VIRTUAL CALLING SERVICE

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ABSTRACT

The ability to transmit and process voice over Internet protocol (VoIP) networks has important implications for technology users especially by the using Asterisk PBX. Many companies nowadays are rushing to bring different VoIP products to market with a wide variety of features.

This project will focus on the introduction of VoIP and its implementation by the use of Asterisk PBX. Firstly, the abstract presents the project objective with some introductory theory about VOIP. Secondly, the project includes report on the viability of utilizing the Asterisk PBX as a foundation for conducting research performance studies for VoIP.

Finally, the project is showing on live experimental studies of SIP voice traffic. The article experimentally studied 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 those VoIP technology attributes that best meet users' needs. Asterisk, the open source PBX of choice is used to show that this is maturing fast and ready for main stream VoIP implementation.

Keyword: - *VoIP (VOICE OVER INTERNET PROTOCOL),Asterisk, PBX (PRIVATE BRANCH EXCHANGE), SIP (Service Initiation Protocol)*

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1. Introduction

An Implementation of a one-to-one model-based virtual phone system, THE VIRTUAL TALK, is presented.Virtual calling service deals with conventional calling between two users without the use of any SIM card or internet connectivity.

In rural areas where communication is mandatory Network service providers do not provide services until there are enough customers to provide satisfactory profits to the company, over coming this problem THE VIRTUAL CALLING SERVICE is going to provide a portable calling service with a wide range of 15 – 20 kms (using repeaters), free of cost.With Additional Features Video calling services will also be provided.

2. Description

Asterisk is a software implementation of a telephone private branch exchange (PBX); it allows attached telephone to make calls to one another, and to connect to other telephone services, such as the public switching telephone network (PSTN) and voice over internet protocol (VoIP) services.
Asterisk is released with a dual license model, using the GNU general public license (GPL) as a free software license and a proprietary software license to permit licensees to distribute proprietary, unpublished system components.Asterisk was created in 1999 by Mark Spencer of Digium. Originally designed for Linux, Asterisk runs on a variety of operating systems, including NetBSD, OpenBSD, FreeBSD, macOS and Solaris, and can be installed in embedded systems based on OpenWrt and on flash drives
Asterisk supports several standard voice over IP protocols, including the Session Initiation Protocol (SIP), the Media Gateway Control Protocol (MGCP), and H.323. 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 (PSTN) via T- or E-carrier interfaces or analog FXO cards. The Inter-Asterisk eXchange (IAX) protocol, RFC 5456, native to Asterisk, provides efficient trunking of calls among Asterisk PBX, in addition to distributing some configuration logic. Many VoIP service providers support it for call completion into the PSTN, often because they themselves have deployed Asterisk or offer it as a hosted application. Some telephones also support the IAX protocol.

3. Implementation

1. RASPBERRY PI:-

The Raspberry Pi is a series of small single-board computers developed in the United Kingdom by the Raspberry Pi Foundation to promote the teaching of basic computer science in schools and in developing countries.The original model became far more popular than anticipated, selling outside of its targets market for uses such as robotics. Peripherals (including keyboards, mice and cases) are not included with the Raspberry Pi. Some accessories however have been included in several official and unofficial bundles.

The Raspberry Pi primarily uses Raspbian OS, Debian- based Linux operating system.Other third party operating systems available via the official website include Windows 10 IoT Core, RISC OS and specialized KODI media center and classroom management. Many other operating systems can also run on the Raspberry Raspberry pi is connected wireless access point by using Ethernet cable.

