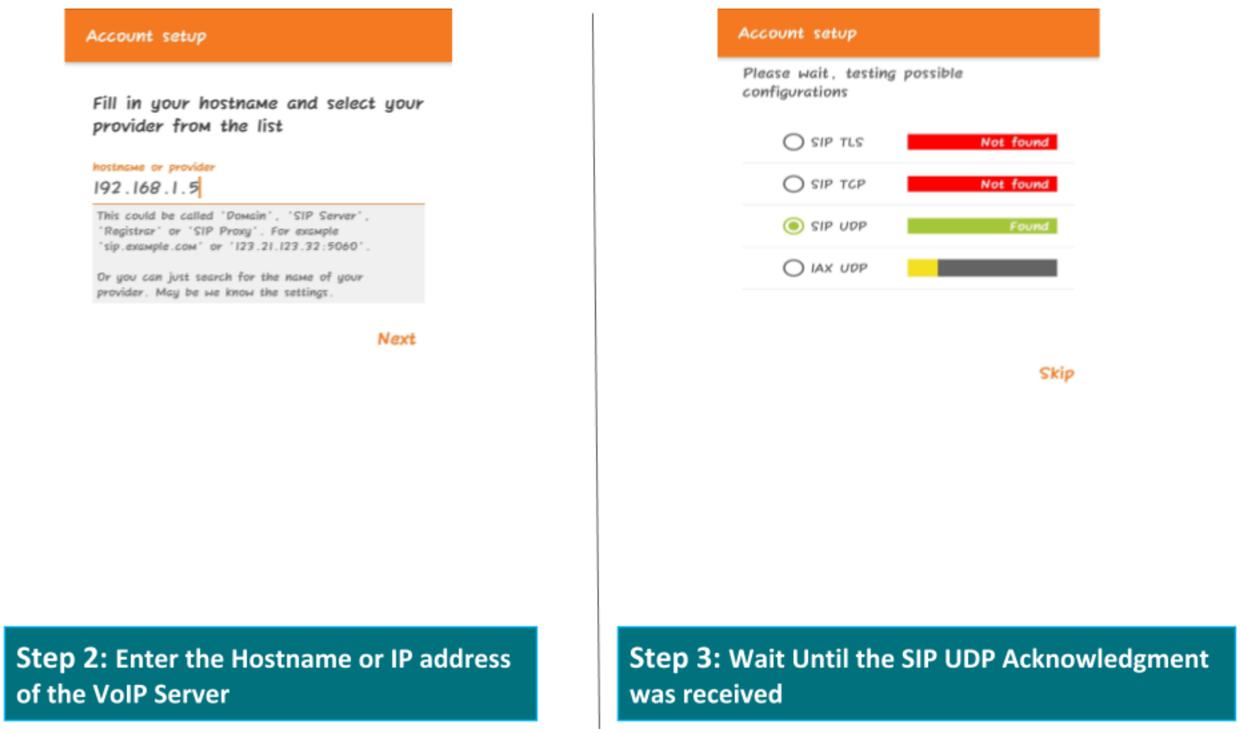


Figure 2.8: Step wise approach of Registering a SIP User on Zoiper:2

The call logs of the VoIP Server and the statistics of the call made can be observed from the Figure 2.9.

