# DESIGN AND IMPLEMENTATION OF FREE WIRELESS INTERCOM SYSTEM

**(Case study: Kampala International University)**

Final Year Project Report Submitted to Kampala International University in Partial Fulfillment of the Requirements for the Award of the Degree

of

Bachelor of Science in Telecommunications Engineering

**BY**

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## DECLARATION

I **ENERST EDOZIE** hereby declare that the material submitted in this final year project has been compiled by me and it has never been published or submitted to any other university or institution of higher learning for the purpose of meeting any academic requirement.

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Signature…………………………… Date………………………………………….

# APPROVAL

We have read and hereby recommend this Final Year Project entitled "**Design and Implementation of free Wireless Intercom System**" for acceptance by Kampala International University in partial fulfillment of the requirements for the award of the degree of Bachelor of Science in Telecommunication Engineering of Kampala International University.

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**Supervisor**

## DEDICATION

This project is dedicated to Almighty God, my dear parents, uncle, brothers, sisters and friends who stood with me in every aspect and supported me in one way or another to see that this final year project was successful. Despite the challenges faced, they have always given me courage and resilience to face them which has always motivated me to continue on to the very end.

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| | |
|---|---|
| 2G | $2^{nd}$ Generation |
| CDMA | Code Division Multiple Access |
| DSL | Digital Subscriber Line |
| GPRS | General Packet Radio Service |
| GSM | Global System for Mobile Communications |
| IP | Internet Protocol |
| IVR | Interactive Voice Response |
| LAN | Local Area Network |
| LTE | Long Term Evolution |
| PABX | Private Automatic Branch Exchange |
| PBX | Private Branch Exchange |
| PC | Personal Computer |
| PCI | Peripheral Component Interconnect |
| POTS | Plain Old Telephone System |
| PSTN | Public Switched Telephone System |
| SIP | Session Initiation Protocol |
| SIP-T | Session Initiation Protocol for Telephones |
| UMTS | Universal Mobile Telecommunications System |
| WAP | Wireless Access Protocol |
| WAN | Wide Area Network |
| WPA | Wi-Fi Protected Access |
| WIFI | Wireless Fidelity |
| P2P | Peer to peer |

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# ABSTRACT

In modern technology, wireless communication is widespread and it started to be used in everyday life, for example, Wireless phone charger, home automation systems, etc. widely use wireless communication. People want smart systems to make their life easy, and I want to discard the cables that limit our mobility.

The purpose of this project is to design and implement a free intercom system that uses WIFI in p2p (Peer-to- Peer) or WLAN (Wireless Local Area Network) as a means of communication between mobile phones, computers (soft phones), and IP phones at no cost. The system will allow users to make voice, Video, and text message conversation. As this system uses internet protocol, the voice data can be send easily on the network between sources to destination. The switching method used is packet switching. In this paper it shown how the voice packet data will be send from source to destination using Asterisk based server and WIFI based clients. As the communication is real-time the session initiation protocol (SIP) is used and some parameters such as packet sending and receiving, jitter and bandwidth utilization will be monitored in the server.

The design provides a solution that eliminates the use of expensive hard-wired intercoms and Private Automatic Branch Exchange (PABX) by making use of Asterisk and WIFI.



**Keywords**: GSM (phone), IP, WIFI, p2p, WLAN, Asterisk.

# CHAPTER ONE

# INTRODUCTION

## 1.0 Introduction

This project presents the design of free wireless Intercom system.
This is an idea of the wireless communication between two smartphones or a smartphone and a telephone module or softphone. In this system, it is used for the reservation of venue/location/time like seminar halls, auditoriums, etc. using Wi-Fi call without using internet and network service providers. This system makes the communication within the organization area at free cost.

## 1.1 Background

The asterisk is the software design for the communication of voice over the internet protocol. This technology is worldwide used in the companies for their internal voice communications. Here the Asterisk is the software used as a backbone for the voice communication. This software is installed on Linux server or any other operating system. This system is the replacement of the old EPABX system which uses wires and the switching hardware. But because of the Internet protocol the IP based private branch exchange having more advantages over the Electronics private branch exchange [3]. Also WIFI communication will be there to distribute wireless in the house or office intercom system [1]. This is client server, in this server I have the Linux, Asterisk and the clients will be desktop or mobile phones. In desktop the system the software will be X-lite or Zoiper which is very popular SIP client by counter path organization. In mobile there are so many SIP apps available. Example Zoiper, X-lite etc. I can use any VoIP softphone in the mobiles. Asterisk supports audio protocols such as SIP which is Session Initiation Protocol used for the audio communication. This package consists of several features such as Call Waiting, Caller ID, Call Transfer, voice call, blacklist, voicemail, recording, interactive voice response, ringing tones, etc.

SIP (Session Initiation Protocol) is a protocol used in VoIP communications allowing users to make voice and video calls, mostly for free and it is a major signaling protocol used in voice over IP [5]. SIP is a protocol that initiates and manages interactive user sessions involving voice, video, instant messaging, and other such multimedia sessions. SIP works with voice over

IP to integrate voice and data on a single network .SIP which is used to initiate, manage, and terminate interactive sessions between one or more users over the Internet.

### 1.2 Statement of the Problem

The African countries are changing (developing) and therefore makes influence on technology requirements. For this reason, I came up with the design of free wireless Intercom system, to help making phone call within user office, home, business areas and in many other places at zero cost, without physical connection between the caller and the receiver.

## 1.3 Objectives

The main objective is to designing a free wireless intercom system using Asterisk, WIFI and softphone to establish call between the caller and the receiver within a WIFI range at zero cost system.

### 1.3.1 General Objective

- ✓ To design a LAN server communicate system.
- ✓ To implement free wireless intercom system.
- ✓ To configure the system and generate numbers.

## 1.4 Research question

- ✓ How to design a LAN server communicate system.
- ✓ How to implement free wireless intercom system.
- ✓ How to configure and generate numbers for users.

## 1.5 Significance of the study

This system will be cost-effective, compact and easily maintainable system that solves the problem of so many officers and other people in the world to use their mobile phones freely during call process.

## 1.6 Scope of the study

### 1.6.1 Content scope

The system is to develop free WIFI call by using asterisk software and create configuration files, to install Linux on computer and to install asterisk on Linux. To program the user registration, the dial plan for calling and to test the calling on WIFI network using softphone.

### 1.6.2 Geographical scope

The study will be conducted in Universities, hospitals, offices, homes, business areas etc. around Uganda.

### 1.6.3 Time scope

This project is based on both theoretical and methodological data, thus it is approximated to take a maximum of twelve (12) months.

# CHAPTER TWO

# LITERATURE REVIEW

## 2.0 Intercom System

In the past, the goal of telecom engineers is to provide better services at any cost. The costs were being imposed on a customer. To this end, only the rich could afford these services. There have been changes to this situation over the years. The industry is driving to the positive direction where better services are being provided at very low cost to customers.

In addition, telecom companies have experienced a significant increase in number, which has led to a high level of competition among them. At the same time, the number of customers has also grown tremendously. Thus, there is need for better management of resources such as optimization of the quality of the services. Trade-offs need to be made between costs, quality, and priorities. There are currently systems like Whatsapp, Imo, Facebook, Telegram, VOIP, etc. which are useful for low cost communication. For using these services we need to have access to internet connection. It could be a costly affair for small companies. Installation and maintenance of wired LAN is long and costly affair.

