Advance Ippbx System for Free of Cost Calling in Campus

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Abstract: An IP-PBX System is a complete telephony system that provide free of cost, without sim card wireless calling. In order to make these measurement we have use the open source operating system such as Cent-Os and some packages, by using this operating system and packages we can implement our IP-PBX server. By Implementing this server we can make wireless communication in the range of Wi- Fi. The traditional EAPBX system is overcome by this IP-PBX System. It is advance technology of communication, so for upgrading day to day business communication, this IP-PBX System is work as a breeze.

Keywords: PSTN, EPABX, IP-PBX,VOIP.

I. INTRODUCTION

Traditional PBX system which mean 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.
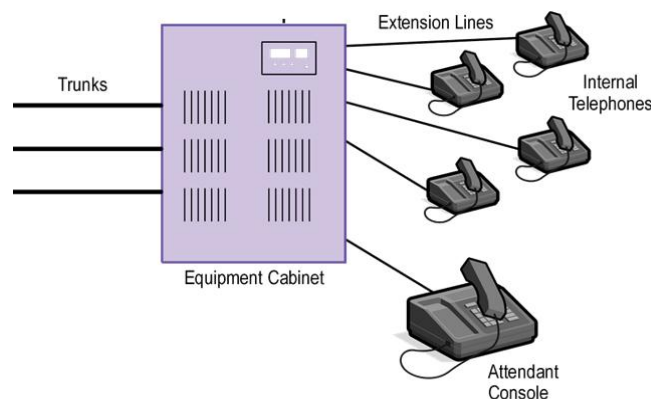

A telephone exchange serving a single organization, having a switch board and associated equipment, usually located on costumer premises; provides for switching calls between any two extension served by the exchange or between any extension.

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.To overcome traditional EPBAX system we implement the IP-PBX system.

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 shared a certain number of external phone line. The typical IP-PBX can also which calls between VOIP user and traditional user, or between three tradional 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.

II. ARCHITECTURE OF IP-PBX SYSTEM

An IP-PBX architecture consist of our implemented IP-PBX server, Wi-fi router because calling is done in the range of Wi-fi and Wi-fi based special series phone. An IP-PBX is delivers dial tone, the ability to conferences, call transfer and most importantly dials other employees by extension numbers. The IP-PBX is software based PBX system that carries out task and services that EPABX server cannot provide like packet switching.
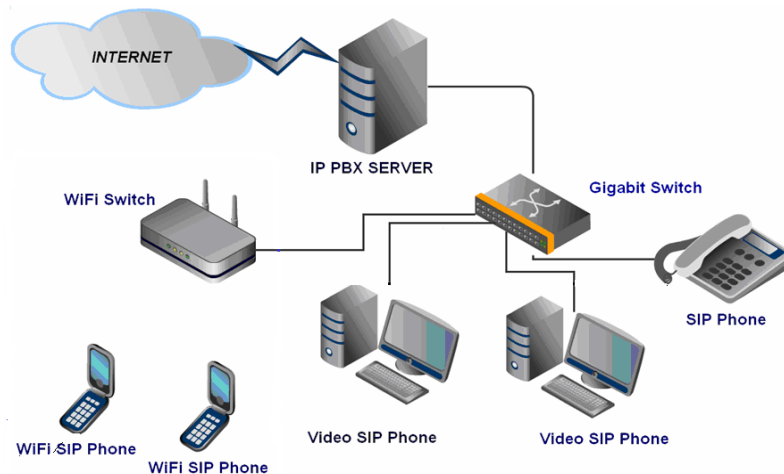
VMware is to create the space for Cent-OS operating system in our computers (server). VMware software provides a completely virtualized set of hardware to the guest operating system. VMware software virtualizes the hardware for a video adapter, a network adapter, and hard disk adapters. The host provides pass-through drivers for guest USB, serial, and parallel devices. In this way, VMware virtual machines become highly portable between computers, because every host looks nearly identical to the guest. In practice, a system administrator can pause operations on a virtual machine guest, move or copy that guest to another physical computer, and there resume execution exactly at the point of suspension. Alternatively, for enterprise servers, a feature called vMotion allows the migration of operational guest virtual machines between similar but separate hardware hosts sharing the same storage. Each of these transitions is completely transparent to any users on the virtual machine at the time it is being migrated.

The server is implemented by the operating system that is Cent-OS, because software that is Asterisk Software, this software are work on only Cent-OS operating system. Cent-OS ("**Community Enterprise Operating System**") is a Linux distribution that attempts to provide a free enterprise class computing platform which has 100% binary compatibility with its upstream source. Originally, Cent-OS Linux (before it was thus named) was a build artifact for cAos Linux. Several of the cAos contributors at the time were really just interested in this build artifact for their own use. Cent-OS is available free of charge. Technical support is primarily provided by the community via official mailing lists, web forums, and chat rooms. Cent-OS version numbers have two parts, a major version and a minor version, which correspond to the major version and update set of Red Hat Enterprise Linux that was used to build that version of Cent-OS. We use Cent-OS 5.4 version for implementation of IPPBX server. During installing the Cent-OS operating system we required some packages such as Zaptel, Asterisk, and Libpri and so on. Without this packages server cannot be proceed, hence, we select these all packages during installation.

