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Asterisk VoIP Private Branch Exchange

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Abstract-This paper intends to gift some vital theoretical and sensible results that we tend to featured throughout setting up a VoIP (Voice over Internet Protocol) server with the well known open supply VoIP server Asterisk. For a totally functional voice exchange we tend to need to set up a server based on Asterisk, connecting shoppers to the server with the help of soft/hard phones then comes the configuration aspects of the soft phones with the server. Here in our implementation we tend to have connected the shoppers to the server with the assistance of SIP protocols.

Keywords-VoIP, Asterisk, PBX, IAX, SIP.

I. INTRODUCTION

The term VoIP stands for voice over internet Protocol. VoIP originated in middle 90's, once hobbyists began to notice the potential of causation voice information packets over the web rather than communication through standard communication systems. The idea is to use the web as a communication network with some additional capabilities. VoIP converts the voice signal from a telephone into a digital signal, sends it through the web, and then converts it back at the opposite end.

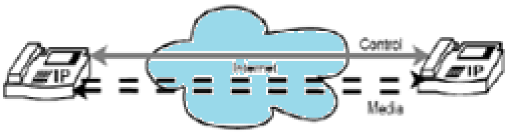


Fig. Setting up of a VoIP call

When we are considering a PSTN line, we tend to generally pay the charges in keeping with the time usages. Additionally we tend to couldn't talk to more than one person at a time. But with VoIP mechanism we will speak all the time, with everybody we wish, as so much as we wish and additionally we will speak with many people at a similar time. If we are still not convinced we will take into accountthat whereas we tend to re talking we will exchange information with the people, we are speaking with.
As we all know that a PBX (Private Branch Exchange) may be a private communication network used among an organization and it handles an organization's voice and data communications. In our experimental setup we tend to tried to establish a PBX system that works with incoming and outgoing voice calls. We tend to used computer to PC communication for simulating the entire task; however IP phones might even be used in the place of PC's. Here in our implementation, even we have not yet done the association of our server with the standard PSTN, though it can be done with the assistance of PCI cards like for instance Digium's TDM400P to validate the association with the existent circuit switched network.

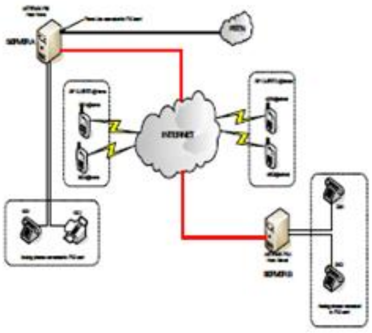


Fig. Experimental setup

To transmit voice as packets over an IP network VoIP uses the internet Protocol. Therefore VoIP may be achieved on any knowledge network that uses IP like the Internet, Intranets and Local area networks (LAN). Here the voice signal is digitized, compressed and converted to IP networks and transmitted over web networks.

II. LITERATURE SURVEY

Asterisk is a complete PBX in software written in C programming language and it runs on Linux operating systems. Asterisk does voice over IP in many protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware like for ex PCI cards. Asterisk in fact creates a PBX that rivals the functionalities of traditional telephone based systems.

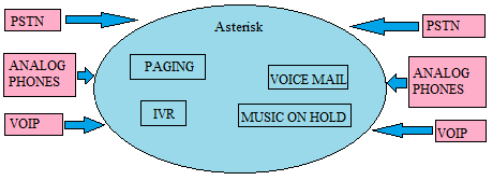


Fig. Model of Asterisk Based Exchange

The benefits associated with an Asterisk based voice exchange could be summarized as:
• Low implementation cost
• Working on TCP/IP protocol
• PBX with enhanced features
• Low Maintenance required
• Convergence of Voice ,Video, Data on a single
Connection
• Easy to add or remove additional extensions.

Asterisk does PBX switching, CODEC translation and various other applications like voicemail, conference bridging, IVR and various others.

1. Architecture of Asterisk Based PBX

Asterisk is successfully designed for maximum flexibility. Unique APIs are defined around a central PBX core system. This advanced core handles the internal interconnection of the PBX, cleanly abstracted from the specific protocols, codecs, and hardware interfaces from the telephony applications. This allows Asterisk to use any suitable hardware and technology available now or in the future to perform its essential functions, connecting hardware and applications. The Asterisk core handles these items internally.