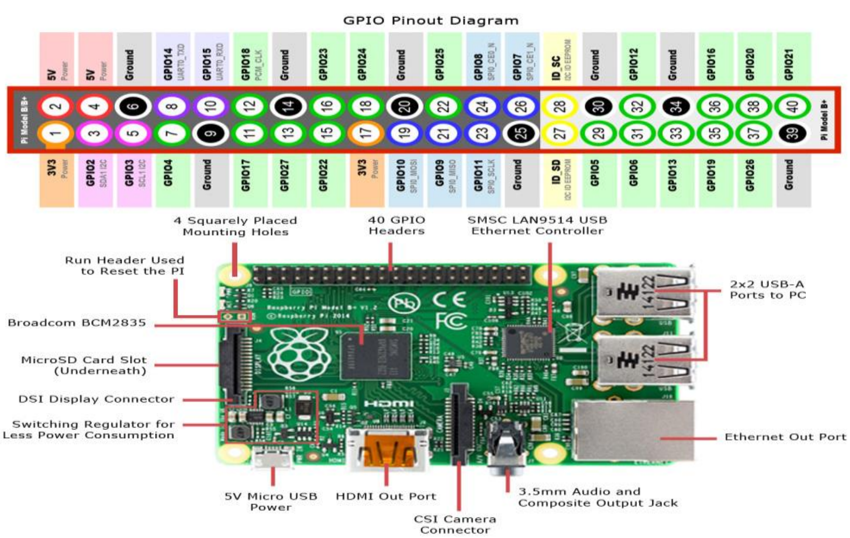

Fig 3.1: Raspberry pi board pin structure

2. WIRELESS ACCESS POINT:-

A **router** is a networking device that forwards data packet between computer network. 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 an internet work 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 network address information in the packet to determine the ultimate destination. Then, using information in its routing policy, it directs the packet to the next network on its journey.

The most familiar type of routers are home and small office router that simply pass IP packet between the home computers and the Internet. An example of a router would be the owner's cable or DSL router, which connects to the Internet through an Internet Service Provider (ISP). More sophisticated routers, such as enterprise routers, connect large business or ISP networks up to the powerful core router that forward data at high speed along the optical fiber lines of the Internet Backbone.Though routers are typically dedicated hardware devices, software-based routers also exist.

Fig 3.2: Wireless Access Point (Router)

Fig 3.2.1: Raspberry pi is connected to router with the help of Eternet cable

3. Zoiper:-

Zoiper is a VoIP application that lets you make chat or make voice and video calls with your friends, family, colleagues and business partners.
Unlike other software like Skype or Viber, it is open and can be used with any VoIP provider or PBX. Allowing for much more flexibility and cheaper or better quality termination.
Zoiper does not rely on Java, Flash or .NET but is written in oldsk00l **C/C++** and assembly. This results in **low memory** and **CPU usage** and makes for quality audio even on older hardware.

The setup consists of **Raspberry Pi 3** as the main server(data base) interfaced with an 300mbps router at 2ghz.

- For any user to access calling features in the virtual calling services needs to be registered on the Server.
- To register the user we need **Zoiper** (Asterisk Based Software/Application)which is to be installed in the mobile phones need to avail the services.
- After The successful registration, the user will be be provided with a unique ID number, this ID number will be used for calling or identifying different users in the Virtual service Range.

3. CONFIGURATION:-

Firstly we have to start wireless access point (WAP) and connect it with server(raspberry pi) providing supply to both modules, then by using IP address of WAP for D-link router that is 192.168.0.1 where we will find out active client IP address (ip address of server).