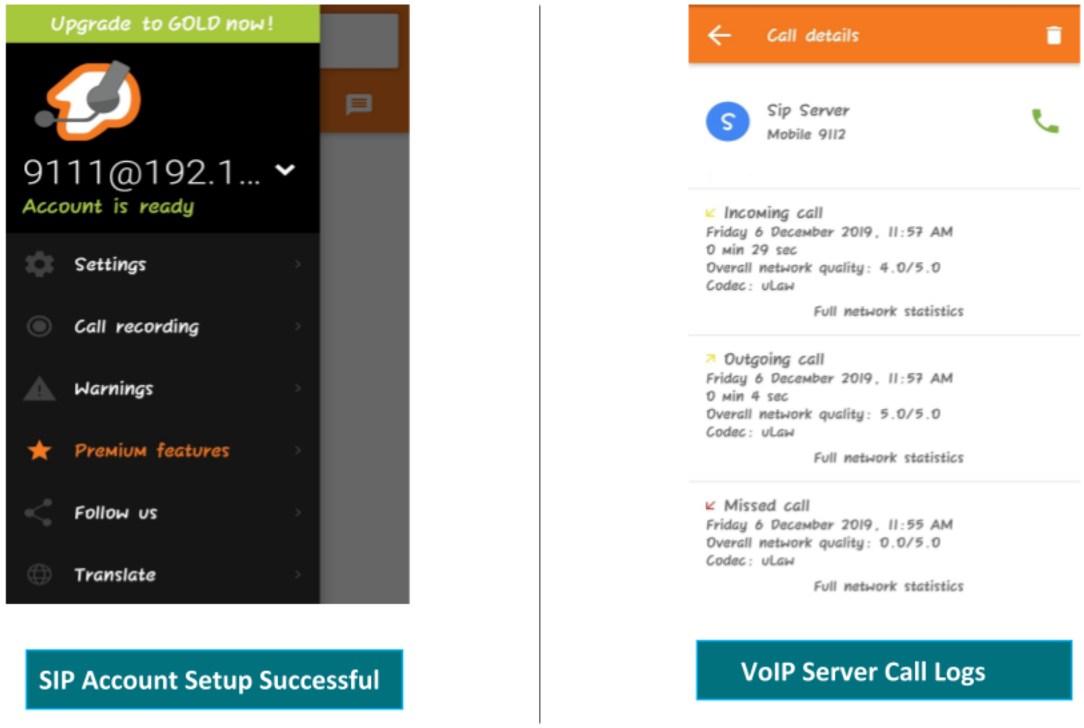


Figure 2.9: Step wise approach of Registering a SIP User on Zoiper:3

2.2.3 PC-Based SIP Client systems

There are some soft-phone applications such as Ozeki through which we can make VoIP calls.These soft-phone application are helpful in making calls through PC which comes up with Windows OS,Ubuntu OS etc.. Ozeki SDK is used to make VoIP Calls in a PC by following a similar approach discussed in previous sections i.e by registering the SIP User with the credentials configured on the VoIP Server database.The Step by step call log changes while initiating the call can be observed from the Figure 2.10 and Figure 2.11.