In early days of voice transfer PSTN networks were used. These consisted of Private Branch Exchange office owned by service providers. Wired LAN was later employed to transfer voice and video over local area network. As it was wired systems connected though it lacked mobility. Also configuring LAN required time. Wires need to be setup to individual-PCs. Troubleshooting and maintaining this type of network was great trouble. Also topology like star could bring whole network down if central hub fails. In the LAN applications fiber lines mostly serve the backbone to interconnect servers and other high-speed elements of the local networks.

Comparatively installation of WIFI is simple and quicker. Maintenance required is also less. It is also easier to troubleshoot. Hence I came up with a free wireless telephony system for audio and video calls. The motive behind this system is to enable the cost effective for voice and video communication. Also there is no need for internet connection for working of this system.

## 2.1 Wired Intercoms

Every intercom product line is different, most analogue intercom systems have much in common. Voice signals of about a volt or two are carried a direct current power rail of 9V, 12V, 24V, 30V or 48V which uses a pair of conductors. Signal light indications between stations can be accomplished through the use of additional conductors or can be carried on the main voice pair via tone frequencies sent above or below the speech frequency range. Multiple channels of simultaneous conversations can be carried over additional conductors within a cable or by frequency- or time-division multiplexing in the analogue domain. Multiple channels can easily be carried by packet-switched digital intercom signals. Portable intercoms are connected primarily using common shielded, twisted pair microphone cabling. Building and vehicle intercoms are connected in a similar manner with shielded cabling often containing more than one twisted pair.

Some digital intercoms use Category 5 cable and relay information back and forth in data packets using the Internet protocol architecture.

Most of the time, it is hard to install an intercom system through a building or apartment, because of the cabling effort.

## 2.2 Wireless Intercoms

For installations where it is not desirable or possible to run wires to support an intercom system, wireless intercom systems are available. There are two major benefits of a wireless intercom system over the traditional wired intercom.

The first is that installation is much easier since no wires have to be run between intercom units. The second is that you can easily move the units at any time. With that convenience and ease of installation comes a risk of interference from other wireless and electrical devices. Nearby wireless devices such as cordless telephones, wireless data networks, and remote audio speakers can interfere. Electrical devices such as motors, lighting fixtures and transformers can cause noise. There may be concerns about privacy since conversations may be picked up on a scanner, baby monitor, cordless phone, or a similar device on the same frequency. Asterisk free wireless intercoms can reduce or eliminate privacy risks, while placement, installation, construction, grounding and shielding methods can reduce or eliminate the detrimental effects of external interference.

## 2.3 Wireless Communication Principles

The term “wireless” is normally used to refer to any type of electrical or electronic operation which is accomplished without the use of a "hard wired" connection. Wireless communication is the transfer of information over a distance without the use of electrical conductors or "wires". The distances involved may be short (a few meters as in television remote control) or very long (thousands or even millions of kilometers for radio communications). When the context is clear the term is often simply shortened to "wireless". Wireless communications is generally considered to be a branch of telecommunications.

It encompasses various types of fixed, mobile, and portable two way radios, cellular telephones, personal digital assistants (PDAs), and wireless networking. Other examples of wireless technology include GPS units, garage door openers and or garage doors, wireless computer mice and keyboards, satellite television and cordless telephones.

The term "wireless" has become a generic and all-encompassing word used to describe communications in which electromagnetic waves or RF (rather than some form of wire) carry a signal over part or the entire communication path.

**Wireless communication may be implemented via:**

- Radio frequency (RF) communication,
- Microwave communication, for example long-range line-of-sight via highly directional antennas, or short-range communication, or
- Infrared (IR) short-range communication, for example from remote controls.

A wireless communication system deals with two directions, a transmitting direction and a receiving direction. Normally, the size of the antenna must be as large as one fourth of the wavelength of the signal to be transmitted or received to get enough efficiency. For this reason, the original signal (normally the voice) with a large wavelength must be transferred to a higher frequency (smaller wavelength) to downsize the antenna. At the transmitting end, the original signal is imposed on a locally generated radio frequency (RF) signal called a carrier. This process is called modulation.

This carrier signal, along with the information signal imposed on it, is then radiated by the antenna. At the receiving end, the signal is picked up by another antenna and fed into a receiver where the desired carrier with the imposed information signal is selected from among all of the other signals impinging on the antenna. The information signal (e.g., voice) is then extracted from the carrier in a process referred to as demodulation.

The propagation of the signal in free space is not fluent. There may be some negative influence to the received signal. First, the signal becomes weaker and weaker during the propagation. Second, interference from some noise will distort the information signal; And then, some propagation mechanisms such as reflection, diffraction and scattering, will distort the information signal too. It is even worse in mobile systems -- where one or both of the terminals (transmitters and receivers) can move about -- due to an environment that changes dynamically from moment to moment. To get rid of these problems, we need more precise models to describe the propagation, and more modulation technology to avoid distortion.

## 2.4 History

The term "Wireless" came into public use to refer to a radio receiver or transceiver (a dual purpose receiver and transmitter device), establishing its usage in the field of wireless telegraphy early on; now the term is used to describe modern wireless connections such as in cellular networks and wireless broadband Internet. It is also used in a general sense to refer to any type of operation that is implemented without the use of wires, such as "wireless remote control", "wireless energy transfer", etc. regardless of the specific technology (e.g., radio, infrared, ultrasonic, etc.) that is used to accomplish the operation.

### Digital Wireless Communication

Digital describes electronic technology that generates, stores, and processes data in terms of two states: positive and non-positive. Positive is expressed or represented by the number 1 (ON state) and non-positive by the number 0 (OFF state). Thus, data transmitted or stored with digital technology is expressed as a string of 0's and 1's. Each of these state digits is referred to as a bit (and a string of bits that a computer can address individually as a group is a byte).

The modulation techniques of digital wireless signal are more complicated than those of analog signals. There are a lot of modulation techniques. Following is just a small list of them:

**Spread Spectrum Modulation Techniques:**

DS-SS: Direct Sequence Spread Spectrum

FH-SS: Frequency Hopped Spread Spectrum

**Linear Modulation Techniques:**

BPSK: Binary Phase Shift Keying

QPSK: Quadrature Phase Shift Keying

DPSK: Differential Phase Shift Keying

**Combined Linear and Constant Envelope Modulation Techniques:**

MFSK: M-ary Frequency Shift Keying

MPSK: M-ary Phase Shift Keying

QAM: M-ary Quadrature Amplitude Modulation

**Constant Envelope Modulation Techniques:**

BFSK: Binary Frequency Shift Keying

GMSK: Gaussian Minimum Shift Keying

MSK: Minimum Shift Keying

When wireless applications, such as mobile phones, were initially introduced into society, they were based on analog technology, as the prime service was voice. Analog was a suitable means, capable of delivering the service. However, the requirements & expectations of the current consumer often exceed this service to include data as well. Thus the market is being driven to satisfy the requirements of data & voice, which can be more adequately delivered by digital technologies. Most consumers prefer digital to analog because of its superior performance & service providers are embracing digital technologies as well. By examination of the various advertising media nowadays, one can see that they are attempting to convince their clients using analog to convert to digital. In a way, analog can be looked on as a predecessor to digital in the field of wireless technologies.

**Some advantages of digital wireless communication over analog:**

- It economizes on bandwidth.
- It allows easy integration with personal communication systems (PCS) devices.
- It maintains superior quality of voice transmission over long distances.
- It is difficult to decode.
- It can use lower average transmitter power.

- It enables smaller and less expensive individual receivers and transmitters.
- It offers voice privacy.

