SMALL OFFICE PBX USING VOICE OVER INTERNET PROTOCOL (VOIP)

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***Abstract* — This paper is intended to present some important theoretical and experimental result that we have faced during setting up a VOIP (Voice over internet protocol) server with the well known open source VOIP server Asterisk@Home♥. The first part of the paper presents the project objective with some introductory theory about VOIP. The second part contains the implementation of GnuGk PBX and at the end of the paper the Asterisk server setup configuration is added. We used GNUGK for testing H.323 PBX performance and Asterisk@Home for SIP performance testing and found much more stability and accuracy using Asterisk@Home.**

***Keywords* — VOIP , MGCP, UDP, Router mode, PSTN, PBX, H.323, SIP, Asterisk@Home, Gatekeeper, Gateway, OpenH323, RTP/RTCP, H.245, Gnugk-cc, AMP, Trunk**

1. Introduction

We know UDP/IP networks are made of IP packets containing a header to control communication and a payload to transport data. VOIP used it to go across the network and come to destination. The term VOIP stands for 'V'oice 'O'ver 'I'nternet 'P'rotocol. As the term says VOIP tries to let go voice (mainly human) through IP packets and, in definitive through Internet. It allows making telephone calls using a computer data network (UDP/IP). VOIP converts the voice signal from telephone into a digital signal and send it through the Internet then converts it back at the other end.
When we are using PSTN line, we typically pay for time used to a PSTN line manager company, more time we stay at phone and more we have to pay. In addition we couldn't talk with other that one person at a time. In opposite with VOIP mechanism we can talk all the time with every person we want (the needed is that other person is also connected to Internet at the same time), as far as we want (money independent) and, in addition, we can talk with many people at the same time. If we are still not persuaded we can consider that, at the same time, we can exchange data with people are we talking with, sending images, graphs and videos.
As we know about the PBX (Private Branch Exchange) system, a central switching system that handles a firm's voice and digital communication. It is a special purpose computer designed for handling and switching office telephone calls at a company site. It can belong to the company or to a telecommunications provider. We tried to establish a PBX system that works with incoming and out going VOIP calls. Not only voice calls but also it can work with many more VOIP features such as call holding, call waiting, 3 way call conferencing etc.
The IP-PBX may use for a LAN where the outgoing VOIP calls will be send and the incoming calls willbe come through the PBX system. We used only PC to PC communication for simulating the whole task. IP phones can also be in the place of PCs. VOIP uses the Internet Protocol (IP) to transmit voice as packets over an IP network. So VOIP can be achieved on any data network that uses IP, like the Internet, Intranets and Local Area Networks (LAN). Here the voice signal is digitized, compressed and converted to IP packets and then transmitted over the IP network.
Signaling protocols are used to set up and tear down calls, carry information required to locate users and negotiate capabilities. Main signaling protocols used for VOIP communication are:
1. H.323 Signaling Protocol
2. Session Initiation Protocol (SIP)
3. Media Gateway Control Protocol (MGCP)

2. IP-PBX Architecture

The IP-PBX may use for a LAN where the outgoing VOIP calls will be send and the incoming calls will be come through the PBX system. We can use telephone sets also for call receiving and sending also instead of computer terminals. At that time we have set up a gateway. For call routing we set up a gate keeper also that is connected with the private office network. The gate keeper is connected with internet and with gateways for PSTN connection. The gatekeeper here works as a NAT server as well as a call router.
VOIP Gatekeeper- A gatekeeper is an entity on a LAN that provides address translation and control access to the LAN for terminals and gateways. The gatekeeper can provide other services to the terminals and gateways, such as bandwidth management and the location of gateways. A gatekeeper maintains a registry of devices in the multimedia network. ***In the case of setting up a small office IP-PBX's perspective of this project, we can use a gatekeeper in our PBX system to regulate the calls among the private network.***

Our implemented PBX system is as follows:

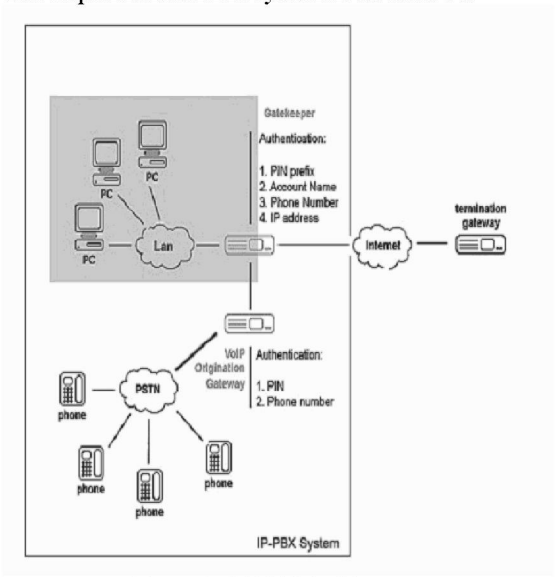

Figure 1 : IP-PBX Architecture

VOIP Gateway- A gateway is either hardware or software that works as an entrance to another network. For example, a gateway is used to connect the legacy PSTN (Public Switched Telephone Network) network to the VOIP (Voice Over IP) network. In 1998, the idea of splitting the gateway into two logical parts was proposed: one part, which contains the call control logic, is called the Media Gateway Controller (MGC) or Call Agent (CA), and the other part, which interfaces with the PSTN, is called the Media Gateway (MG).***In the case of setting up a small office IP-PBX's perspective of this project, we can use a gateway in our PBX system if we want to use telephone sets at each desk of the officials. So, adding a Gateway is optional here.***

3. Implementing GNUGK for testing H.323 PBX performance (OpenH323)

The GNU Gatekeeper is an open-source project that implements an H.323 gatekeeper. A gatekeeper provides call control services to the H.323 endpoints. GNUGK contains such features:

• It is a open source server with much improvement opportunities in future.
• It is also distributed as executables for Linux, Windows, FreeBSD, Solaris and MacOS X as well as with the C source files.
• Can be run as a Windows service
• Accounting and call authorization via SQL database, Radius, file or external application
• Flexible call routing
• Number rewriting (calling and called)
• Support for NAT traversal
• Can act as full H.323 proxy
• TCP interface to applications
• CTI functions (e.g. VOIP call-center, call transfers)
• Gatekeeper clustering support (neighbors, parent/child, alternates)
• H.235 security

GateKeeper Configuration- The following figure illustrates the front end interface of GnuGk.

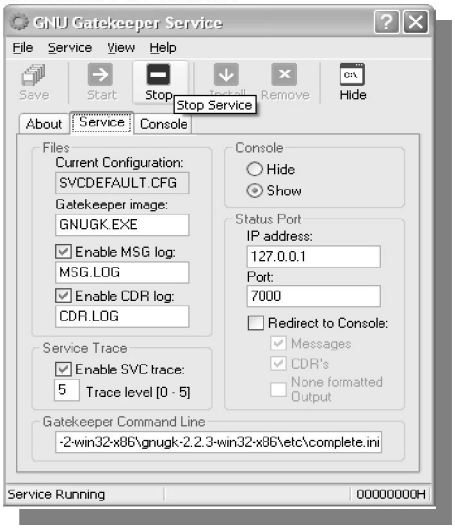

Figure 2 : GnuGk Front end

The field IP Address is the IP address of the host computer or the server. The default port for connecting with GnuGk is 7000. The server keeps a log file of all the services whish stands for MSG.log here. The Start and Stop button uses for server terminating and starting. When server starts, there is a console window opens that is illustrated at the following figure.

The behavior of the gatekeeper is completely determined by the command line options and configuration file. Some command line options may override the setting of the configuration file.

Configuration of gatekeeper.ini file:

This file is created by the user and should be placed at same directory from where the server runs. This file is used to protect the server from unauthorized
users. There should be a password generated by another utility called 'addpasswd'. The file will look
like as follows:

```
[Gatekeeper::Main]
Fourtytwo=42
[GkStatus::Auth]
rule=password
gkadmin=QC7VyAo5jEw=
```

The 'QC7VyAo5jEw=' is an encrypted password with user name.

The file complete.ini contains all available sections for the GnuGk. In most cases it doesn't make sense to use them all at once. The file is just meant as a collection of examples for many settings.

For this project the following lines of the configuration file (etc\complete.ini) has been uncommented.

• Fourtytwo=42
• Name=OpenH323GK
• Home=192.168.1.1
• NetworkInterfaces=192.168.1.1/24, 10.0.0.1/0

• EndpointIDSuffix=_gk1
• TimeToLive=300
• TotalBandwidth=100000.

Call signaling messages may be passed in two ways. The first method is Direct Endpoint Call Signaling, in which case the call signaling messages are passed directly between the endpoints. The second method is Gatekeeper Routed Call Signaling. In this method, the call signaling messages are routed through the gatekeeper between the endpoints. When Gatekeeper Routed call signaling is used, the gatekeeper may choose whether to route the H.245 control channel and logical channels.

Case I: The gatekeeper doesn't route them. The H.245 control channel and logical channels are established directly between the endpoints.

Case II: The H.245 control channel is routed between the endpoints through the gatekeeper, while the logical channels are established directly between the endpoints.

Case III. The gatekeeper routes the H.245 control channel, as well as all logical channels, including RTP/RTCP for audio and video, and T.120 channel for data. In this case, no traffic is passed directly between the endpoints. This is usually called an H.323 Proxy, which can be regarded as an H.323-H.323 gateway.

