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FREE COST WIRED INTERCOM CALL COMMUNICATION WITH SMART PHONE USING Wi-Fi

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ABSTRACT:

Unified communication is the latest research topic and many organizations are working on it in all over the world. cellular and wireless communication are the now a days improvement more then applications used smart mobile. The intercom network is the telecommunication phone network already wired networks free cost. This mobile needed for using one mobile to other mobile voice call in through sim.it has GSMsim are signals are cost by the service provider. This project is main concept free costwired intercom network to call communication with smart phone using Wi-Fi signals.

An IP-PBX System is a complete telephony system that provide free of cost, without SIM card wireless calling. This project defines the structure and functions of Elastix it implemented the function of VOIP (voice over internet protocol) like voice call. This project provides great portability, flexibility&cost effective solution to organization. The different types of communication device like android, iptelephone,laptop& intercom connected to Wi-Fi server.

KEYWORDS:*IP-PBX,VoIP,ElastixSW, Intrcom,Wi-Fi server.*

INTRODUCTION

Communication is the essential part of our day to day life wireless Technology has improved a lot and hence everyone seeks for convenient and efficient way ofcommunication. In an organization like colleges, corporate offices, communication channels among all employeesplays a vital role. Considering all these facts, it isnecessary to develop such a system which is easy deploy, easy to run, and easy to access it. Manyorganizations are providing Wi-Fi facility in order toprovide internet connectivity. Inthis projectenhances the use of WLAN, offered by Wi-Fi as a medium for voice transmission. Wi-Fi enabled smart phones can be connected to the router and can communicate with each other. This system can prove as the best alternative for existing intercom system. Use of existing resources put no burden on institute as infrastructure cost.[3].

RELATED WORK

1.EPABX:

Traditional PBX system which means that EPABX system (Electronic Private Automatic Branch Exchange).It is a phone line sharing device. The device lets you connect the extension. It works like mini telephone exchange. It comes under the categories of business

phone system which serves as business environment. It is an office equipment of immense use for telephone connectivity with extension of single phone line.

A telephone exchange serving a single organization, having a switch board and associated equipment, usually located on customer premises; provides for switching calls between any two extension served by the exchange or between any extension.
EPABX is used for the internally incoming and outgoing of telephonic call. If we have two telephone line it can be divided into 8 lines and incoming and outgoing call can be done from these 8 number and the same time by having only two telephone line or number. There are some disadvantages EPABX system and these are less security, less flexibility for user, No expanding facility and so on.

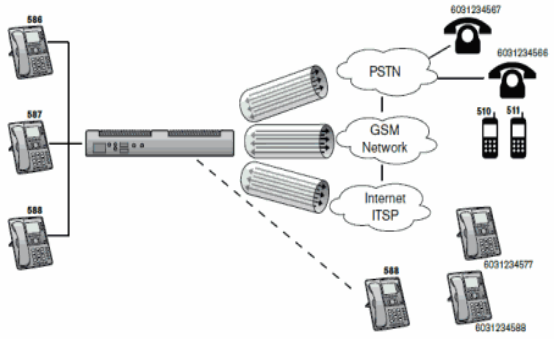


Fig. Block diagram of EPABX

2. Voice Transmission over LAN Using

Bluetooth

Two cell phones, connected via Bluetooth, can exchange mediafiles. Constraint over this is geographical area covered by Bluetooth is limited. Even today, Bluetooth enabled laptops and desktops are in the market, using which we can obtain the wireless communication. Area covered by Bluetooth's range is very small, resulting in limitations over users' mobility. In ad-hoc network, power consumption is major issue.[2].

PROPOSED SYSTEM/WORK

IP-PBX stands for Internet Protocol Private Branch Exchange (telephone switching within an enterprise) that switches call between VOIP (Voice over Internet Protocol) uses on local lines while allowing all users to share a certain number of external phone line. The typical IP-PBX can also which calls between VOIP user and traditional user, or between three traditional telephone user in the same way that conversational PBX does. One of the main advantage of an IP-PBX is the fact that it employees important data and voice network. This means that internet access, as well as VOIP communications and traditional telephone communications, are all possible using a single line to each user, this provide flexibility as an enterprise grows and can also reduce long term equation and maintains cost. IP-PBX system does not required extra hardware and software. It can eliminate phone wiring. This system required only VOIP server And IP Phone.

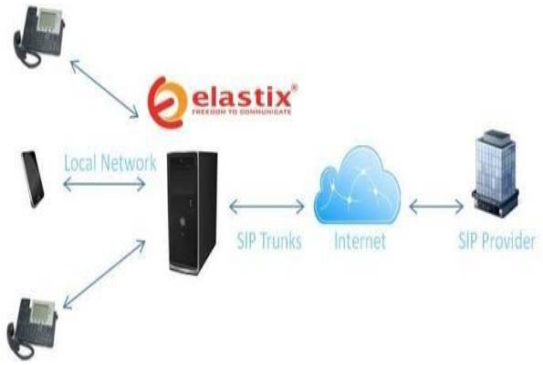


Fig. Block diagram of proposed system

Asterisk™, is a Linux based IPBX application developed by Mark Spencer of Digium™, the company behind Asterisk. Elastix™, evolved from the core Asterisk 1.4. It is made up of several major components. These were developed under GPL supported relatively by users themselves. It consists of applications, a provisioning system, an installer, and an operating system that, together, make a complete package ready for use as an out of the box PBX.