## 2.5 Voice Coding

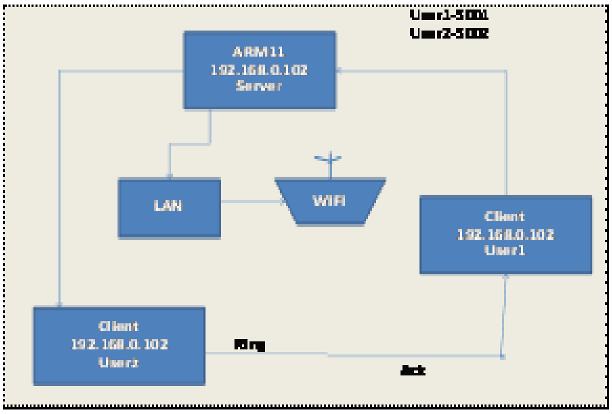
Digital audio uses digital signals for sound reproduction. This includes analog-to-digital conversion, digital-to-analog conversion, storage, and transmission.

Digital audio has emerged because of its usefulness in the recording, manipulation, mass-production, and distribution of sound. Modern distribution of music across the internet through on-line stores depends on digital recording and digital compression algorithms. Distribution of audio as data files rather than as physical objects has significantly reduced costs of distribution.

The objective of speech is communication whether face-to-face or cell phone to cell phone. To fit a transmission channel or storage space, speech signals are converted to formats using various techniques. This is called speech coding or compression. To improve the efficiency of transmission and storage, reduce cost, increase security and robustness in transmission, speech coding attempts to achieve toll quality performance at a minimum bit rate.

## 2.6 Previous Projects

Most of the research previously done has been focused on providing communication explanation for businesses and other establishments consuming several different systems and only few actually utilizes asterisk as the telephony switching PBX. Hence, one of the existing structure is identical to the proposed result. The similar project was done by Miss.Pramila. B. Bamnote (M.E(Electronics),RAIT, Mumbai University, India) in the year 2016, her aim is introduce of VoIP and implementation of WIFI based intercom system using ARM11. Her project is cheap to use, but not completely free, because the clients must use internet. The block diagram of her project is shown below.



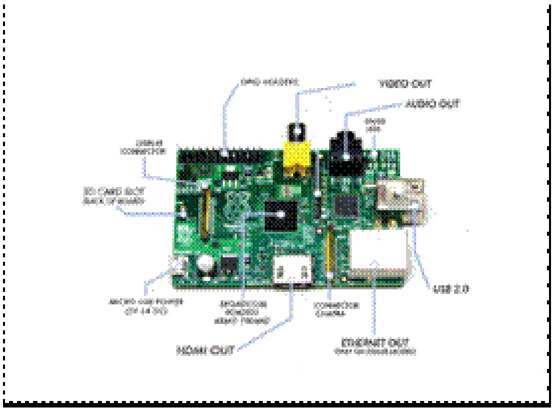



Figure 1.0 Block of the previous

# CHAPTER THREE

# METHODOLOGY

## 3.0 Introduction

The server will be Linux based computer system. In this minimum AMD system with Asterisk version 1.4.x, Linux kernel 2.4 and above, min 512 MB RAM, and 80GB hard disk is used. The client will be any laptop or mobile handset.

## 3.1 The Design of free wireless intercom System

In this project, I used Linux OS (operating system). As in windows7, 8, or 10 I can't install asterisk directly, a Linux OS is flexible in installing asterisk. This product is largely composed of software packages distributed under free software license and source code. To open asterisk Linux commands are necessary and useful. To copy files or to change directories etc., I used Linux commands in asterisk. I installed Linux operating system first, which can be flexible and act as an operating system for installing of asterisk. Asterisk can run a number of operation systems.

Linux is the only officially supported OS.

Asterisk supports most SIP telephones, acting both as registrar and back-to-back user agent, and can serve as a gateway between IP phones and the public switched telephone network. So after installing asterisk we have to login to Linux machine as the root user.

The Basic block diagram of the free wireless intercom system is as shown below. In the diagram, the server is connected to users using WIFI and give IP address to the users.


Figure 1.1 Block diagram of the system

The first thing to do is to create a configuration file in sip.conf which can register a device with asterisk. In sip.conf I created very simple configuration file that allows the SIP phones to connect with asterisk. Dial plan applications are used in extension. conf to define the various actions that can be applied to call. Calls come in on channels and are then handed to the "extensions.conf" file. Dial plan contains logical sections of matches called 'Contexts,' and each channel sends a call into the dial plan with a context name and a dialed number. The dial plan then matches the number being dialed, and runs applications accordingly. Each match on the dialed number has an order of steps called 'Priorities', and are indicated with an integral incrementing number. Asterisk.conf configuration shows how asterisk runs as a whole. After creating all the above information now the users are ready to make a call, I opened the root firstly, make the system to connect to the Wireless LAN. To see whether it is pinging or not. I typed ifconfig to see the server IP, I gave the mobile softphone the same IP to connect asterisk server.

### 3.2 WORKING Principles

In the server I created users having username and password with a number
(Example 1000, 1111, 3332, 1003, 1004 even 2018 or 2020…..)
When the server is connected to the WI-FI the smart phone app get connected to server via WI-FI.

This client and server systems will get the IP address from WI-FI access point.
All are now in the network and the service of the asterisk server will start in the system.
Now the call can be establish in the WI-FI network.

## 3.3 HARDWARE Resources

### 3.3.1 Wireless Router

A router is a networking device that forwards data packets between computer networks. Routers perform the traffic directing functions on the Internet. A data packet is typically forwarded from one router to another router through the networks that constitute the internetwork until it reaches its destination node. A router is connected to two or more data lines from different networks. When a data packet comes in on one of the lines, the router reads the address information in the packet to determine the ultimate destination. Then, using information in its routing table or routing policy, it directs the packet to the next network on its journey. This creates an overlay internetwork.

D-Link WIFI router is the corporate name of a designer and manufacturer of networking, broadband, digital, voice and data communications solutions. The company offers consumer devices in addition to providing network connectivity solutions to small and medium-sized business. The company's products and services include networking solutions for wireless bridging, network storage, cable and DSL modems, PoE adapters, network adapters, routers and more. D-Link was founded in 1986.

### 3.3.2 Computer (Desktop or Laptop)

**A computer** is an electronic device that processes data, into meaningful information that is useful to people. It is an electronic device that accepts input, processes it, stores data, and produces output. Or an advanced electronic device that accepts input, processes it, stores data, and produces output.

**Basic Functions**

- ❖ **Accepts data (input):** Receives data from outside (input device) for processing.
- ❖ **Process data (Processing):** Performs operations or manipulations on data particularly numerical data.
- ❖ **Produce output (output)** Produces data from within for external use.
- ❖ **Stores results (Storage):** Holds data internally before, during and after processing. Hard disks, CD-ROM, DVD ROM, Tapes and others.

## 3.4 SOFTWARE Resources

### 3.4.1 Asterisk:

Asterisk is basically a telephony toolkit enabling developers to create numerous types of applications that interface with telephone networks. The most obvious application is that of a PBX. Asterisk can also be used as an IVR (Interactive Voice Response) system, for teleconferences and as a voicemail system. Asterisk is, however, most commonly used to build hybrid PBX systems that utilize modern PCI cards instead of banks of switches and relays, and software instead of custom hardware. By using relatively simple PCI cards in a standard x86 computer system running on Linux, the cost to build a working system is greatly reduced as compared to the often expensive and inflexible traditional PBX

Asterisk is also primarily developed on GNU/Linux for x/86. It is known to compile and run on GNU/Linux, OpenBSD, FreeBSD, and Mac OS X Jaguar.