III. ASTERISK INFORMATION

Asterisk is technology and protocol Gnostic, which means that you can connect it to the outside world using VOIP or traditional telephony technologies. It also means that you can use virtually any standard based IP phones; Asterisk includes drivers for SIP and other protocols.

we can now install Asterisk:

```
$ cd ~/src/asterisk-complete/asterisk/1.8
$ ./configure
$ make
$ sudo make install
$ sudo make config
```

With the files now installed in their default locations, we need to modify the permissions of the directories and their contents.
Now for zaptel installation

cd /usr/src/zaptel
make config
vi /etc/init.d/zaptel

The new placement of your configuration files is now
/etc/default/zaptel
/etc/zaptel.conf
/etc/asterisk/zapata.conf
Asterisk PBX Configuration :

The following lines need to be added to the Asterisk configuration, in order to acomplish routing of incoming and outgoing calls to and from Topex Mobilink IP.

Outgoing calls:
Edit /etc/asterisk/extensions.conf file
[users]
exten => _70X,1,Dial(SIP/${EXTEN})
exten => _072.,1,Dial(SIP/${EXTEN}@192.168.52.59)
exten => _074.,1,Dial(SIP/${EXTEN}@192.168.52.59)

Save the file and restart asterisk.
All outgoing calls with prefix 072 and 074 will be routed to the Topex Mobilink IP Gateway.
Incoming calls:
Edit /etc/asterisk/extensions.conf file

[Mobilink IP]
exten => _70X,1,Dial(SIP/${EXTEN})

Edit /etc/asterisk/sip.conf file and add the following:

[Mobilink IP]
type=peer
host=192.168.52.59
fromdomain=192.168.52.59
trustrpid=yes
port=5060
disallow=all
allow=ulaw
allow=alaw
canreinvite=no
context=Mobilink IP
dtmfmode=auto

Save the file, restart asterisk. Now you will receive calls from Topex Mobilink IP Gateway.
All incoming calls from mobilink IP Interface mobile1 will be directed to user 701.
All incoming calls from mobilink IP Interface mobile2 will be directed to user 702

For Libpri installation
Check out the latest version of LibPRI and compile it like so:

```
$ cd ~/src/asterisk-complete/
$ mkdir libpri
$ cd libpri/
$ svn co http://svn.asterisk.org/svn/libpri/tags/1.4.<your version number>
$ cd 1.4.<your version number>
$ make
$ sudo make install
```

IV.   IMPLEMENTATION OF ASTERISK

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For main component need to be setup:

- The Asterisk powered IP-PBX.
- The VOIP Gateway services that lets call other VOIP user and people on the PSTN (Public Switching Telephone Network)
- Have a home network and broadband access with a router and hub's/ switches if needed. It does not matter if there is a firewall behind
- There is no requirement to do anything special like a running IPPBX out NAT or DMZ. However, it is important to ensure that there enough bandwidth (upstream & downstream) to carry voice traffic. For this, a broadband connection will be the better choice than a dialup connection

V.   REQUIREMENT NEEDED

The IPPBX system needs a fully functional computer. The minimum requirement will be as follows:

- 250MHz Pentium 11 PC or better.
- 120MB RAM- the more the better.
- 4GB Hard disk space –minimum.
- 10/100 NIC.
- CD-ROM Drive.
- 10/100 4 or 8 port Wi-fi router.

Now we have to create a dialer plan. This dialer plan will be created in file name **Sip.conf.** It is a special file in the Cent-OS operating system.

Involving this dialer plan we also required to design some coding and this create in the file name as extension.conf**.**

In this way the server is ready with dialer plan. In dialer plan, we can increase the no. of extensions as the no. of user increases. These are all about our IPPBX server. To get this IPPBX server in working we required the Wi-fi router. We have to connect this router with sever by LAN cable.

Now for calling we required the Wi-fi based special series phone. With the help of IPPBX server we can't used our phone keypad, for calling. For this we have to installed the mobile application i.e. **CSIP SIMPLE. This application is used for calling.**
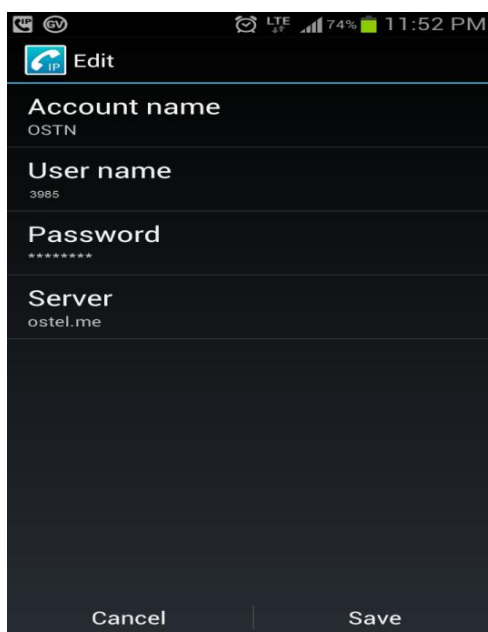
ABOUT CSIP SIMPLE

It is VOIP application for Google Android operating system using SIP. It is an open source and free software released under the GNU General Public License.

SETTING OF SOFTPHONE:

Dialer plan:

CSIP SIMPLE the mobile application is used for calling . From the configuration panel, CSIP SIMPLE was configured as follows:

Account Name: (Name of user)
Caller ID: (Extension no.)
Server: (IP Address of server)
User Name: (Extension no.)
Password: (Created in the server)

VI.   RESULT

In this way, we have to installed the IPPBX server. By implementing this IPPBX server, we can make wireless communication in the range of Wi-Fi. And an IPPBX sever provides free of cost calling without sim card. So, we can say that, it is an advanced technology of communication system.s

VII.   CONLUSION

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.

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