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# Portable Wi-Fi Calling And Interactive Voice Response System

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*Abstract* - **Implementation of a VoIP telephony system using an IP Telephony solution in the organization as IVR. A new technology VoIP or Internet Telephony means that your voice is carried over the IP network, otherwise known as the Internet. Voice which is an analog signal, is converted to digital data, which is then disassembled and transmitted through the Internet or Internet to be recovered back to an analog signal an the other using an IP Telephony solution which is an Linux base system.**

**This service can be properly managed and deployed over a network with less stress and expenses. The system main server also has integrated in its other communication services such as voicemail, mobile calling on Wi-Fi network without using any cellular network connection and has many advance features.**

*Keywords:---* **Voice over Internet Protocol (VoIP), Internet Protocol (IP), Interactive Voice Response (IVR), Session Initiation Protocol (SIP), Private Branch Exchange (IP-PBX), CentOS, Asterisk.**

## I. INTRODUCTION

Mostly Electronic Private Branch Exchange (EPBX) is used to conduct telephone calls over a wired network. With the development of computing technology, Internet Protocol Private Branch Exchange (IP-PBX) has been established as an alternative to traditional EPBX system. Evolving from circuit switching to a more efficient packet switching model the VoIP protocols and codecs has enabled a remarkable change in the transmission of vocal communications.

While implementing Internet Protocol Private Branch Exchange Linux based operating system is required. In this paper, CentOS Linux based operating system is used as it is user friendly. After the installation of CentOS operating system open source package Asterisk is installed. With Asterisk the IP-PBX can be designed according to the requirement of the organization, the required features can be added according to the need. For communication hard phones or softphones installed in computers or laptops can be used. For communication using smartphones an

***Application called CSIPSimple is used.***

Wireless IP-PBX utilizes WIFI technology for communication, the same wireless infrastructure used for your corporate network. Just as we use mobiles and laptops within this wireless infrastructure to gain access to information, now we can use wireless IP phones system as this system uses the telephony function directly into an already existing data network. This provides an advantage that voice and data network can be used together using single system. One of the major advantages of the IP-PBX wireless phone is that you can carry your extension with you inside a wireless networked environment. The wireless IP phone carries your personal extension and is part of your corporate phone system.

In large or small organisations, IP telephony offers many benefits. The major benefit will be productivity. By extending mobile communication in the organisation, the productivity of the users will be increased when they are not at their desk using wireless IP telephony. This improves the business response results by enabling users to answer critical business calls anywhere anytime within the wireless campus.

Another advantage of this system is that it is cost saving. Calling can be done free of cost within the campus. Expanding the communication system is easier and cheap. As it uses wireless infrastructure there is no need of additional hardware or wiring for new user. IP-PBX system is easy to install as compared to proprietary phones. Proprietary phones may be cumbersome and somewhat difficult to install and configure. A computer savvy person can easily install Internet Protocol Private Branch Exchange.

## II. LITRATURE SURVEY

In the past, the goal of telecom service providers was provide better services at some cost. The costs were then being levied on the customer. To this end, only the rich persons could afford these services. Over the years, there have been changes to this situation. The industry provides services at very low cost to the customer. In recent years telecom companies experienced a significant increase in number which have led to high level competition among them. At the same time customers also grown tremendously. Thus, there is need of the better and best management of resources like optimization of the quality of the services that telecom service providers provide to users and carrier customers. Some systems like Whatsapp, Hike, imo which provides low cost communication. For example Hike allows free call. It needs the Internet connection to make the call and we need to pay tariff to the service provider or cellular companies.
For using these services we need to have access to internet connection. It could be costly affair for small organizations. Installation and maintenances of Wired LAN is costly. Comparatively installation of Wireless LAN is easier and simple and also require low maintenance. It is also easy to troubleshoot. So we propose a wireless system for audio call.

The goal behind the system is to enable the free cost voice communication. We have designed a client server model based system. Our server being accessible to administrate it is easy to control over system. N0 need of internet connection for working this system.