## FEATURES

### Presence

Asterisk provides limited SIP presence support. SIP subscriptions are used to show Online/Offline status. It's limited in the way that you can't see busy or away status. Presence with full functionality can be implemented in Asterisk with a third party IM services called Wildfire. The connection between these systems is a plugin called Asterisk-IM.

### SIP proxy

Asterisk is not a SIP proxy. A SIP proxy handles call control on behalf of other user agents and usually does not maintain state during a call and therefore is never the endpoint of a call. Asterisk, as a server, is a SIP registrar and location server and also acts as a user agent endpoint (soft phone).

### Protocols

- Asterisk supports a wide array of different protocols:
- IAX
- H.323
- SIP
- MGCP (Media Gateway Control Protocol)
- SCCP (Skinny Client Control Protocol)

### Management

Configuration in Asterisk is text based. This is fast and effective, but requires more knowledge from staff.

### Scalability

The number of users in Asterisk is not defined, but is thought to be CPU bound.

### Community Support

Asterisk has been around for a long time, and has enormous support from the open source community. It is the classical open source solution.

**Installation**

- Need extensive Linux knowledge to get the system up and running with source install.
- No binary support
- Maintainers of package based Linux systems like (apt and yum) make's is possible to install and update the system with one simple command.
- Very good install documentation
- Easy to add phones.
- Username and password is the only requirement.

### 3.4.2 LINUX Operating System

Linux is the best-known and most-used open source operating system. As an operating system, it is a software that sits underneath all of the other software on a computer, receiving requests from those programs and relaying these requests to the computer's hardware.

Linux is also a Unix-like, open source and community-developed operating system for computers, servers, mainframes, mobile devices and embedded devices. It is supported on almost every major computer platform including x86, ARM and SPARC, making it one of the most widely supported operating systems.

For the purposes of this page, I used the term "Linux" to refer to the Linux kernel, but also the set of programs, tools, and services that are typically bundled together with the Linux kernel to provide all of the necessary components of a fully functional operating system. Some people, particularly members of the Free Software Foundation, refer to this collection as GNU/Linux, because many of the tools included are GNU components.

### 3.4.3 Softphone

A soft phone is a software program for making telephone calls over the Internet using a general purpose computer, rather than using dedicated hardware. The soft phone can also be installed on a piece of equipment such as a workstation, portable computer, tablet or even a cell phone and allows the user to place and receive calls without requiring an actual telephone set. Often a soft phone is designed to behave like a traditional telephone, sometimes appearing as an image of a phone, with a display panel and buttons with which the user can interact. A soft phone is usually used with a headset connected to the sound card of the PC, or with a USB phone.

**Zoiper**

Zoiper is compatible with most VoIP service providers and PBXs. Enjoy free calls between Zoiper users or combine our dialers with your favorite provider for the cheapest calls. Combine multiple providers for the cheapest route to every destination.

## 3.5 Implementation

### 3.5.1 Configurations

Registering users with the Asterisk Server The "sip.conf" file contains parameters relating to the configuration of sip client access to the Asterisk server. Clients must be configured in this file before they can place or receive calls using the Asterisk server.

The following lines are to be written in the "sip.conf" file in the "/etc/asterisk". The users defined in this file are only allowed to connect to the Asterisk server using a Soft phone for computers and smart phones. If a user is not declared here, he or she cannot be connected to the server system and cannot place calls using the Asterisk server.

### 3.5.2 Codes

```
[hod-elect]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0

[hod-civ]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0

[hod-mec]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0
```

```
[tlc-dept]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0

[exam-room]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0

[dean-office]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0
```

```
[admin-office]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0

[ict-office]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0

[security-dept]
videosupport=yes
disallow=all
allow=g729
allow=gsm
type=friend
alwaysauthreject=yes
canreinvite=no
secret=mysps123
permit=192.168.56.101/255.255.255.0
```

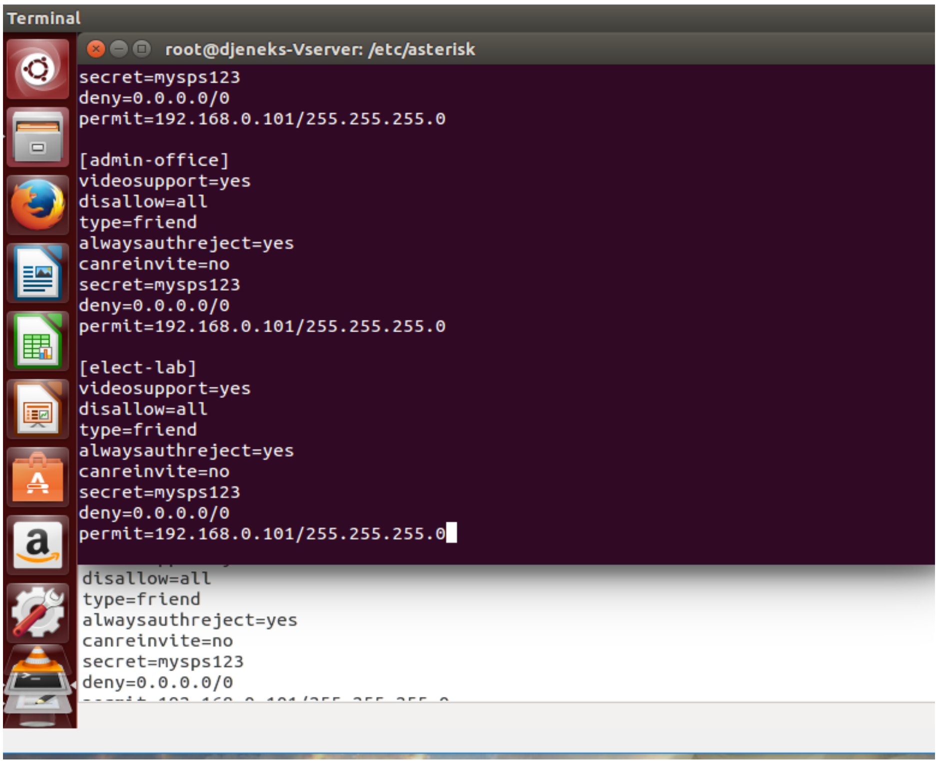



Figure 1.2 The users' configuration

### 3.5.3 Creating a Dial Plan

After declaring the users, now comes the task of creating a dial plan which describes the call flow and defines what actions will the Asterisk server performs when a specific number is dialed by a user. The "extensions.conf" file lays out the dial plan, bringing channels together with applications and services. "extensions.conf" features extension matching logic and intelligent call routing logic. The dial plan is defined in the "extensions.conf" file present in the"/etc/asterisk" folder. The following lines provides an example of a dial plan used for implementing the work

## [Users]

exten=>1111,1,Dial(SIP/hod-elect,20)

exten=>1112,1,Dial(SIP/hod-civ,20)

exten=>1113,1,Dial(SIP/hod-mec,20)

exten=>1114,1,Dial(SIP/tlc-dept,20)

exten=>2221,1,Dial(SIP/exam-room,20)

exten=>2222,1,Dial(SIP/ict-office,20)

exten=>2223,1,Dial(SIP/project-room,20)

exten=>1234,1,Dial(SIP/admin-office,20)

exten=>3331,1,Dial(SIP/elect-lab,20)

exten=>3332,1,Dial(SIP/dean-office,20)

exten=>3333,1,Dial(SIP/staff-room,20)

exten=>3344,1,Dial(SIP/hod-bic,20)

exten=>3334,1,Dial(SIP/eneks-josh,20)

exten=>3335,1,Dial(SIP/security-dept,20)