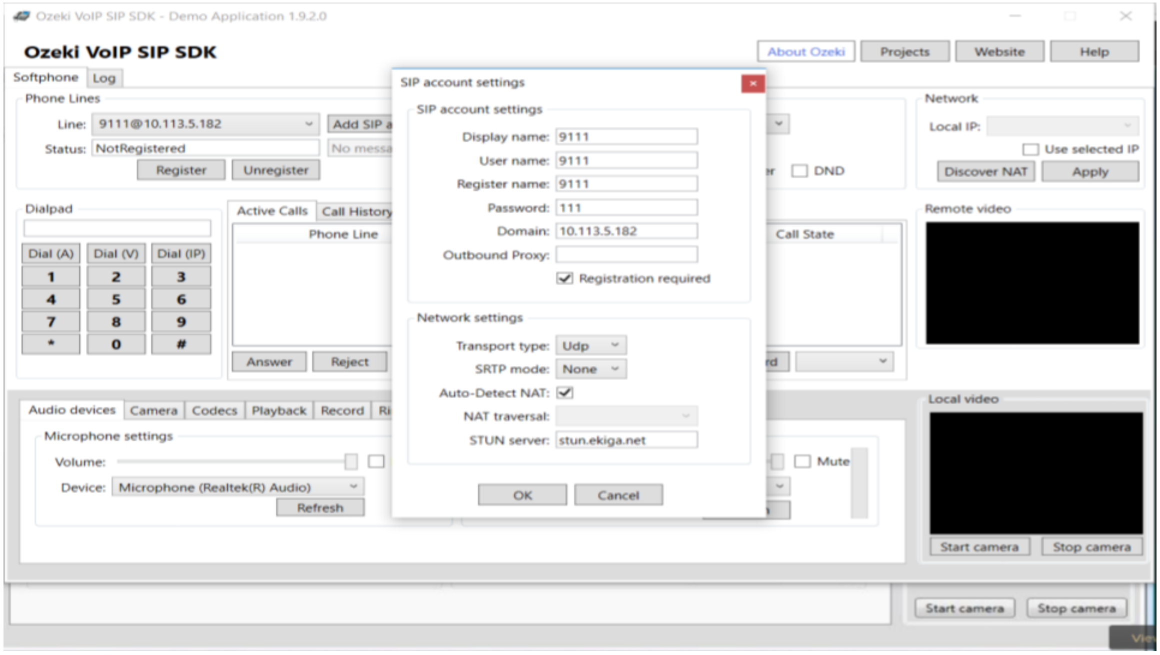


Figure 2.10: Step wise approach of Registering a SIP User on PC:1

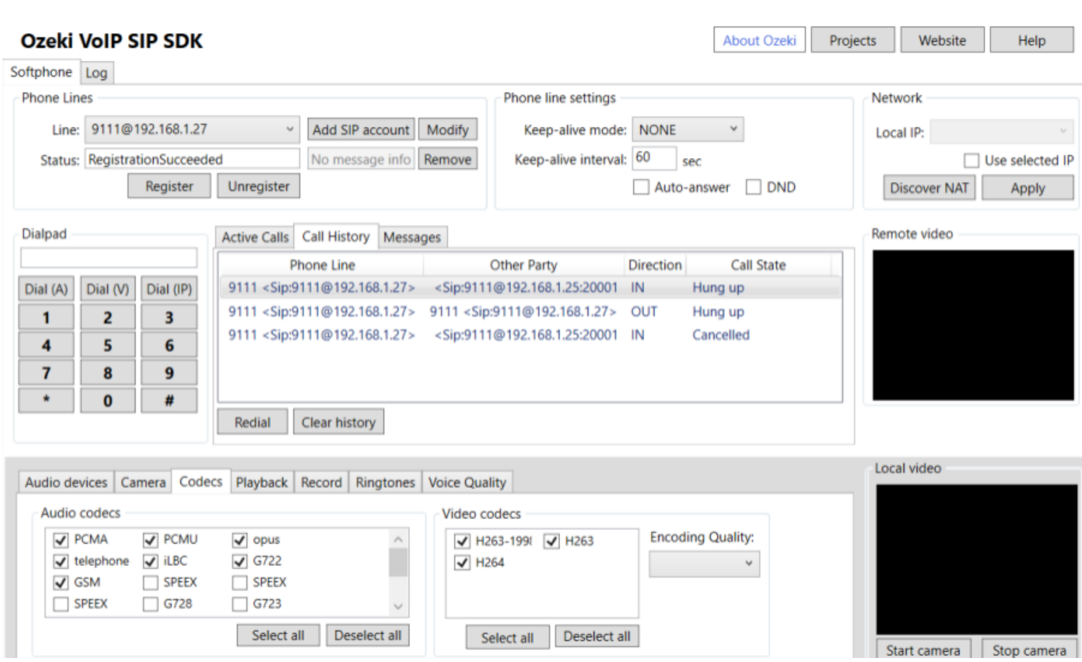


Figure 2.11: Step wise approach of Registering a SIP User on PC:2

2.3 Network Topology

The Network Topology developed in the Cisco Packet traces shows all the different scenarios in which the fishermen are benefited with the deployment of VoIP Server on OceanNet Backhaul Network.The OceanNet Backhaul network can be further extended to Fisherman Community Network through which an IP Phone can be used as observed in the Figure 2.12.

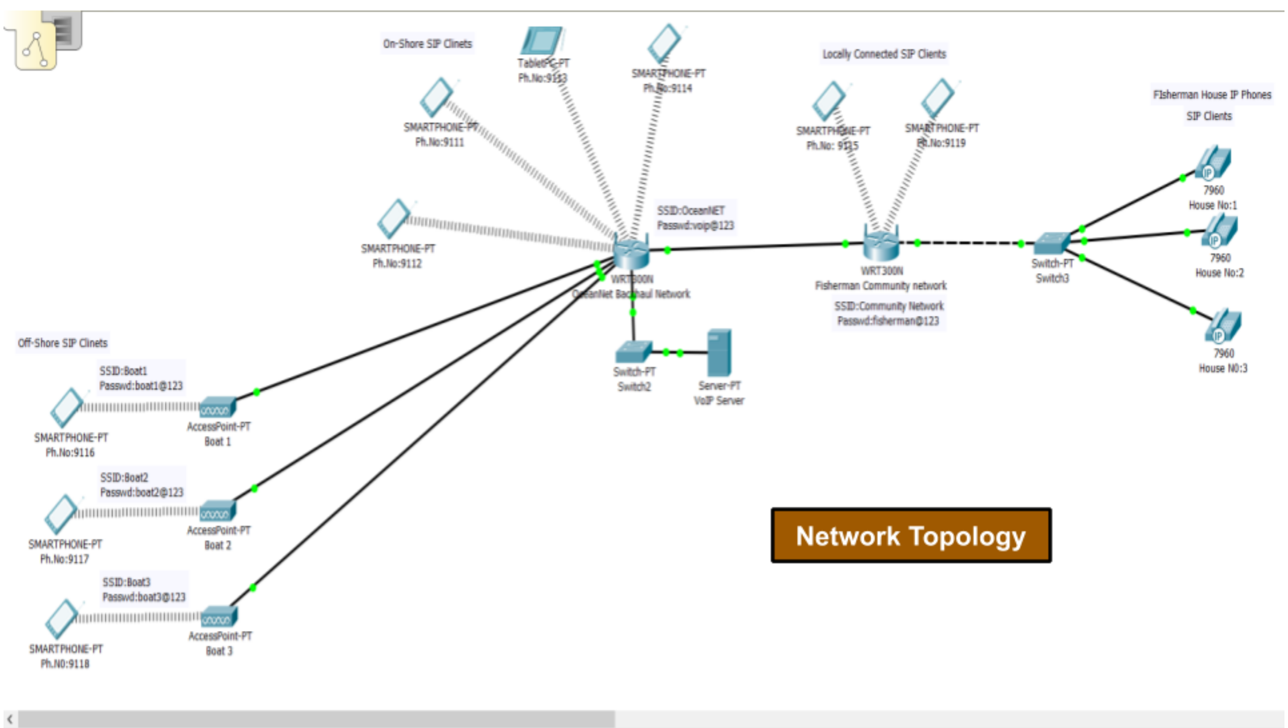


Figure 2.12: Network Topology

2.4 Network Flow of a SIP Call

The step by step packet transmission between the two SIP Users can be observed in the wireshark.Through the observation data generated in the wireshark the call flow of the data set can be observed from Figure 2.13.