PBX Switching: The essence of Asterisk, of course, is a Personal or Private Branch Exchange Switching system, connecting calls together between various users and automated tasks. The Switching Core transparently connects callers arriving on various hardware and software interfaces. **Application Launcher** launches applications which perform services for uses, such as voicemail, file playback, and directory listing. **Codec Translator** uses codec modules for the encoding and decoding of various audio compression formats used in the telephony industry. A number of codecs are available to suit diverse needs and arrive at the best balance between audio quality and bandwidth usage. **Scheduler and I/O Manager** handles low level task scheduling and system management for optimal performance under all load conditions.

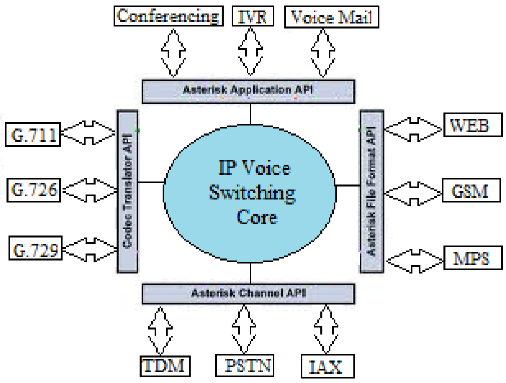


Fig. Asterisk's Architecture

2. Asterisk's Services

VoIP generally uses two types of protocol:1)Signalling Protocols-for setting up a conversation 2) Media transfer protocols for actual transfer of data, once the connection has been set. Session Initiation Protocol (SIP) is an applicationlayercontrol (signalling) protocol for creating, modifying, and terminating sessions with one or more participants. Thesesessions include Internet telephone calls, multimedia distribution, and multimedia.

SIP has the following features:
• Lightweight, in that SIP has only six methods,reducing complexity.
• Transport-independent, because SIP can be used withUDP, TCP, ATM & so on.
• Text-based, allowing for humans to read SIPmessages.Firewalls typically block media packet types such as UDP,though one way around this is to use TCP tunnelling and relays for media in order to provide NAT and firewall traversal.

One solution involves tunnelling the media packets within TCP or HTTP packets to a relay. This solution uses additional functionality in conjunction with SIP, and packages the media packets into a TCP stream which is then sent to the relay. The relay then extracts the packets and sends them on to the other endpoint. If the other endpoint is behind a symmetrical NAT or corporate firewall that does not allow VOIP traffic, the relay would transfer the packets to another tunnel.

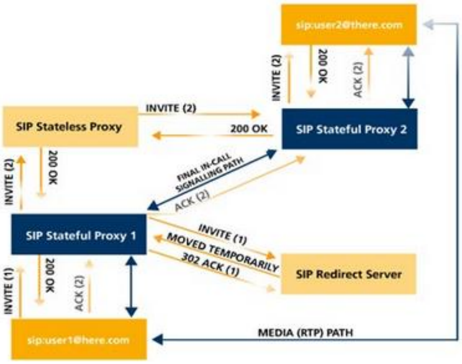


Fig. SIP allowing

III. IMPLEMENTATION DETAILS

For configuring Asterisk as a voice exchange, the administrator must create:

• Dial Plan to make Asterisk respond to users through their devices.

• Devices that allow Asterisk to communicate through a voice path that uses that channel. Asterisk is controlled by editing a series of configuration files. Users connecting to asterisk all belong to a specific context (specified in the channel configuration file), which is where asterisk looks for advice on how to handle the calls placed by that user, checking the access rights to expensive lines, with different rule sets for local users and contacts calling from an outside line.

• /etc/asterisk

Contains all of asterisk configuration files and logic information.

• /usr/lib/asterisk/modules

Contains all of asterisk's loadable modules, operating asterisk functionality.

Applications, channels and resources are located in this directory.

• /var/lib/asterisk/sounds

Contains all of asterisk's sound files for playback and preloaded applications (eg: Voicemail).