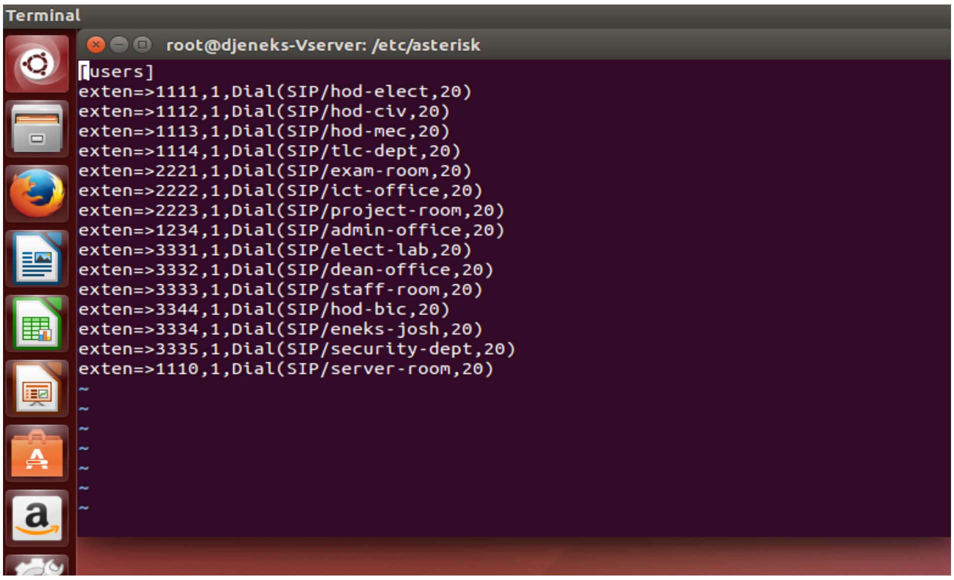



Figure 1.3 Dial plan configuration

### 3.5.4 Server logs

There are several types of logs available while a call is in active state. Writing “sipdebug=yes” in the “sip.conf” file enables viewing these logs. The following types of logs are enabled and stored for the implementation of this work:

Logs when user authenticates:

The security aspect of Asterisk is further enhanced, with the registered user authentication that it performs, each time user enters in the Wi-Fi cloud. In the above fig, SIP 200 OK status code indicates successful authentication. As can be seen, the requesting IP192.168.0.101 from port 3210 is authenticated by a registered SIP client caller id "ict-office". Whereas, if the authentication fails, status code of form 4XX indicating error is displayed by the server. On successful completion of authentication

SIP provide further message requests to clients which are listed under "Allow" attribute of authentication. In addition, further client related details like SIP client used for access, are maintained by the server.

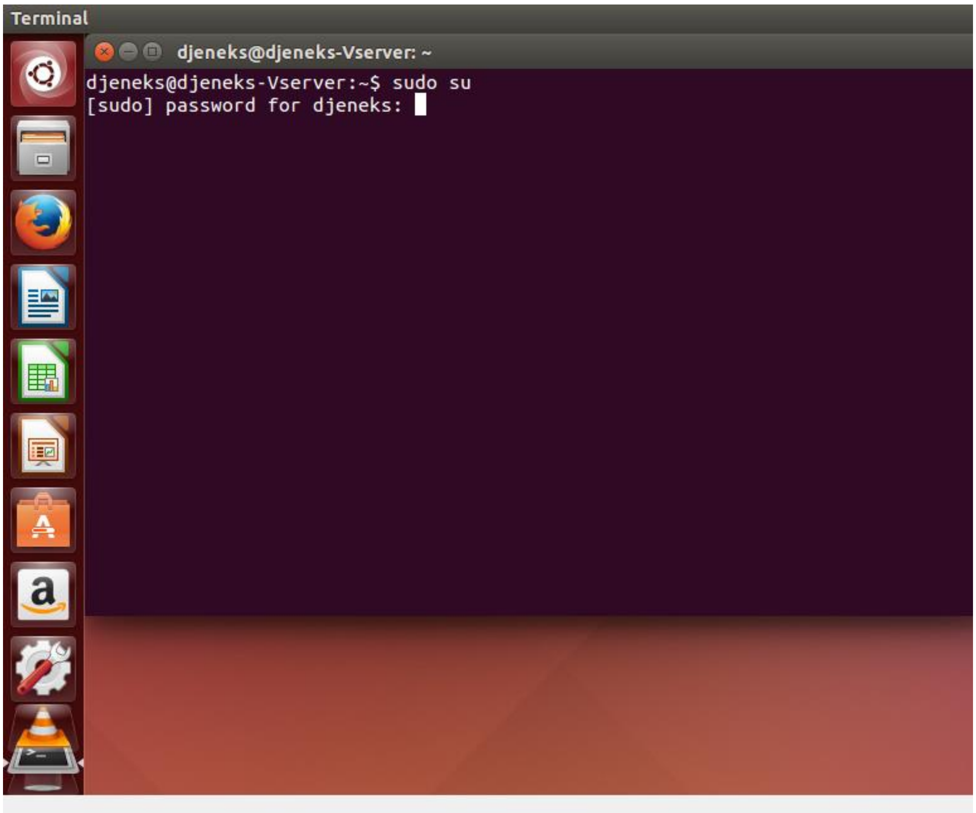



Figure 1.4 Root login page (enter the password created during installation)

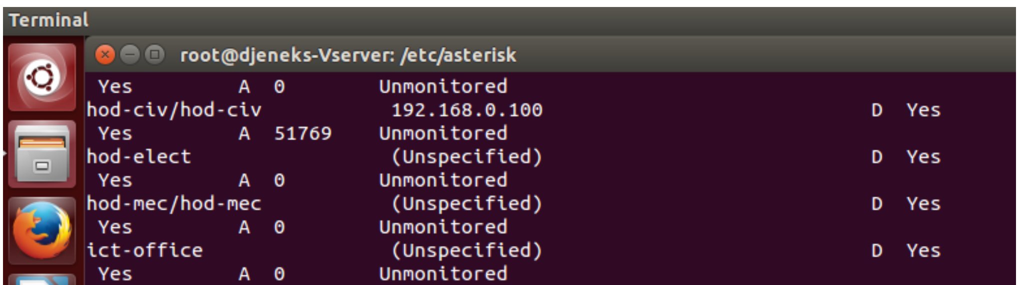


Figure 1.5 Online and offline users

### 3.5.5 Experimental setup

Experimental setup Steps for calling the users

- ✓ On the Wi-Fi access soft phones registers its fixed IP. Where the Wi-Fi will update this soft phone being active.
- ✓ Each phone is identified by a user name. Updating the IP address with a corresponding username.
- ✓ If you call any user name and the information is available to all users logged in the network.
- ✓ When the Wi-Fi range is not available, then call handoff.