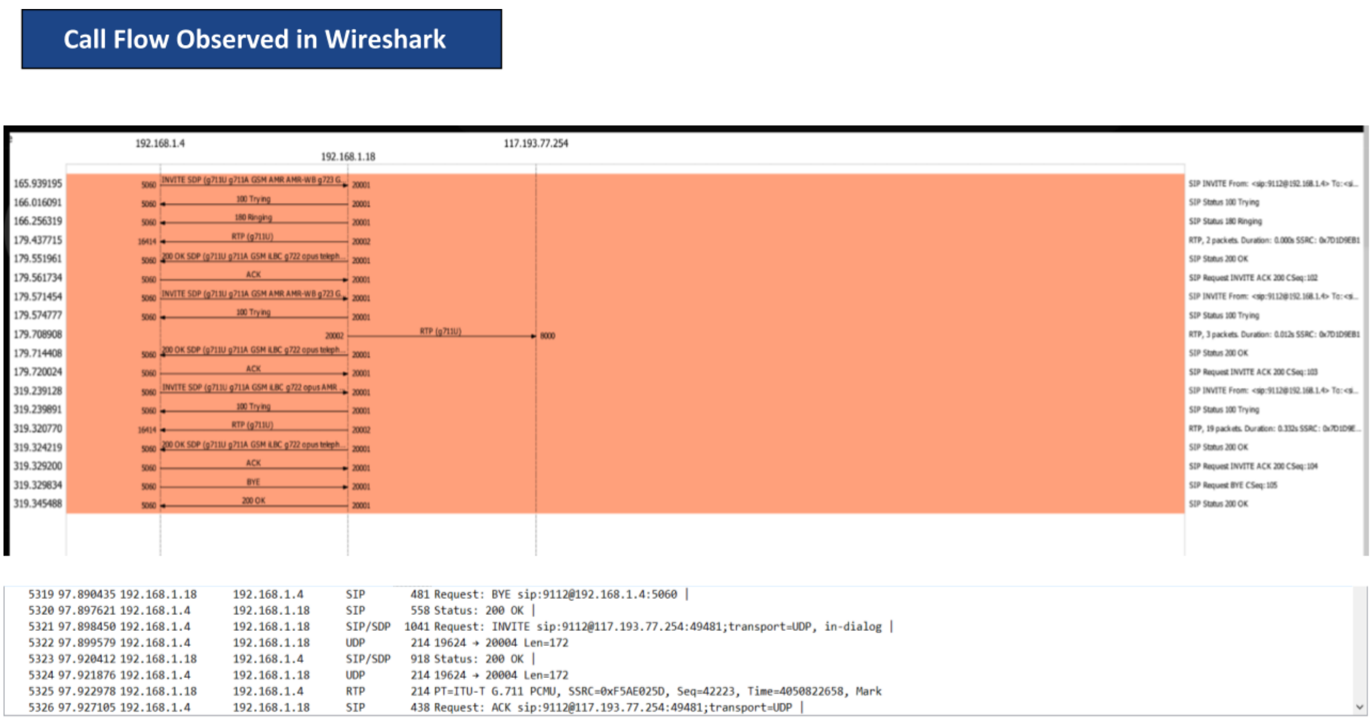


Figure 2.13: Network Flow of a SIP Call

Chapter 3

Summary, conclusions and scope for further research

Communication services provided on land are termed as onshore and cellular network is an example. While the latter provides services in ocean and air. Long range communications such as VHF, satellite links are examples. Same network scenario applies for both types of communication but unfortunately the cellular network providers limit their connectivity to 10 nautical miles for offshore communication due to OpEx and CapEx where the revenue they get from each user is much less in offshore communication. We know that we humans earn for our livelihood in different ways. Among the human species there is one category whose livelihood relies on fishing and they are Fishermen, who travel for very far distances into the sea in search of good Fishing zones and even stay for a few weeks having lost their contacts with their families. Currently the marine fishermen are using some of the legacy systems such as Marine VHF Radios and satellite Radios for communication which comes up with various disadvantages. OceanNet is a solution to provide low cost reliable internet service for marine applications. This system covers around 100 km in the ocean. Fishermen use this service to browse the internet and stream videos on their smart devices. Apart from this, the system also helps the boat owners and families to track the fisherman in real time.The Proposed solution provides a voice over the OceanNet Service, using which the fisherman can communicate within the network of boats and to their families. Apart from day to day communications, this voice service can be used by government officials to provide warnings during natural calamities and provide a safe route to return home.

3.1 Conclusion

A VoIP server was configured on the Raspberry Pi to enable a low cost marine communication system for the fisherman. The VoIP Service uses the commercially available OceanNet network to provide communication services. The system is validated on a laboratory setup and proved to be efficient and reliable.From the Performance evaluation done on the server developed it is known that the Raspberry pi as a VoIP Server is stable while maintaining the multiple calls and the call latency was observed to less and the quality of the VoIP call made is good.Through this results it is known the VoIP Server developed was more stable and can be used for the Maritime Communication considering the disadvantages of the legacy Systems used for Offshore communications in sea.

3.2 Scope of Future Research

With the established VoIP communication supported by the OceanNet System we can further increase the number of services mentioned below and can make the Maritime Communication stable.

1) Low cost Wireless Access Point: Through which the fisherman can get the connectivity offshore with the help of Nano Station on the boat and can be used for extending the network Range.

2) Voice and Video Calls from Edge Server: A VoIP Server which is integrated with the OceanNet base station on the shore can maintain the Voice and Video calls through which the fisherman can communicate with their families through the connectivity provided by the Wireless Access point on the boats.

3) GPS Geo-Location and Live tracking: This Emergency service is used by the fisherman to get track their position in the deep sea and through this the fisherman can get notified if they cross the fishing zone or even the country borders with the help of GPS Geo-Location.

