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Implementation and Analysis of A Free Wireless Intercom System

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Abstract: *In this modern technology era, wireless technology is one of the most widely used technologies which support communication from one place to another without any physical wired connection. 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 (softphones), and IP phones at no cost. The switching method used is packet switching. The system will allow users to make voice, Video, and text message conversation using internet protocol, the packet will be sent from source to destination using the 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 using 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: Asterisk, GSM, IP, WIFI, P2P, PBX, WLAN

1.Introduction

Many years ago voice transfer PSTN networks were used to convert consumer`s voice into packets and were improved to Private Branch Exchange(PBX)[1]. PBX system performs communication tasks such as inbound calls and outbound calls. A PBX system is performing VoIP algorithm. VoIP technology is used to transmits voice signals over the internet using a packet switching technique. Wired LAN was later employed to transfer voice and video over a local area network which lacks mobility. Also configuring LAN required time. Wires need to be set up to individual-PCs. Troubleshooting and maintaining this type of network was great trouble. Also, topology like star could bring the whole network down if the central hub fails. In the LAN applications fibre lines mostly serve the backbone to interconnect servers and other high-speed elements of the local networks. Comparatively installation of WIFI is simple and quick. Maintenance required is also less[2]. It is also easier to troubleshoot, cost-effective for voice and video communication. Also, there is no need for wide area network.

1.1 Aim and Objective

Aims and Objectives of different part describe in stages concerning to complete of this project.

1. To develop free WIFI call by using asterisk software.
2. To create configuration files.
3. To install Ubuntu server on a computer.
4. To install asterisk on Linux.
5. To program the user registration.
6. To program the dial plan for calling.
7. To install the developed system higher institution.
8. To test the calling on WIFI network using a softphone.
9. To make a voice call where there is no wide area network.

2. Literature survey

In 2012 Sandeep and Khanapurkar design and implement IP-PBX for a small business organization which uses Intranet as a backbone as each organization has pre-installed local area network. The network architecture for the IPPBX system is easy to design & Implement [3].In 2014 Mohammed et al propose a VoIP implementation system using Asterisk PBX. The researcher carried out live experimental studies of SIP voice traffic. the performance of voice calls initiated using SIP simulator for testing SIP protocol performance and found much more stability and accuracy using Asterisk PBX. The purpose is to suggest that VoIP technology attributes that best meet users' needs[4].in 2016 Pramila and Shweta design and implement a wife based intercom system using Arm11 [5]. In 2017 Akshay et al proposed portable voice communication system on raspberry pi, In 2018 Euclid et al propose wireless VoIP implementation using Asterisk PBX and open source softphone which is cost-effective and enable IP network infrastructure which in recent times could be extended to allow inter-institutional linkage to allow collaboration and research[6].

3. Methodology

Linux-based server operating system was used with a minimum AMD configuration of 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 Design of free wireless intercom system

This intercom system is largely composed of software packages distributed under a free software license and source code. To operate asterisk Linux commands are necessary and useful. To copy files or to change directories etc., Linux operating system was first installed on the host machine, which makes the setup more flexible and acts as an operating system for installing asterisk. Asterisk can run a number of operating systems. 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 log in to Linux machine as the root user.

3.2 The block diagram

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 an 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 an asterisk. In sip.conf I created a very simple configuration file that allows the SIP phones to connect with an asterisk. Dial plan applications are used in the 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 dialled number. The dial plan then matches the number being dialled and runs applications accordingly. Each match on the dialled 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 by navigating to the terminal as the root user, make the system to connect to the Wireless LAN. To see whether it is pinging or not. typed ifconfig to see the server IP, allocate an IP address to the mobile softphone which is on the same subnet to connect the asterisk server.

3.3 Working principles

In the server, username and password with a number (Example 1000, 1111, 3332, 1003, 1004 even 2018 or 2020) were created when the server is connected to the WI-FI the smartphone app gets connected to the server via WI-FI. This client and server systems will get the IP address from the 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 established in the WI-FI network.

3.4 Hardware resources

3.4.1 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.

Figure 1.2 Wireless router

3.4.2 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.

3.5 Software resources

3.5.1 Asterisk

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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.

3.5.2 Linux operating system

Ubuntu server is an open-source 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.