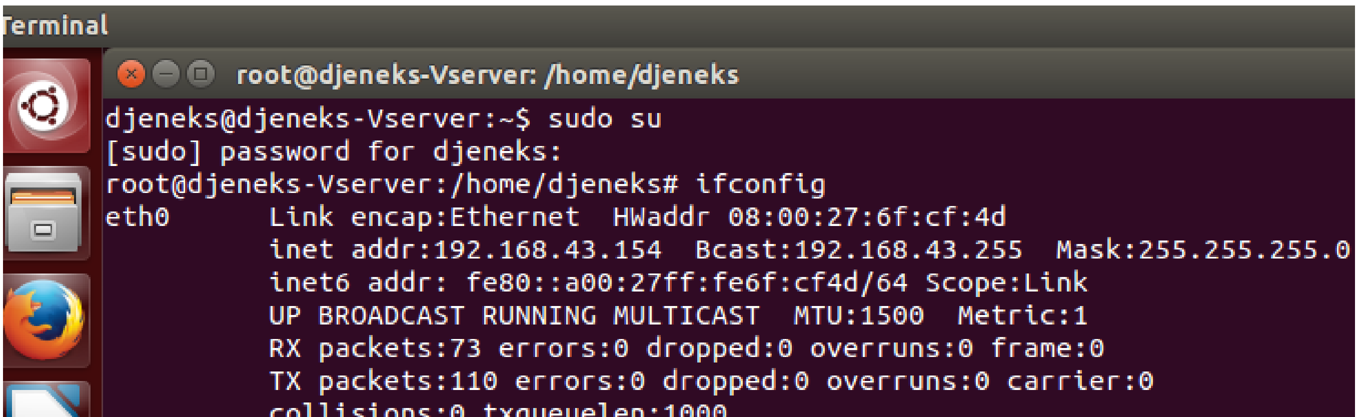
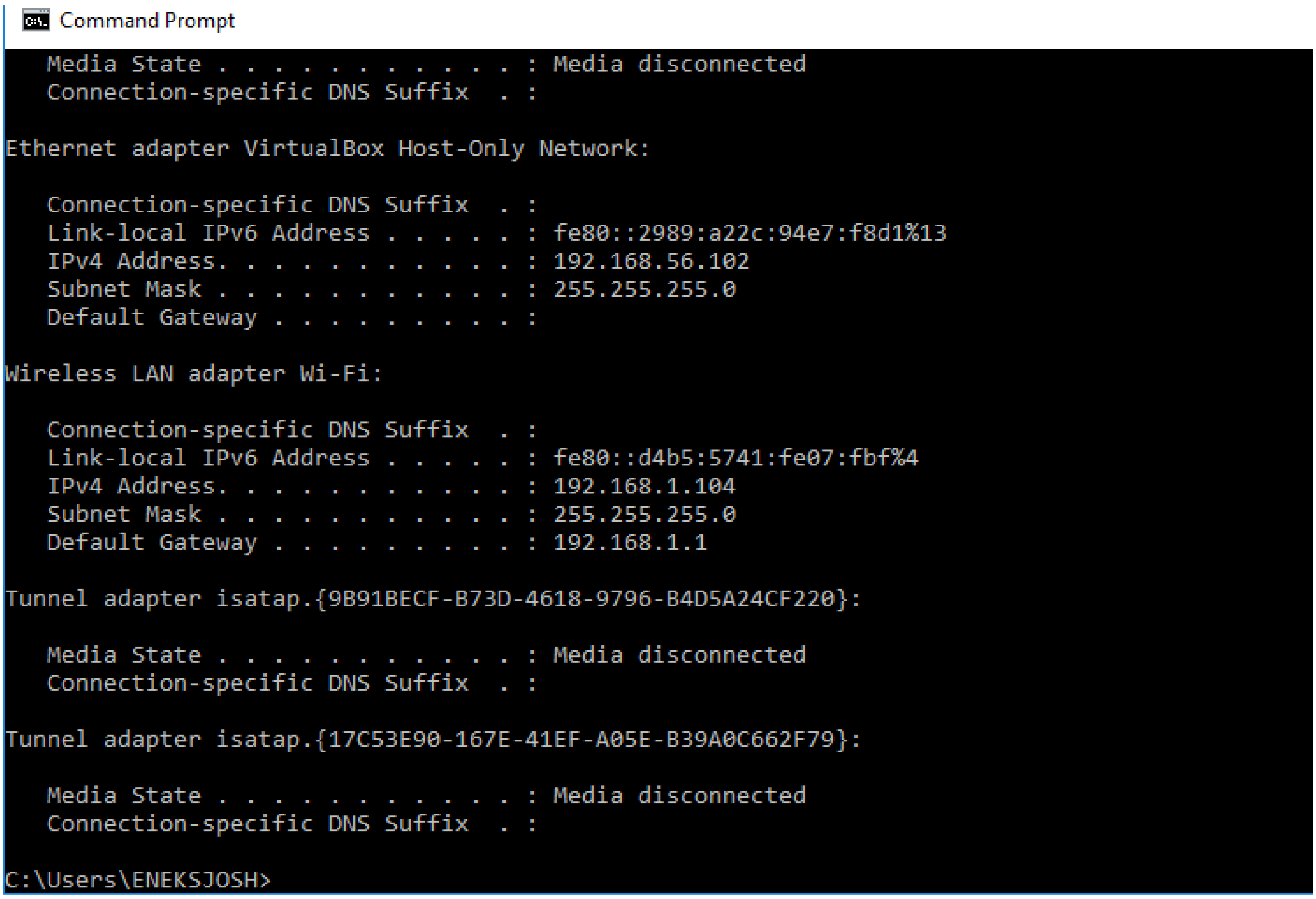


Figure 1.6 The Network IP address



```
Command Prompt
   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Ethernet adapter VirtualBox Host-Only Network:

   Connection-specific DNS Suffix  . :
   Link-local IPv6 Address . . . . . : fe80::2989:a22c:94e7:f8d1%13
   IPv4 Address. . . . . . . . . . . : 192.168.56.102
   Subnet Mask . . . . . . . . . . . : 255.255.255.0
   Default Gateway . . . . . . . . . :

Wireless LAN adapter Wi-Fi:

   Connection-specific DNS Suffix  . :
   Link-local IPv6 Address . . . . . : fe80::d4b5:5741:fe07:fbf%4
   IPv4 Address. . . . . . . . . . . : 192.168.1.104
   Subnet Mask . . . . . . . . . . . : 255.255.255.0
   Default Gateway . . . . . . . . . : 192.168.1.1

Tunnel adapter isatap.{9B91BECF-B73D-4618-9796-B4D5A24CF220}:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

Tunnel adapter isatap.{17C53E90-167E-41EF-A05E-B39A0C662F79}:

   Media State . . . . . . . . . . . : Media disconnected
   Connection-specific DNS Suffix  . :

C:\Users\ENEKSJOSH>
```