4) Accident detection in boats: During the night times the fisherman use to anchor their boats,many of the bigger boats which are coming by that way hits the small boats which are anchored.This Emergency service which uses the IMU Sensor readings on the boat whenever the readings go beyond the threshold an accident alert will be sent to the fisherman with the help of Buzzer and also turns on the LED Setup on the boat such that other boats which are in and around its vicinity can be notified.

5) Sea State Alerts-The current state about the sea Level can be known to the fisherman with this service and the imbalances in the boat moment can also be notified.

6) Online Fish market- The VoIP Edge Server is configured with a special Business Extension link similar to a customer care line which can be only accessed to the customer and the fisherman.Through this service fisherman can have deal with their customers on the shore even when they are in the ocean.

7) Ocean weather and natural calamities warnings-In addition to the Sea state alerts the fisherman can be able to know the weather conditions and warnings about the Natural Calamities can be notified to the fisherman by the government officials as the edge server proposed will provides the link between the offshore and onshore communication.

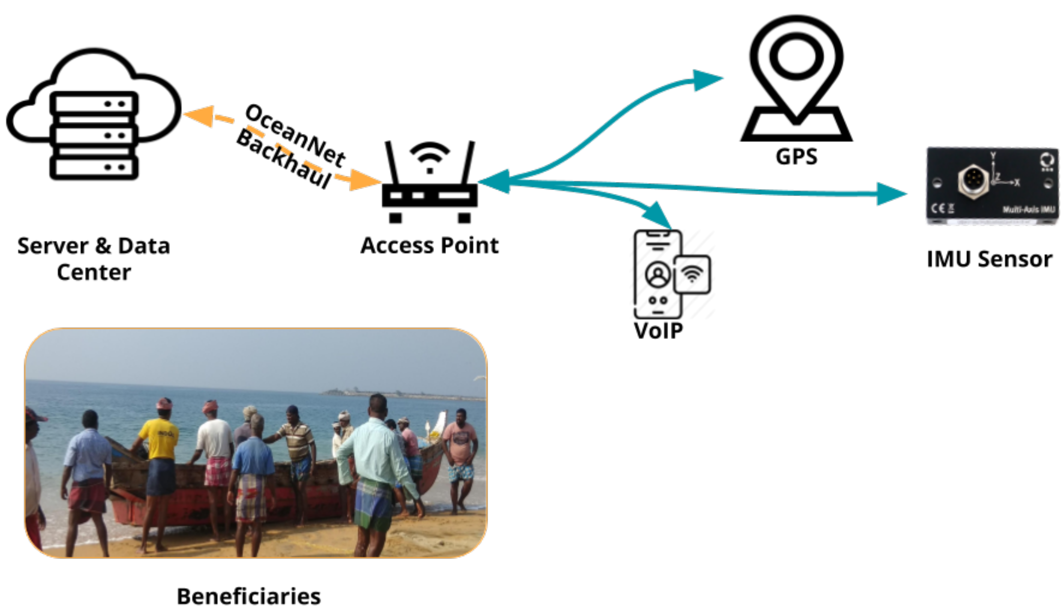


Figure 3.1: Architecture for Future Research

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Publications

1. Harshith Nadella, Charitha Nagamalla, Sai Shibu N B, Sethuraman N Rao,Arjun D "Low Cost VoIP Service for Marine Fishermen: Development and Performance Evaluation", 2020 International Conference on Wireless Communications Signal Processing and Networking, Chennai. [Accepted]

Extras

1. Participated in 3-Minute Project Thesis competition conducted by IIST, Trivandrum and qualified for the first round.

2. Participated in IICDC Contest organised by DST and Texas instruments. Got shortlisted for semifinal round.

Low Cost VoIP Service for Marine Fishermen Development and Performance Evaluation

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***Abstract*—Fishermen travel far up to 50 nautical miles into the ocean for fishing. They are often stressed and worried about their families as they cannot contact them for many days. Also, it is very difficult to pass any information regarding cyclone or other natural calamities to the fishermen. As there is no scope for mobile communication offshore beyond 8-9 nautical miles due to high operational expenses(OPeX). Traditional communication systems like Marine VHF radios and satellite phones are costly and restricted to only the boats in the ocean. This paper proposes a cost-effective, full duplex communication system for the marine fishermen to communicate among the boats in the ocean as well as with their families on the shore. This paper also presents the performance evaluation results of the system setup in a lab environment. The results obtained from the tests are satisfactory and the system can be implemented on the boats.**