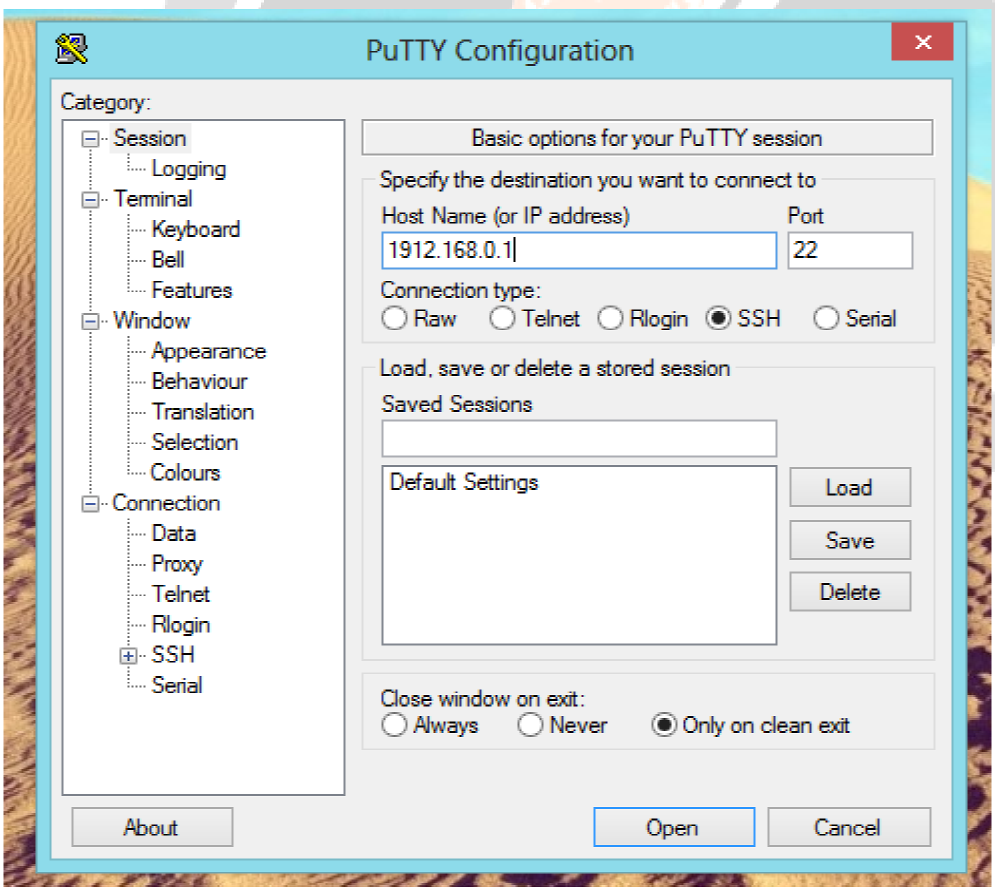

Fig 3.3: Putty Configuration

After getting IP address of server, open putty configuration app and write that IP address in Host name which result in opening of login window.

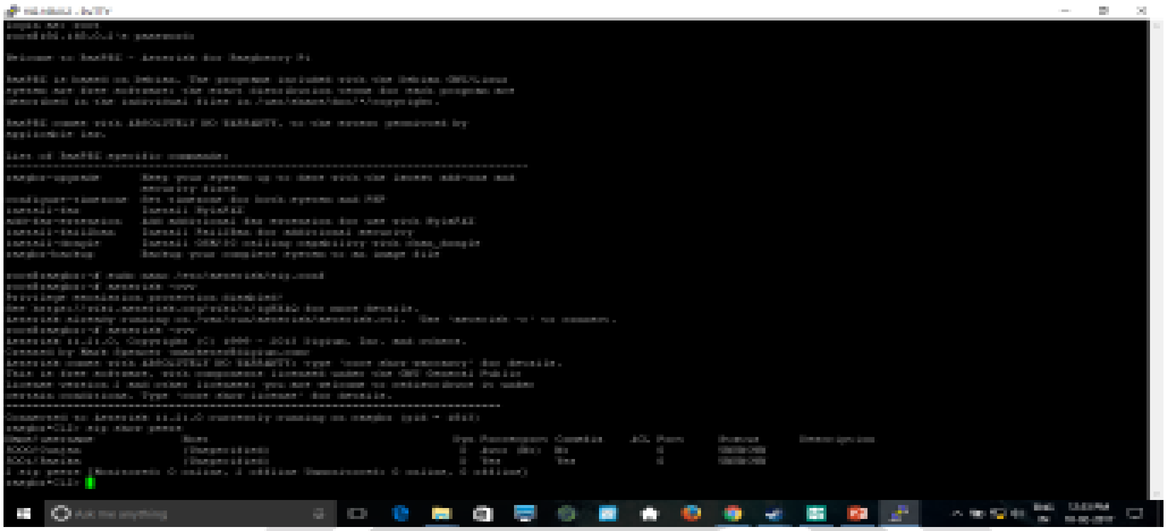

Fig 3.3.1: Login window of putty

In this window we start asterisk using command

asterisk -cvv

asterisk -rvv

When asterisk get start we program server using sip and extensions conf

4. BASIC CODING FOR ASTERISK

Basic code for SIP.CONF which is used for creating users is

```
[general]
port =5060
bindaddr=0.0.0.0
[1000]
username=sanket
secret=123456
host=dynamic
context=intercom
type=friend
allow=alaw
qualify=yes
```

Basic code for EXTENSIONS.CONF for establishing dialing is

```
[general]
static=yes
writeprotect=no
[global]
[default]
[intercom]
exten=>_1xxx,1,Dial(SIP/${EXTEN}];
```

exten=>_1xxx,2,Hangup();

Then using "core reload" and "sip show peers"command avaibility of user can be found. On other side ZOIPER app has to be downloaded on mobile or computers, after registering on app with proper username call can be establish with in that wifi region without SIM/INTERNET.