Within this document, Elastix and Asterisk will be referred to frequently and they are interchangeable as Elastix is in essence a superset of the Asterisk.

METHODLOGY

These are the following methods (or) components are used in the project that are given below,

1.**ELASTIX** :Elastix is a collection of "best of breed" Open Source products and tools compiled together to become an integrated IP PBX. Correctly implemented, this system will provide you with a PBX system that will rival almost any other, not just in PBX functions, but ability to integrate with other products to make the system even more powerful.

The major components that make up Elastix are;

• **Asterisk** (currently v1.4), the core PBX (Made by Digium)
• **vTigerCRM**® and **SugarCRM**®, CRM systems
• **A2Billing**® – Calling Card platform and billing application for Asterisk.
• **Flash Operator Panel**, a screen-based operator's console
• **Hylafax**® software based FAX System
• **Openfire**® - Jabber Compliant Server for Instant messaging, presence Management, SIP Phone
• **Conferencing** control application
• **freePBX**® (embedded and standalone) a web User Interface tool for Elastix.
• **A report system** – the part of Elastic (and freePBX) that provides CDR reporting.
• **A Maintenance system**, also part of Elastix, which provides low level interfaces to some components and real time system information
• **OSLEC** - Software Based Echo Cancellation
• **Postfix**®, a well known mail server.
• **Round Cube webmail** – Webmail Interface

• **CentOS**®, a version of Linux related to a very well known Enterprise Linux (but without the branding and support).

Each of these proven products is written and maintained by separate companies or entities, and in many cases small and large companies are using these products in production.

Fig.Elastix IP PBX Server administrator login page

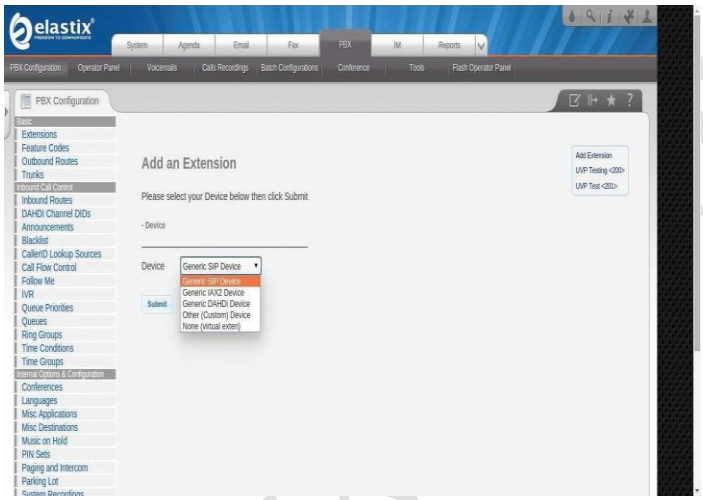


Fig.Elastix IP PBX Server Add Extension page

Elastix developers have written a Web Interface that allows you to access these Programs, so that they, in general, look like one complete product. Elastix have also written certain software such as reporting programs, Hardware detection, Network Configuration, Software Update Module, Backup Restore module, User Management and many more modules themselves.

2.VoIPPhone: The Voice-Over Internet Protocol (VoIP) technology allows the voice information to pass over IP data networks. This technology results in huge savings on the amount of physical resources required to communicate by voice over long distance. It does so by exchanging the information in packets over a data network.