***Index Terms*—Raspberry Pi; Asterisk; SIP user; VoIP server; VoIP Client, Marine Communication.**

I. Introduction

The transition in the way of communication between the users began in the early 19th century. The Radio technology was still in its infancy in 1912, it was limited in the Morse code for transmissions. Most of the radio transmitters at the time were called "spark" transmitters which depended on electrical energy sparks for transmitting signals. This form of transmitter could not continuously emit radio signals, making voice messages virtually impossible. Spark communications also covered a large bandwidth, making interference from outside messages inevitable. As a result, radio was considered an unreliable form of emergency communication, despite being marketed as maritime technology. The transition continued and after the radio waves were adapted into a communication system in the late 19th century, Marine VHF Radio also called as VHF Amateur Radio was invented. The use of this Amateur Radio was started by the marine fisherman for the communication among the nearby boats. According to the article [1], the Titanic sinking had some inevitably large effects on radio and broadcasting. Only four months after Royal Mail Ship (RMS) Titanic was lost, American government passed the Radio Act of 1912 to gain control over the airwaves and all operators to hold a valid federal license to use radio equipment and it also restricted amateur users to bands less than 200 meter wavelengths far below the official maritime communications would be conducted, which reduced the chances of interference with transmissions.This way of communication is not efficient for long range communications. OceanNet, developed by our research center, is a commercially available internet service for the marine fishermen, across the Arabian sea. The OceanNet system uses proprietary long range WiFi technology to provide internet service in the ocean for upto 100km. Long Range WiFi Base Stations are deployed along the shores and the boats subscribing to this service are provided with a long range consumer premises equipment and control box. Fisherman can connect their mobile phones to the WiFi on this system to connect to the internet when they are in the ocean. This paper presents an efficient, cost-effective way of communication for marine fisherman through Voice over Internet Protocol (VoIP) technology through the use of long range WiFi [2], [3] and [4] services. The OceanNet system consits of a control system to automate the antenna orientation and track the boats in real time using GPS. The control system is developed using Raspberry Pi [5]

A fisherman community in the Azheekal village, near Kollam, Kerala, India was considered to understand the current system being used by the fisherman. A survey was conducted to understand the issues faced by them and their requirements for a better system. Figure 2 shows the VHF Amateur Radio system used on the boat. The Amateur radios use the spectrum of radio frequencies to exchange non-commercial messages that are used primarily for emergency broadcasts and experimentation purposes. VHF Amateur radios operate within frequency range from 156MHz-172MHz [6]. Marine VHF Radios operating at these radio frequencies are affected by changes in weather and terrain conditions, which require a power source for their operations and are not user-friendly. On account of these drawbacks of using marine VHF radios, there is a need to use VoIP service to provide a cost-effective mode of full duplex communication for the marine fisherman that can solve the disadvantages of using VHF Amateur Radios for communication. The paper is organized as follows: section 2 describes the state of the art and similar work performed by others, section 3 describes the proposed system, section 4 shows the performance results and the paper is concluded in section 5.

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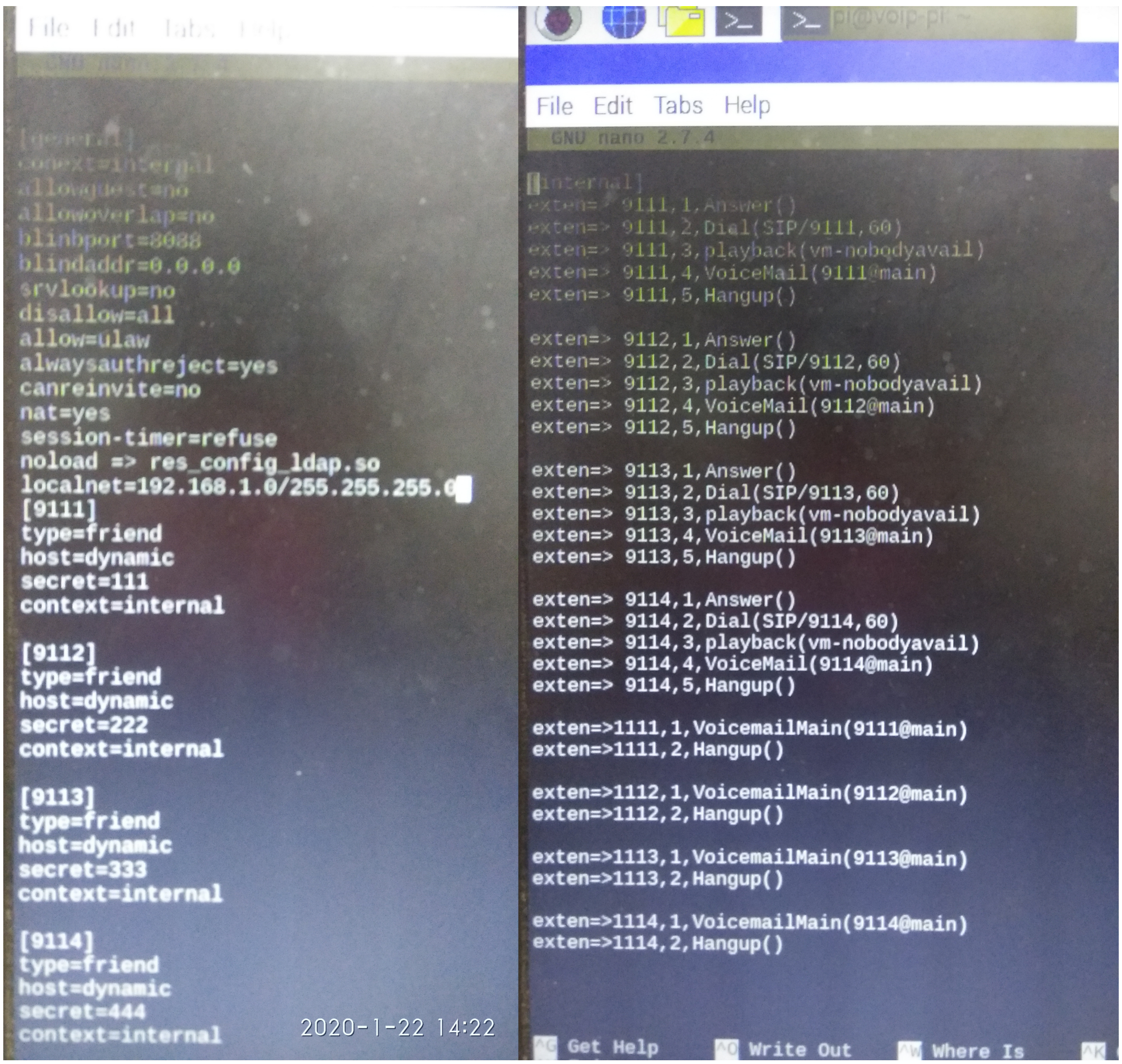
Appendix A