For this project the gatekeeper routed option is used. The following lines are uncommented for that purpose.

• GKRouted=1
• H245Routed=1
• CallSignalPort=0
• CallSignalHandlerNumber=2
• AcceptNeighborsCalls=1
• AcceptUnregisteredCalls=1
• RemoveCallOnDRQ=0
• DropCallsByReleaseComplete=1

We can also configure the files automatically without remembering the commands using GnuGk-CC. GnuGk Control Center (gnugk-cc) is Graphical User Interface (GUI) for monitoring, management and configuration of GnuGK gatekeeper. Gnugk-cc connects with status port of gnugk and follows the RAS messages and the status of endpoints and calls. It provides an intuitive customer interface for operating with gnugk, without need to remembering and typing commands. The gnugk-cc helps to change all configuration parameters without need to restart the gatekeeper.

Anyone can use telnet to connect with the gatekeeper form his/her desk. The approach will be just to hit the port of the gatekeeper through 7000 port. Another approach to connect and call through gatekeeper is using the Microsoft Net meeting software. The terminals at the local area network each will be equipped with Microsoft Net meeting or Genome meeting. Each terminal should have a microphone and a sound output connected with a headphone or speaker. The endpoints will connect to the server with the Net meeting software and can call any terminal of the network. The gnugk will monitor and establish the connection.

This gatekeeper can only setup calls among the private network. There are no internal call setup facilities yet.

4. Implementing Asterisk@Home for SIP PBX performance

Four main components need to be set up:

•The Asterisk powered IP PBX
•The phones (or softphones) and
•The VOIP gateway service that lets to call other VOIP users and people on the PSTN.
•Have a home network and broadband access with a router and hubs/switches if needed. It does not matter if there is a firewall behind. There is no requirement to do anything special like running the IP PBX outside the NAT or in the DMZ.

However, it is important to ensure that there is enough bandwidth (upstream and downstream) to carry voice traffic. For this, a broad band connection will be the better choice than a dial up connection.

Requirements needed: The IP PBX system needs a fully functional computer. The minimum requirement will be as follows:

•250Mhz Pentium II PC or better.
•128MB RAM – the more the better
•4 GB Hard disk space - minimum.
•10/100 NIC
•CD-ROM Drive
•10/100 4 or 8 ports Ethernet hub/switch (not required if a spare ported router used)

Phones

XLite softphone is used here for testing call diversion here.

SIP Gateway

Since part of the call is being carried on the circuit switched network, it costs real money. This means that someone will have to pay for this part of the system. Setting up gateways will be covered in the installation section below. BroadVoice service is used here for incoming gateway and VoipJet is used for outgoing services.

Office Network

It is rigidly needed that there should be a local area network, a broadband service, set up behind a Gateway Router of some sort (otherwise known as a NAT firewall). There will need to pick a static IP address for the IP PBX that is on the network (e.g. 92.168.0.101).

When the installation is done, it is ready to configure the IPBX system. Asterisk@Home provides a management portal by running up a HTTP server. So to get the management portal, the administrator has to browse from another workstation by http://ipaddress/ (e.g. http:192.168.1.100) to configure Asterisk@Home.

The first thing of this step is to select General Setting and set it up as illustrated below.

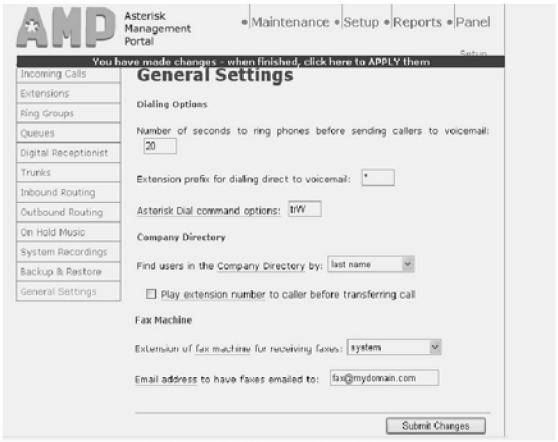

Figure 3 : General Ssettings

Hovering the mouse on the corresponding field description with a yellow/amber underline will display the purpose of the fields. In the Asterisk Dial command option, there is a option to customize the preference to the way asterisk behave (e.g. if someone want the caller to hear music instead of the standard ringing sound, he/she may replace the "r" with an "m"). For further options, just to hover the mouse on the label and the user will be informed of the other options.

After setting up the General Settings, the Submit Changes button and the red bar on top of the screen will help the changes to take effect.