3.5.3 Soft-Phone

A softphone is a software program for making telephone calls over the Internet using a general-purpose computer, rather than using dedicated hardware. The softphone 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 softphone 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 softphone is usually used with a headset connected to the sound card of the PC, or with a USB phone.

3.5.4 Zoiper

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

3.6 Implementation

3.6.1 Configuration

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 Softphone for computers and smartphones. 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.

3.6.2 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 dialled 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 provide an example of a dial plan used for implementing the work

Users

```
[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)
```

Server logs

There are several types of logs available while a call is in an 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 a user authenticates:

The security aspect of Asterisk is further enhanced, with the registered user authentication that it performs, each time the user enters 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, the status code of form 4XX indicating an error is displayed by the server. On successful completion of authentication

SIP provides 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.

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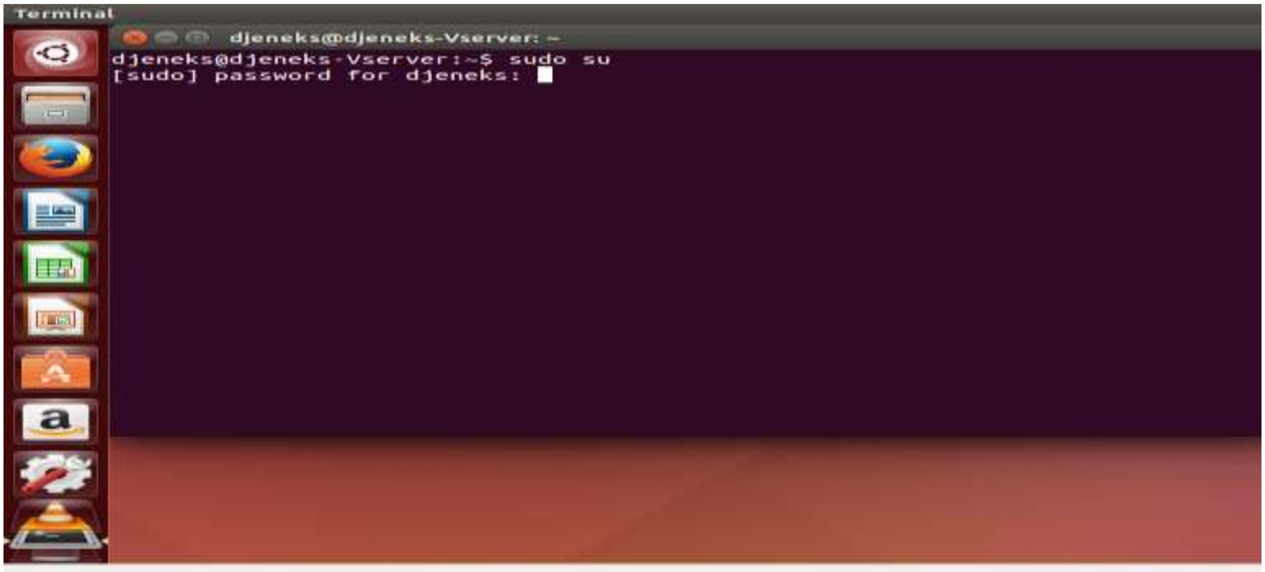

Figure 1.3 Root login page (enter the password created during installation)

3.7 Experimental setup

Experimental setup Steps for calling the users

- On the Wi-Fi access, softphones register its fixed IP. Where the Wi-Fi will update this softphone 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 1.4 The Network IP address