## III. PROPSED WORK

The proposed work includes the following.
The application on implementing SIP-based VoIP (Voice over Internet Protocol) application for smartphones such as Android Mobiles.
The purpose of this application is to implement a telephony program that uses Wi-Fi in Point-To-Point or WirelessLAN as a communication between users at no cost. The system will allow users to search for other individual within the Wi-Fi coverage and to establish no cost Point-To-Point connection for voice communication.
Voice over Internet Protocol is used for communication of two persons by sending voice
packets. Various protocol are involved in implementing VoIP.
The main tasks are divided into two parts.
The major task is to establish a session between the two communication users. The protocols involved in establishing the session are called as control plane protocol. Session Initiation Protocol and H.32 are some of the control plane protocols. These protocols are also called as signalling protocols as they are used to establish session between the users.
In this application we implemented SIP as our signalling protocol. The purpose of this product implements a telephony program that uses Wi-Fi in a Point-To-Point or Wireless LAN for communication between the users at free cost. The system will allow users to search for other individuals within Wi-Fi coverage and establish no cost Point-To-Point connection for voice communication.
3.1 Implementation
Initially we need to implement IP-PBX sever. Firstly we have to install the Linux based operating system. In this paper we are using CentOS Linux based operating system which is user friendly and easily to install. Second step is to install Asterisk- the open source package. With Asterisk the IP-PBX can be designed according to the requirement of the organization, the required features can be added according to the need. For installation of Asterisk following package sources are required.

- Asterisk main program
- Zapata Telephony Driver (zaptel)
- PRI libraries (libpri)
- Asterisk sound package

X-lite software is used for communication through laptops and personal computers. X-lite has a dialling pad and all other calling options which are normally found in a phone. After the registration of X-lite peer with the server calling can be done.

### *3.2 SIP Framework*

Session Initiation Protocol (SIP) is used in IP-PBX for setting up calls between the peers. Its purpose is to allow two end points talk to each other but does not deal with media of the call. Figure1 shows sequence of steps followed while making a call between two users. First the users are registered with Asterisk server; once the users are registered call can be placed between the users. While calling an invite request is first sent to asterisks server then via asterisk server it is send to the called party. SIP only takes and makes call while the media session is carried by another protocol RTP. RTP protocol is used to deliver voice.

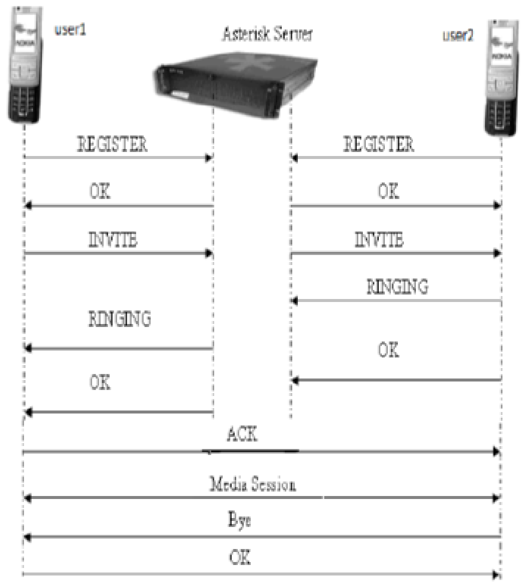


*Figure 1. SIP Calling Sequence*

## IV. CONFIGURATION ON SMART PHONE

In IP-PBX for communication using smartphone CSipSimple application is used. CSipSimple is android application which uses Session Initiation Protocol (SIP). Generally normal calling using smartphone requires SIM card whereas for IP-PBX calling there is no need of SIM card. The only things required are CSipSimple application and Wi-Fi facility in the smartphone. Within the range of Wi-Fi call can be done from anywhere. Figure2 shows demonstration of call using Smartphone.

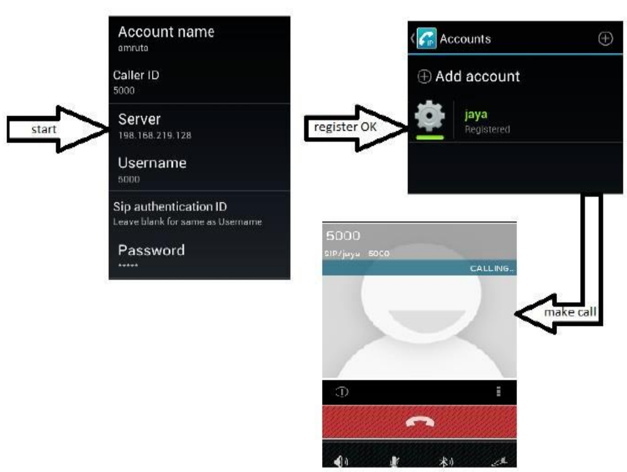


*Figure2. Demonstration of call*

## V. CONCLUSION

This paper describes Internet Protocol Private Branch Exchange with smartphones. This system is cost efficient and reliable as calling is done within the range of Wi-Fi and there is no need of SIM card for calling. Internet Protocol Private Branch Exchange is easy to configure and install only few steps are required for configuration and installation. As far as future work is concerned code for addition features can be designed like call center functions, ring group etc.

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