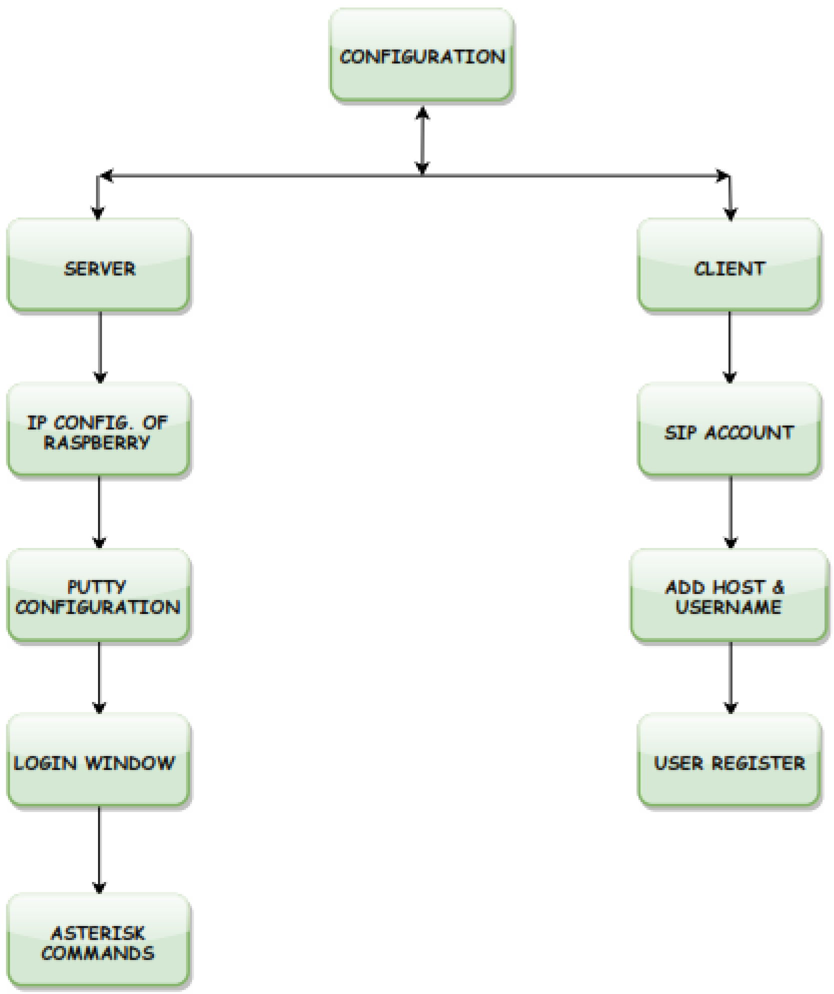

Fig 4: Configuration

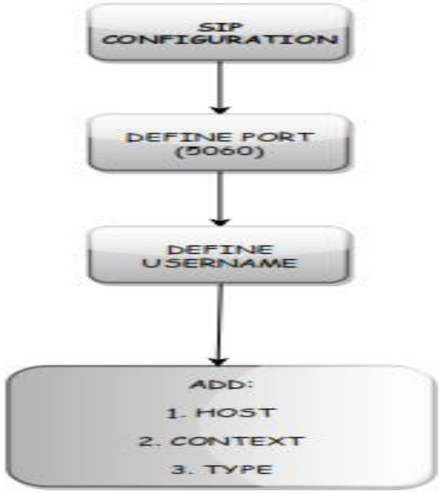

Fig 4.1: SIP Configuration

5. RESULT

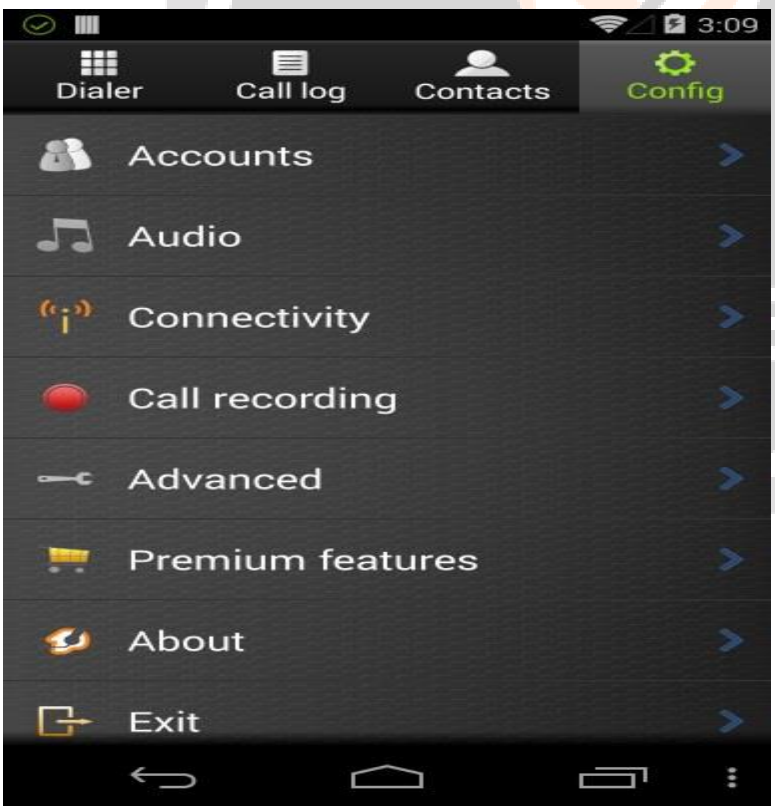

Fig 5: Zoiper Account

Fig 5.1: Zoiper Account Creation

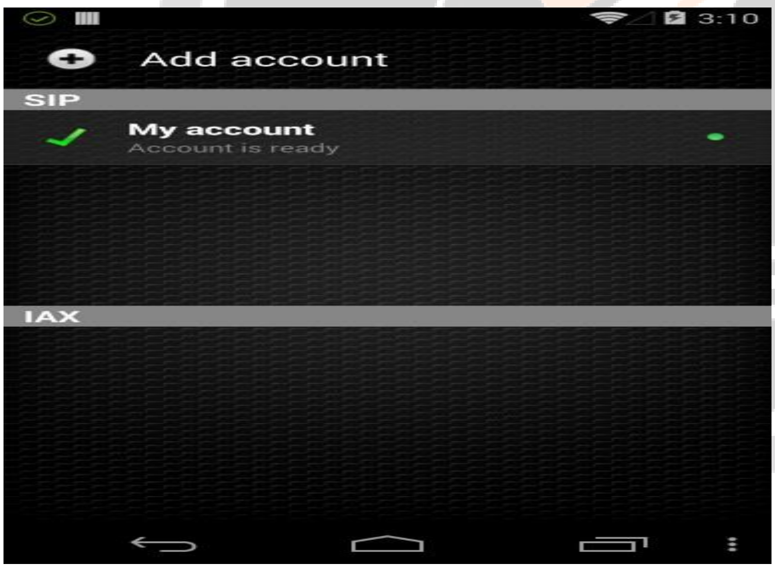

Fig 5.2: Zoiper Account Ready

Once account you can call the registered number within your network area.

In this paper, we have demonstrated that calling without sim and internet is possible.We have successfully develop model on virtual calling service by using asterisk on raspberry pi. We have also learn that distance depends on wifi range of router. More power the router higher the range for calling. It is possible to increase range up to 50 km.

6. SCOPE:-

1.Free communication on smartphone in the wifi range of system without internet and without SIM card.
2. Reception is voice recorded message and as per the number pressed the call is transfer to the smartphone in the network.
3.Reducing the cost of Communication.
4.Video calling can be establish for free in future.
5.WiFi range can be increased for higher range routers or using repeaters which can increased distance upto 50km.

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