Figure 1.7 Computer and VirtualBox IP address

# CHAPTER FOUR

## 4.1 Result

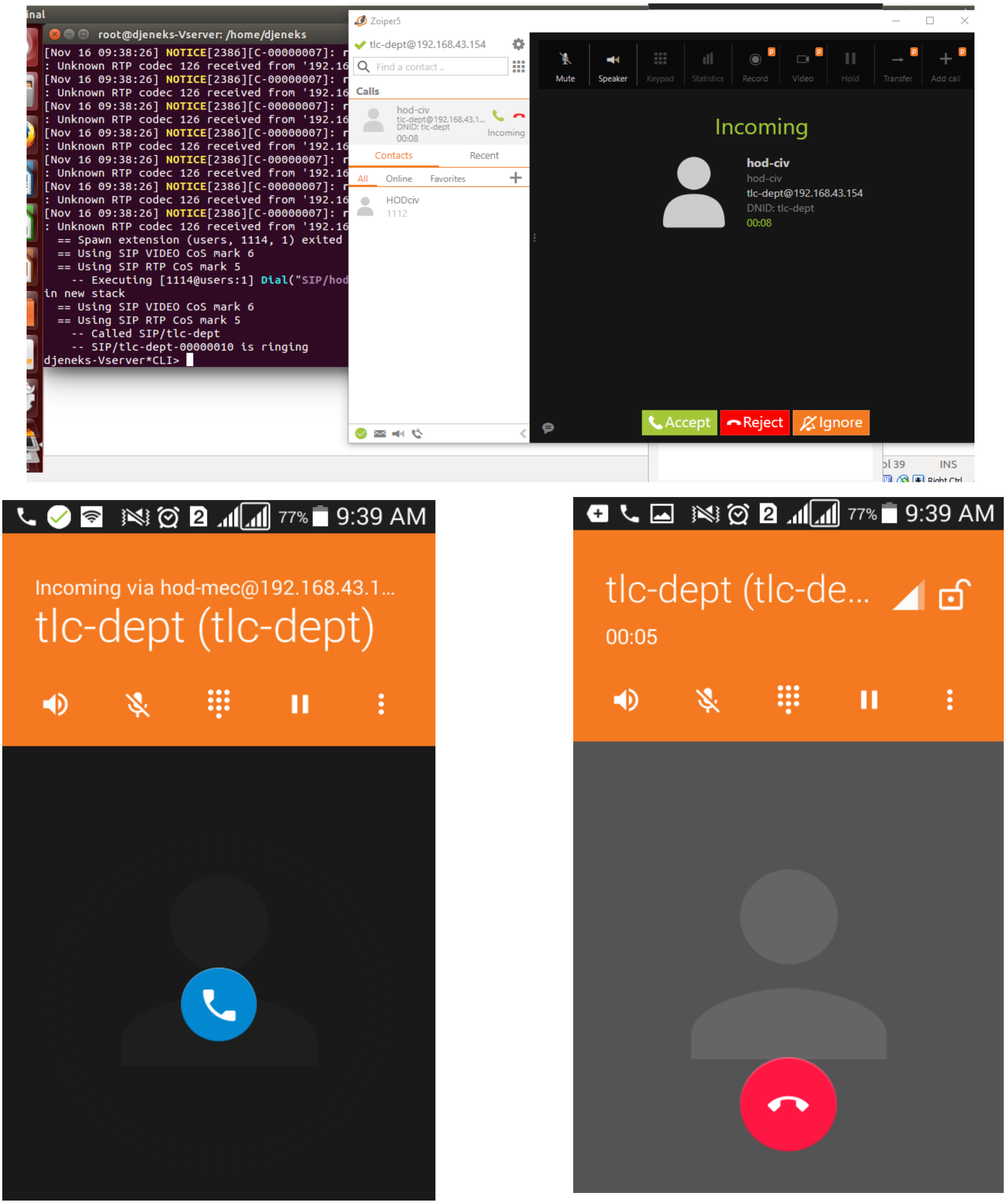



Figure 1.8 Incoming and Received call

**After testing:**

- ➢ Able to make free voice call without SIM and Internet using WIFI.
- ➢ Call is in process with high voice clarity, there is no interference
- ➢ Call Waiting feature is also there, you here beep notifications of an incoming call when you are already busy attending another call.

**Maximum numbers of calls are in process**

- ➢ Simultaneously.
- ➢ Only limitation is that, calls can be made within the Wi-Fi range, but for increasing range of my network I have to use router with high range or I can use Wi-Fi boosters also.

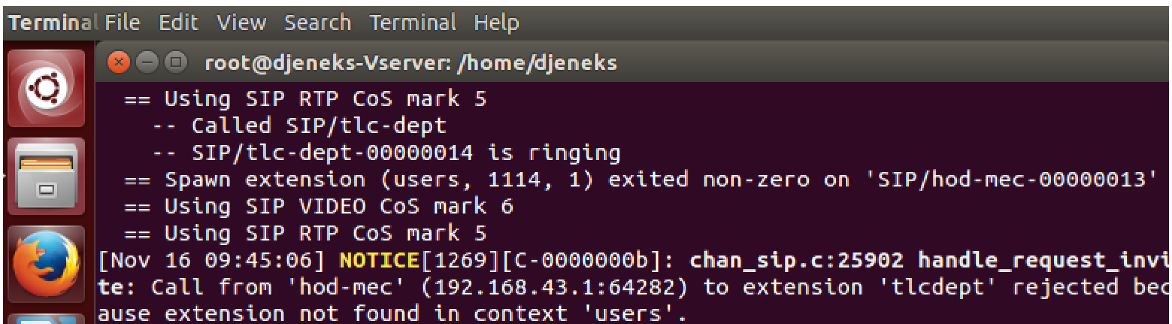
**It consumes very less power supply**

- ➢ 12 Volt for Wi-Fi Router and 20 Volt for computer system.

This project can be implemented in colleges, Universities, big and small organizations etc so that the departments can communicate with each other free of cost.

## 4.2 Experimental results

Following results are obtained when a call is established between two softphones either computer or smart phone.