• /var/lib/asterisk/agi-bin

Contains all of asterisk's AGI scripts and AGI logic.

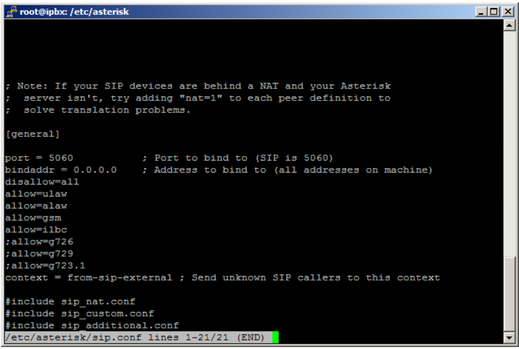
For our experimental setup we configured the SIP and the Extensions at the following:

SIP: /etc/asterisk/sip.conf

Extensions: /etc/asterisk/extensions.conf.

For setting up a client on SIP client on Asterisk we do the following:

```
;[phone1(ale)]
;type=friend
;secret=2222
;auth=md5
;host=dynamic
;reinvite=no
;canreinvite=no
;qualify=1000
;dtmfmode=inband;callerid="ale"<2222>
;disallow=all
;allow=gsm
;context=incoming.
```



```
root@ipbx: /etc/asterisk

; Note: If your SIP devices are behind a NAT and your Asterisk
;  server isn't, try adding "nat=1" to each peer definition to
;  solve translation problems.

[general]

port = 5060           ; Port to bind to (SIP is 5060)
bindaddr = 0.0.0.0    ; Address to bind to (all addresses on machine)
disallow=all
allow=ulaw
allow=alaw
allow=gsm
allow=ilbc
;allow=g726
;allow=g729
;allow=g723.1
context = from-sip-external ; Send unknown SIP callers to this context

#include sip_nat.conf
#include sip_custom.conf
#include sip_additional.conf
/etc/asterisk/sip.conf lines 1-21/21 (END)
```

Fig. The sip.conf file

The other being extensions.conf, where the administrator defines what actions Asterisk will take when calls are answered. A native language is used to define contexts, extensions, and actions. Each context defines where a device starts its dial plan, and therefore restricts what extensions the device may access. Extensions are written within contexts, and consist of numbered lines, each line performing either logic on known variables to the dial plan, or executing one of many applications available in Asterisk.

For editing the Extensions configuration file
/etc/asterisk/extensions.conf
[inbound-from-sip];
Our context for SIP clients
exten => extension no, priority, application (argl,arg2,...)
exten => 1111,1,Dial(SIP/${EXTEN})
exten => 2222,1,Dial(SIP/${EXTEN})
exten => 3333,1,Dial(SIP/${EXTEN})
exten => 4444,1,Dial(SIP/${EXTEN})

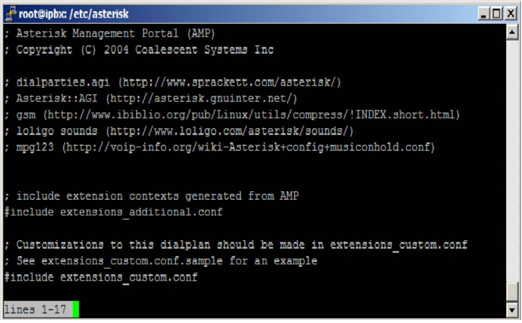


Fig. Extensions.conf file

Once we are done with this, we need to concentrate on the installation and registration of the soft phones that we are going to use at the client end.

Fig.X-litesoft phone at the client's end, along with its configuration

IV. CONCLUSION

We predict that design and implementation shown during this paper are going to be a valuable developing guide for similar operations. Asterisk based voice exchange provides us with a much better alternative solution. It's not only cost effective however provides us with various options that we tend to typically don't get with the traditional circuit switched based mostly PBX .Moreover the system also provides for unlimited expansions and since it runs on a secure software system like UNIX system, it's a lot of less vulnerable to viruses, worms and hackers. As so much as future work is concerned, we would wish to work on connecting our Asterisk PBX with the conventional circuit switched networks with the assistance of PCI cards like considering Digium's TDM400P as an example.

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