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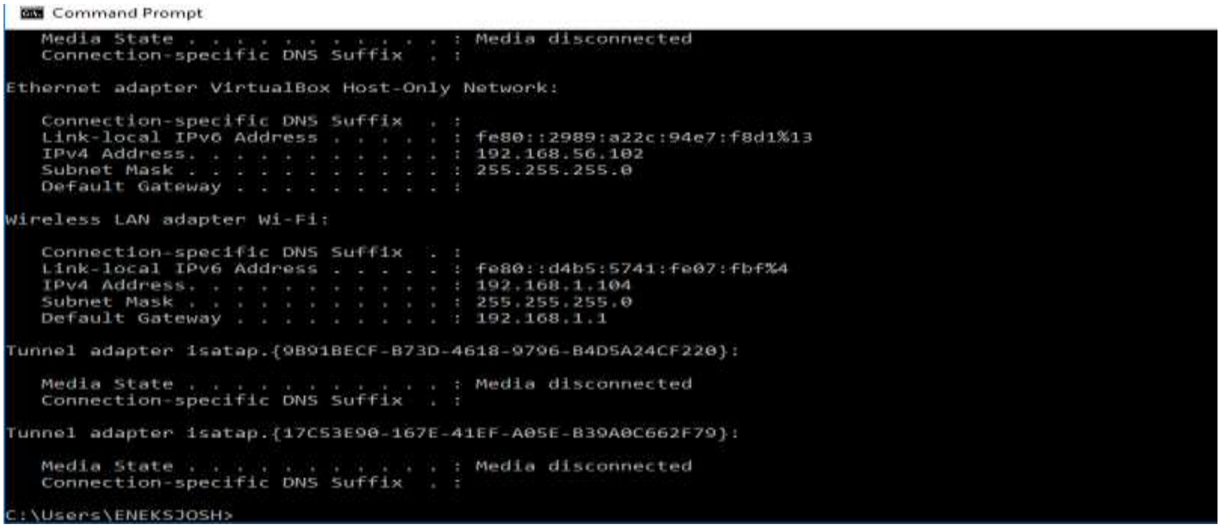