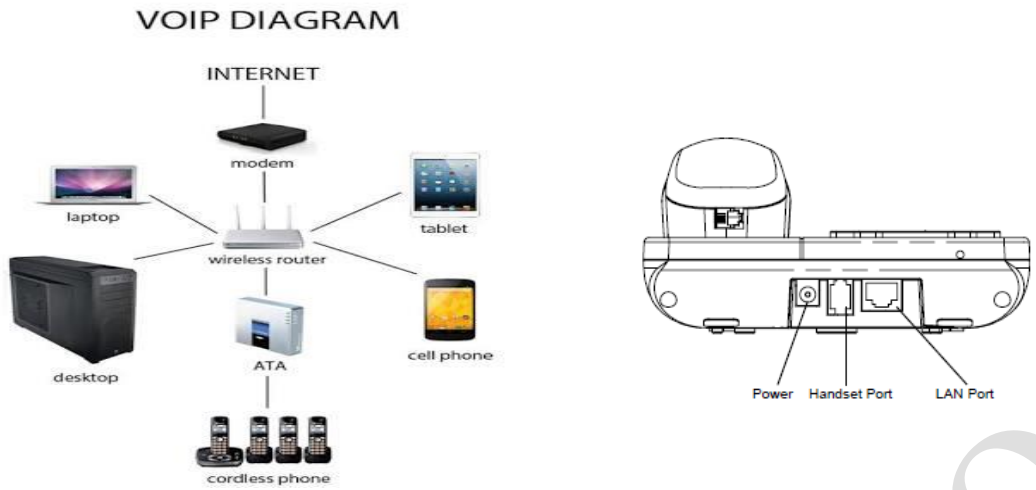


Fig .VoIP PHONE CONFIGURATION

The basic functions performed by a VoIP include – signalling, data basing, call connect and disconnect, and coding/decoding. The steps involved in originating and internet telephone call are the conversion of the analogue voice signal to digital format and compression/translation of the signal into internet protocol (IP) packets for transmission over the internet; the process is reversed at the receiving end. VoIP software's like Vocal TEC or Net 2 Phone are available for the user. With the exception of phone to phone, the user must possess an array of equipment which should at minimum include VoIP software, an internet connection, and a multimedia computer with a sound card, speakers, a microphone and a modem. The VoIP network acts as a gateway to the existing PSTN network. This gateway forms the interface for transportation of the voice content over the IP networks. Gateways are responsible for all call origination, call detection, analogue to digital conversion of voice, and creation of voice packets.**[1].**

3.ATA: Analog Telephone Adapter

An ATA is device for connecting traditional analog telephones, fax machines, and similar customer-premises devices to a digital telephone system or a voice over IP telephone network.

Fig.analog telephone adapter basic setup outline

4.SIP Gateway

It is most likely that you want to communicate with others on the PSTN network, so you need to obtain a VOIP gateway service. Since part of the call is being carried on the circuit

switched network, it costs real money. This means that you will have to pay for this part of the system.

Setting up gateways will be covered in the installation section below.[5].

5.Zoiper android App: Zoiper is the free softphone to make VoIP calls through your PBX or favourite SIP provider. zoiper runs on a multitude of different platforms. No matter if you are using mac,linux or window[6].

Fig. App home page

Fig. App configuration process

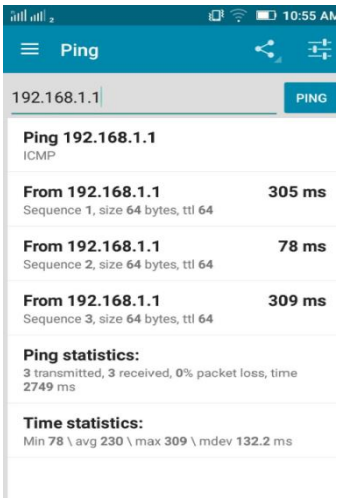
6.Home Network:

The fact that you are reading this and contemplating on installing Elastix, I assume that you have a home network, a broadband service, and are set up behind a Router of some sort.

You will need to pick a static IP address for your IP PBX that is on your home network e.g. 192.168.0.100. To keep it simple, please ensure that the Asterisk box is in the same network segment of your existing LAN.

EXPERIMENTAL RESULTS

Following results are obtained when a call is established between two cell phones and telephones i.e intercom through Wi-Fi



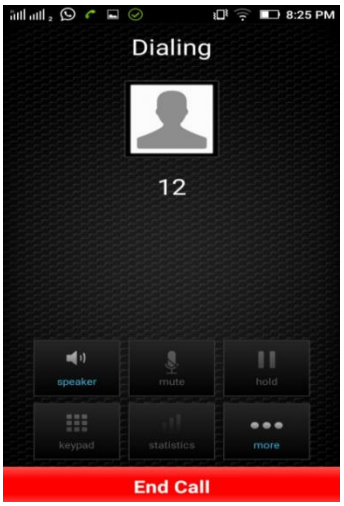


Fig. connectivity condition of android mobile

Fig. outgoing and incoming call through android **ZIOPER App**

Comparison of EPAPX &IPPBX

PBX	IP-PBX
Separate Wiring system	Use LAN
Higher operational cost	Lower operational costs
Difficult to configure	Easy to configure
Complex Management	Simpler Management
Less Features	More Features
Classical and No Future	Future Development
Uses a separate infrastructure	Integrates into Data Network

ADVANDAGES:

- Low cost
- Simple hardware tools
- Simple software codes
- Plug & play
- Easy to operate anyone

APPLICATION:

- Call center
- Star hotels
- Small business telephone systems

- Industries
- Hospitals
- Schools & college

CONCULUSION

IPPBX is revolutionizing the way our telephone system works. Instead of having fixed circuit switches as the medium for voice transmission, which is very costly, digitizing voice and video and using the widespread IP network to transmit the real time data not only works, but is very cost efficient. Since IPPBX is becoming the new age of telephones, it is important to understand how IPPBX works. In a nutshell, there are several processes that take place over a IPPBX phone call.

VoIP technology offers broadband services and the integration of voice and data at all levels. One key factor that is driving the VoIP application development and deployment is reduced voice service charges. In addition to cost advantages, VoIP services have compelling technical advantages over circuit switching. VoIP networks are based more on an open architecture than their circuit-switched contemporaries. This open, standards-based architecture means that VoIP services are more interchangeable and more modular than those offered by a proprietary voice-based PSTN network. Open standards translate into the realization of new services that one can rapidly develop and deploy. Moreover, VoIP is suitable for computer telephony integration and other next generation applications.

FUTURE SCOPE

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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