File Edit Tabs Help
GNU nano 2.7.4
[general]
conext=internal
allowguest=no
allowoverlap=no
blinbport=8088
blindaddr=0.0.0.0
srvlookup=no
disallow=all
allow=ulaw
alwaysauthreject=yes
canreinvite=no
nat=yes
session-timer=refuse
noload => res_config_ldap.so
localnet=192.168.1.0/255.255.255.0
[9111]
type=friend
host=dynamic
secret=111
context=internal
[9112]
type=friend
host=dynamic
secret=222
context=internal
[9113]
type=friend
host=dynamic
secret=333
context=internal
[9114]
type=friend
host=dynamic
secret=444
context=internal
2020-1-22 14:22
pi@voip-pi: ~
File Edit Tabs Help
GNU nano 2.7.4
[internal]
exten=> 9111,1,Answer()
exten=> 9111,2,Dial(SIP/9111,60)
exten=> 9111,3,playback(vm-nobodyavail)
exten=> 9111,4,VoiceMail(9111@main)
exten=> 9111,5,Hangup()
exten=> 9112,1,Answer()
exten=> 9112,2,Dial(SIP/9112,60)
exten=> 9112,3,playback(vm-nobodyavail)
exten=> 9112,4,VoiceMail(9112@main)
exten=> 9112,5,Hangup()
exten=> 9113,1,Answer()
exten=> 9113,2,Dial(SIP/9113,60)
exten=> 9113,3,playback(vm-nobodyavail)
exten=> 9113,4,VoiceMail(9113@main)
exten=> 9113,5,Hangup()
exten=> 9114,1,Answer()
exten=> 9114,2,Dial(SIP/9114,60)
exten=> 9114,3,playback(vm-nobodyavail)
exten=> 9114,4,VoiceMail(9114@main)
exten=> 9114,5,Hangup()
exten=>1111,1,VoicemailMain(9111@main)
exten=>1111,2,Hangup()
exten=>1112,1,VoicemailMain(9112@main)
exten=>1112,2,Hangup()
exten=>1113,1,VoicemailMain(9113@main)
exten=>1113,2,Hangup()
exten=>1114,1,VoicemailMain(9114@main)
exten=>1114,2,Hangup()
^G Get Help
^O Write Out
^W Where Is

VOIP Server Configurations

1) sudo su

2) apt – get install asterisk **For Asterisk Installation**

Asterisk: Asterisk is a software implementation of a private branch exchange.Asterisk is used to establish and control telephone calls between telecommunication end points and services on Voice over Internet Protocol. The name Asterisk came from the symbol "*" which is used to represent a signal used in Dual tone multi frequency which are widely used for telecommunication signalling between telephone handsets and switching centers over analog telephone lines in the voice-frequency bands.

---

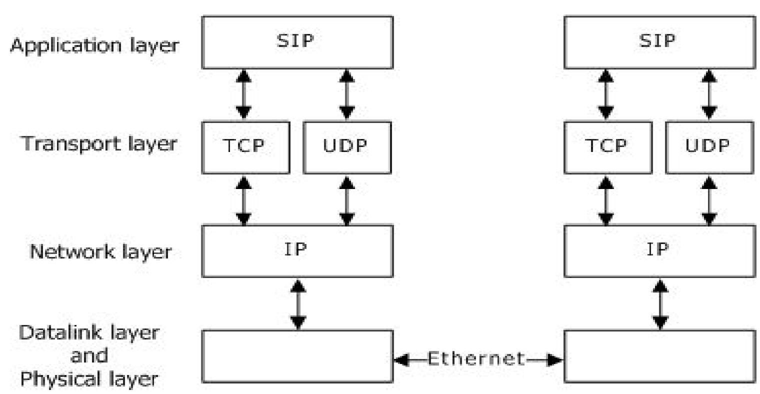
3) asterisk –r

4) mv /etc/asterisk/sip.conf /etc/asterisk/sip.conf.orig

For creating a file in which we can give the username, password and categorize dynamically accessed SIP users

5) nano /etc/asterisk/sip.conf

SIP Protocol: The acronym SIP stands for Session Initiation Protocol. It is an application layer protocol that works in conjunction with other application layer protocols to control multimedia communication sessions over the Internet. SIP is often used in Voice-over-IP telephony to establish the connection for telephone calls. The Users or subscribers who are able to send or receive data through this protocol are referred as **SIP-USERS.** This is a standardized protocol which establishes either of TCP or UDP connections between the END Users for the data transfer.This exchange of data between the end users is called a session.