Figure 1.5 Computer and VirtualBox IP address

3.8 Result

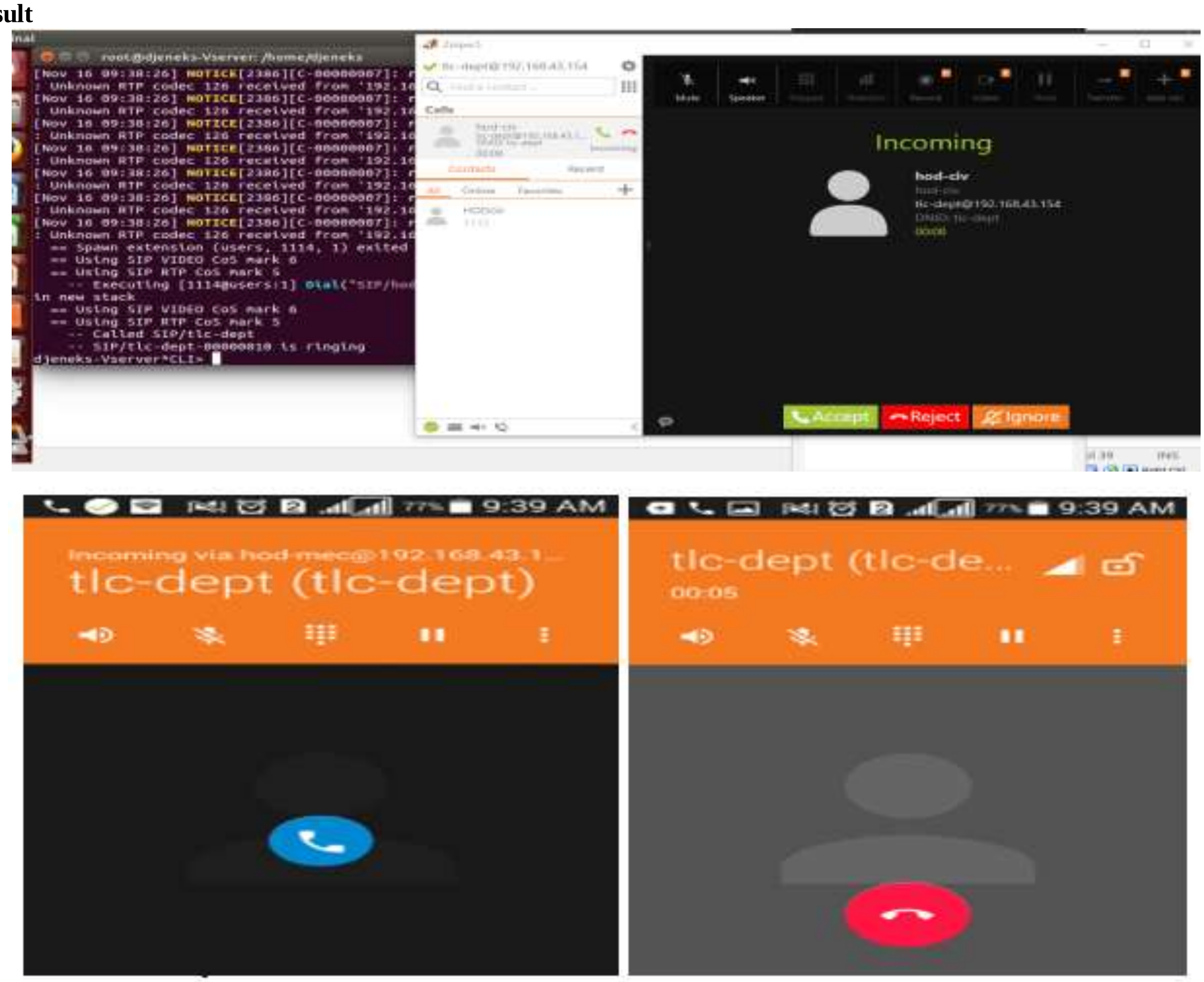

Figure 1.6 Incoming and Received call

Maximum numbers of calls are in the process Simultaneously.

It consumes very less power supply 12 Volt for Wi-Fi Router and 20 Volt for the computer system. This system can be implemented in colleges, Universities, big and small organizations etc so that the departments can communicate with each other free of cost.

Experimental Result

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

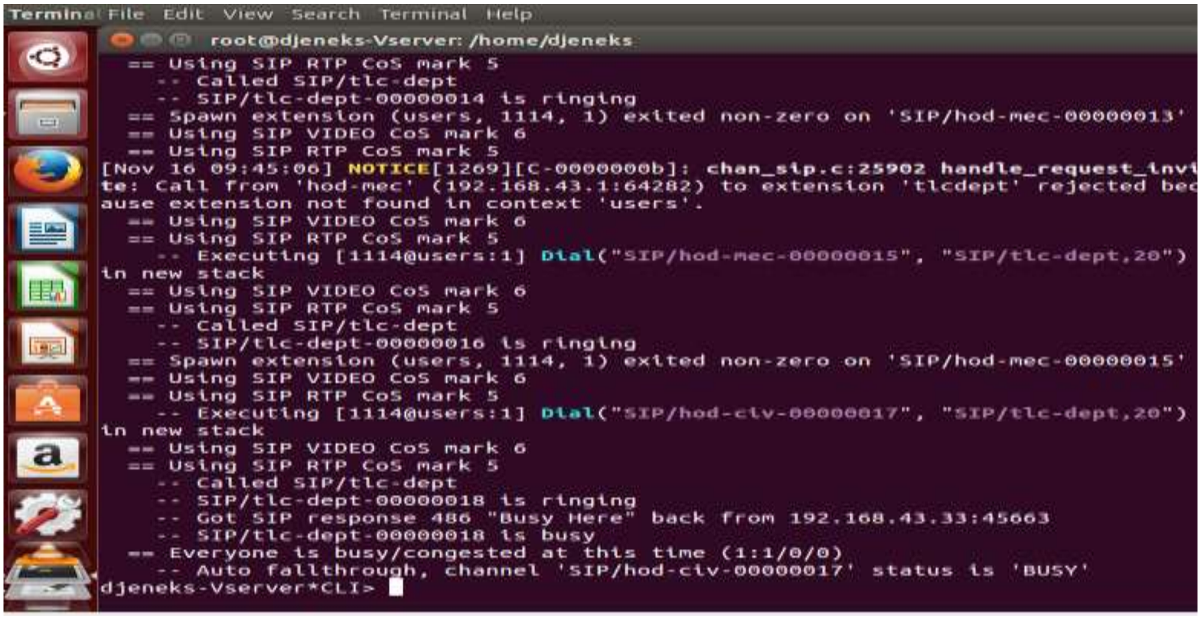

Figure: 1.7 The softphone keypad and call status

3.9 Conclusion

Wireless technology is one of the most widely used 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. The computer is where the Linux operating system runs 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.

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