The number of extensions to be set up depends on the administrator. The users can have soft phones installed in 4 or 5 computers or mixture of ATAs and SIP SoftPhones. In this case two extensions to experiment with two softphones were used. The Extension set-up screen between AAH 1.x and 2.x differs in look but the basic concept is similar

Dial Pattern: Dial patterns are part of Outbound Routing. They act like a filter for matching numbers dialed with trunks. The various patterns can enter are similar to Asterisk's definition of them:

•X — Refers to any digit between 0 and 9

•N — Refers to any digit between 2 and 9

•Z — Any digit that is not zero. (E.g. 1 to 9)

• [Various] — Match only one character that matches any of the one in the square brackets. (E.g. [02-68*#] would match 0, any number between 2 and 6 inclusive, 8, * and #. Or, another way of saying this would be 'Match* or #, or a number that isn't 1,7 or 9')

• . — Wildcard. Match any number of anything. Must match *something*.

•| — This lets you use a '0 to dial out' (or '9' in the US) by matching anything before the line, but not sending it to the trunk.

Trunk: A trunk is the telephony service line that would be using to make an external call on. A VOIP service provider (VSP) that have signed up with is also a trunk. If anyone have paid for VOIP service to enabled to make calls out through PSTN, he/she can use this as a trunk for that purpose otherwise ,only use it for making calls using VOIP between subscribers of the VSP only (unless there are peering arrangements with other VSPs) someone can have several trunks if he/she wants to. To make external, PSTN or VOIP calls; there must have at least one trunk.

Set Up BroadVoice Trunk: "Trunks" are configured from AMP trunks section. "Add SIP Trunk." is selected. There needs to fill out the main items on the SIP/Trunk screens. Here BroadVoice is used for incoming service only, so there is need to administer outgoing trunk information.

```
Outbound caller ID:
7237570239
Max channels: 1
```

The outgoing settings can be left as is, except to fill in the trunk name as BroadVoice. In Incoming Settings, the fields were filled as the following:

User Context: 7327570239
User Details:

```
callerid=7327570239
context=from-pstn
dtmfmode=rfc2283
fromdomain=sip.broadvoice.com
host=sip.broadvoice.com
insecure=very
secret=********
type=user
user=7327570239
username=7327570239
```

The format of the Register String is:
username:password:phone_number @provider_domain
For BroadVoice this is used (For privacy the used ID is not mentioned in this paper)

```
7327570239@sip.broadvoice
.com:**********:732757023
9@sip.bro advoice.com/200
```

Once all of this information is entered, Submit button saves all the data.

Set Up VoipJet Trunk: The VoipJet is used for out going calls. Selected the "Trunks" and then "Add IAX2 Trunk." A blank IAX2 trunk form is viewed.

Filling are as follows:

```
Outbound Caller ID:
7327570239
Maximum channels: 1
```

The outgoing dial rules control how numbers are processed before they are sent to the trunk. In the case of VoipJet, phone numbers must be in the form "1+Area Code+Local Number". If the number appears to have an area code but not a "1", then it is added. If it does not have an area code, then "1732" is added (I am in area code 732). So the dial rules are:

```
1+NXXNXXXXXX
1732+NXXXXXX
```

Outgoing settings:

```
Trunk name:: voipjet
Peer Details:
auth=md5
secret=**********
(get this from voipjet
config info)
host=216.118.117.46
(get this from voipjet
config info)
type=peer
username=***** (get
this from voipjet
config info)
notransfer=yes
context=default
```

"Incoming Settings" and the "Register String" kept blank.

Setting up softphone: XLite is used to call here. From the configuration panel, XLite was configuredasfollows

User name: 200
Authorization user: 200
Password: abc1
Domain/Realm: 109.254.17.0
SIP Proxy: 109.254.17.0

Editing the .conf files: A number of .conf files may require editing to get asterisk to work, depending on the individual requirements. In this section, as a base indication.conf, enum.conf, extension.conf, file is needed to be changed.
Indications.conf: In the general section ensure the following exist:[general] country = bd ; (The default is country = us, therefore replaced "us" with "bd" for Bangladesh)
enum.conf: [general] ; ; The search list for domains may be customized. ; Domains are searched in the order they are listed here. ; search => e164.org search => e164.arpa
extensions.conf: There is a change in the extensions.conf. This can be done through the AMP Config Edit. Locating the Inbound Contexts [from] in the extensions.conf file and edited the Lines to the following:
;##############################
###; Inbound Contexts [from];
##############################
##
[from-sip-external] include => from-pstn ;
;give external sip users congestion and hangup.Asterisk@Home Without Tears Page 42 of 137
;exten => _.,1,AbsoluteTimeout(15) ;exten => _.,2,Congestion ;exten => _.,3,Hangup
This will allow external callers to come in and not get the congested signal.

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