---

6) The SIP users can be Configured as follows:

<u>Function 1</u>: Configuring SIP Users

[general]

Context=internals

allowguest=no

allowoverlap=no

bindport=8088

bindaddr=0.0.0.0

srvlookup=no

disallow=all

allow=ulaw **To Use ulaw codec (coder-Decoder)**

alwaysauthreject=yes

canreinvite=no

For Enabling Network Address Translation to increase the interconnection between as many as SIP users

nat=yes

session-timers=refuse

localnet=192.168.1.0/255.255.255.0

IP Address of the Network through which Raspberry Pi is connected with subnet Mask.

- **SIP user1 which can be dynamically accessed by the**

Username: 9111

Password: 111

[9111]

type=friend **For Categorizing the sip users**

host=dynamic

secret=111

context=internal

Further configurations can be done in INTERNAL Function.

- **SIP user2 which can be dynamically accessed by the**

Username: 9112

Password: 222

```
[9112]
   type=friend
   host=dynamic
   secret=222
   context=internal
```

- **SIP user3 which can be dynamically accessed by the**

Username: 9113

Password: 333

```
[9113]
  type = friend
  host = dynamic
  secret= 333
 context=internal
```

- **SIP user4 which can be dynamically accessed by the**

Username: 9114

Password: 444

```
[9114]
 type =friend
 host = dynamic
 secret= 444
 context=internal
```

Similarly we can configure as many as sip users needed.

*** 9111 ,9112,9113,9114 are the sip users which can be accessed dynamically with the secrets 111, 222, 333, 444 respectively.**

7) mv /etc/asterisk/extensions.conf   /etc/asterisk/extensions.conf.or

To Create a file for Voice Extensions for the above SIP Users

8) nano /etc/asterisk/extensions.conf

9) **Function 2:** SIP Extensions for call processing

[internal]

```
exten =>9111 ,1 ,Answer()
exten => 9111,2,Dial(SIP/9111,60)        Syntax: Dial(type/identifier, timeout)
exten =>9111,3,Playback(vm.nobodyavail)  If the call goes unanswered
exten =>9111,4,VoiceMail(9111@main)      Syntax: VoiceMail(Sipuser@context)
exten => 9111,5,Hangup()                 To Hung Up the call if the call goes unanswered
```

Dial( ) application will ring all of the specified destinations simultaneously and bridge the inbound call with whichever destination channel answers first.

```
exten =>9112 ,1 ,Answer()
exten => 9112,2,Dial(SIP/9112,60)
exten =>9112,3,Playback(vm.nobodyavail)
exten =>9112,4,VoiceMail(9112@main)
exten => 9112,5,Hangup()
```

```
exten =>9113 ,1 ,Answer()
exten => 9113,2,Dial(SIP/9113,60)
exten =>9113,3,Playback(vm.nobodyavail)
exten =>9113,4,VoiceMail(9113@main)
exten => 9113,5,Hangup()
```

```
exten =>9114 ,1 ,Answer()
exten => 9114,2,Dial(SIP/9114,60)
exten =>9114,3,Playback(vm.nobodyavail)
exten =>9114,4,VoiceMail(9114@main)
exten => 9114,5,Hangup()
```

* **For VoiceMail Forwarding:**

exten =>1111,1,VoiceMain(9111@main) exten => 1111,2,Hangup()   **To Hung up the call after answered.**
exten => 1112,1,VoicemailMain(9112@main) exten => 1112,2,Hangup()
exten => 1113,1,VoicemailMain(9113@main) exten => 1113,2,Hangup()
exten => 1114,1,VoicemailMain(9114@main) exten => 1114,2,Hangup()

10) mv   /etc /asterisk/voicemail.conf   /etc/asterisk/voicemail.conf.orig

From the syntax VoiceMail(Sipuser@context**) whenever a call is to be processed it should be checking for the password for that particular sip user which are specified in the "**context=**[main]" function written in the following file.**

For creating a file in which we can give the password

11) nano   /etc /asterisk/voicemail.conf

12**) [CRYPTO(3)] -Voicemail configurations**

[main]

```
9111 => 111
9112=> 222
9113 => 333
9114 => 444
```

13) nano   /etc /asterisk/ari.conf

ARI (Asterisk Rest Interface)

FUNCTION 3: Ari.conf

```
[general]
    enabled=yes
    allowed_origins=localhost:8088,http://ari.asterisk.org
```

```
[asterisk]
    type=user
    read_only=no
   password=asterisk
```

14) nano /etc /asterisk/http.conf **(HTTP Configuration for binding with Router if needed)**

```
[general]
enabled=yes
bindaddr=0.0.0.0
bindport=8088
```

If we need to extend our network , the raspberry pi voip server can be communicated with the Routers using this bindport 8088 configured.

15) asterisk – r

16**)** reload **command for Error checking**

17) sip show peers **To check the sip users which are connected and active.**

18) core show calls **To check the active calls**