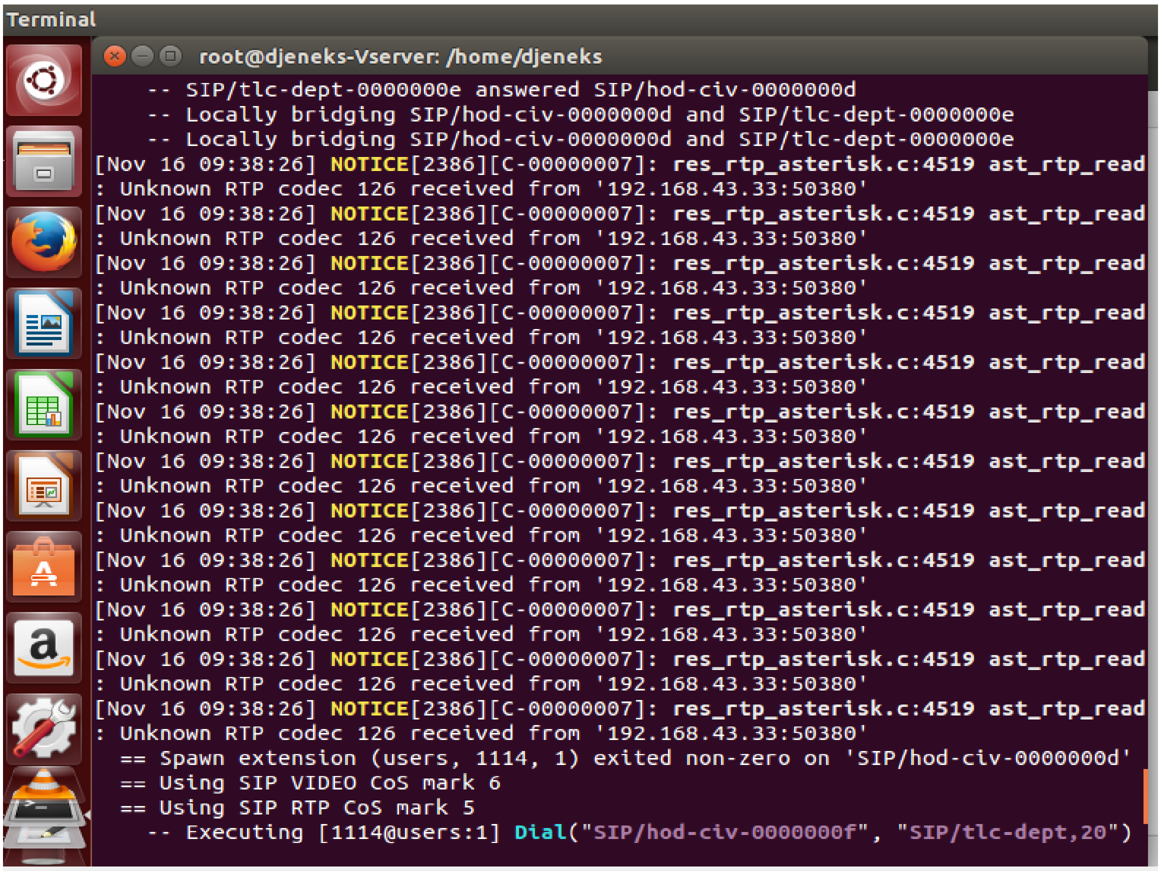
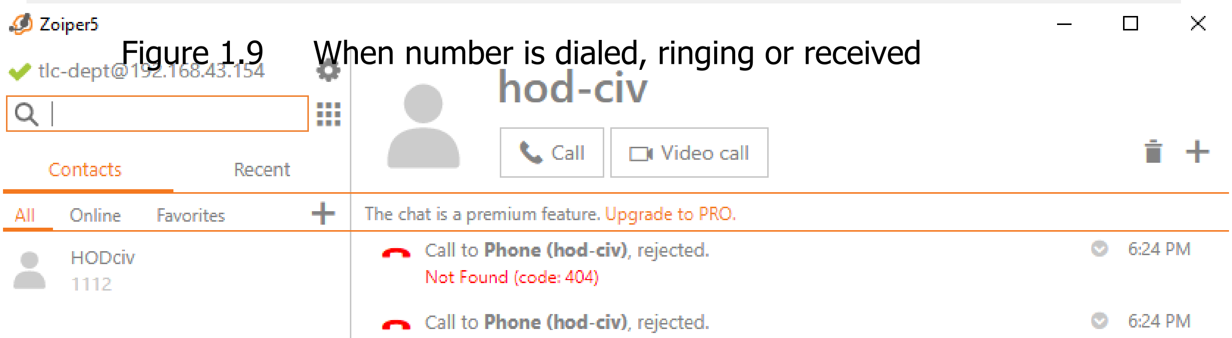



Figure 1.9 When number is dialed, ringing or received



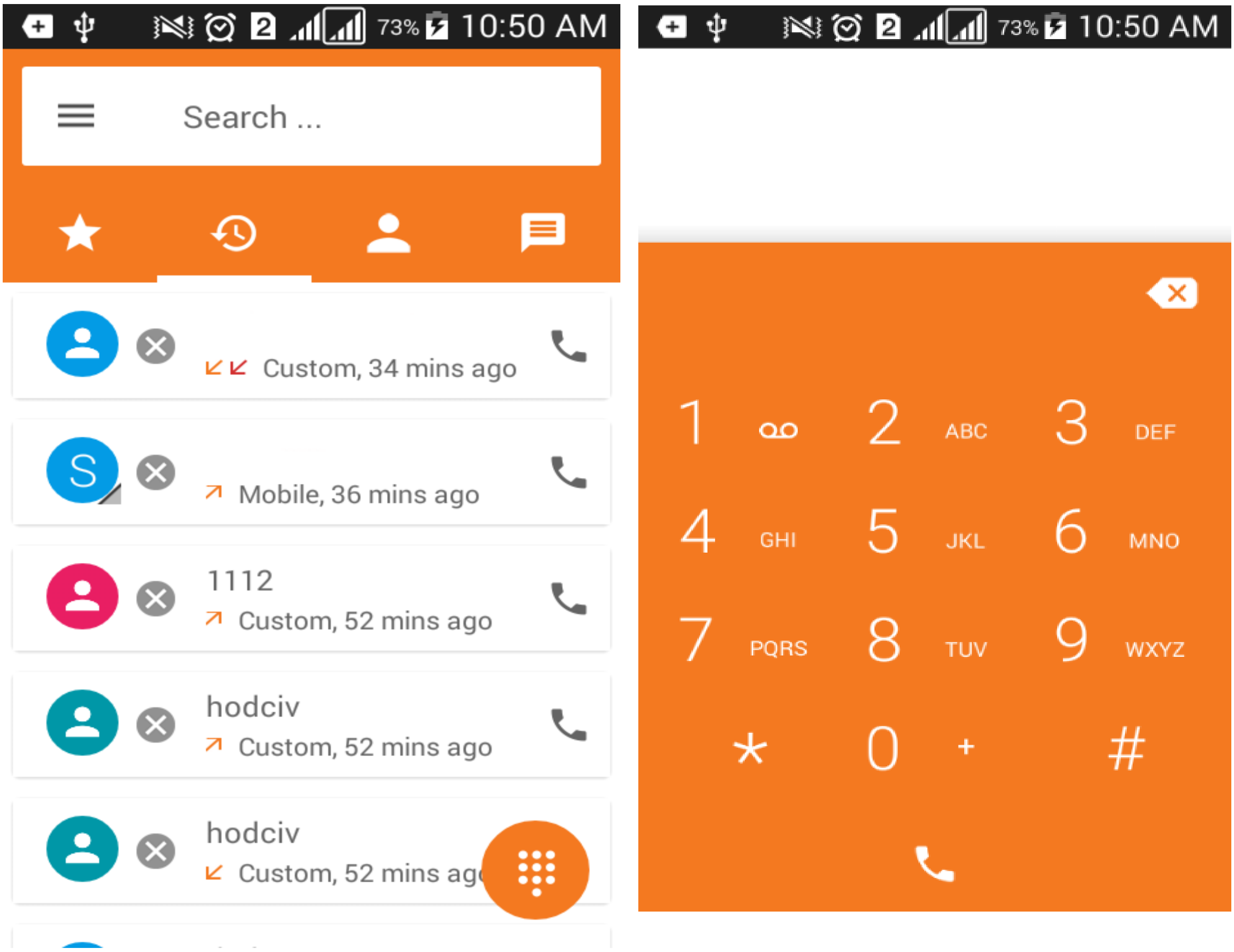



Figure 2.0 The softphone keypad and call status

**CHAPTER FIVE**

## 5.0 Conclusion

Wireless technology is one of the most widely using technologies which support to deal with communication from one place to another without any physical wire connection. This free wireless intercom system is enabling not just free calls but also providing more advantageous and rich features and more flexible services. Although, challenges stay behind, this new technology will play a key function in businesses communications, small and big offices, schools, homes etc. Computer is where the Linux operating system run smoothly and so as per the cost factor the system for Calling on Wi-Fi as an intercom system where there is no need of internet and SIM card.

## 5.1 Recommendation

1. The use of a SIP trunk to connect two Asterisk servers located in different locations
2. The use of freePBX, an open source graphical user interface to notify the configuration file.
3. For high availability it is recommended to deploy at least three controllers to provide multiple controller nodes.

## 5.2 Future scope

1. To set up this network in city like Kampala.

2. To increase the number of calls and find the performance parameters.

3. To make a conference call without buying any software license key.

4. This project can be further extended to perform call handoff between multiple Wi-Fi routers so that user is allowed with greater mobility.

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