# Elastix without Tears

The Elastix IPBX Distribution Development

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# **Without Tears**

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For Elastix<sup>®</sup> v1.x-x with unembedded freePBX<sup>®</sup> 2.x

#### (Document Revision 1.6) ALWAYS A DRAFT (This is a live document and constantly under construction)

by

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This document is released under the <u>Creative Commons By-Attrib Non-Commercial Share-Alike 2.1</u> licence. This work is dedicated to my wife, who has been very patient with me, The Elastix Development Team and Rob Thomas (you can find him on facebook) and his freePBX Team whose devotions to freePBX made it all possible.

My gratitude also goes to Bob Fryer of <u>VOICEINTEGRITY</u> who provided me with many valuable materials for this document and hardware to play with.



Stay tuned for our next episode – Elastix version X

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My heartful thanks to the following sites, without which I would have not been able to distribute this document.

If integrity is what you want, you need **VOICEINTEGRITY** 

If you are stuck and chomping at the bits, <u>MBIT</u> Has the answer

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# **GOING FORWARD**

Following the release of the Elastix<sup>®</sup> Distribution of the ever-popular Asterisk<sup>®</sup> based IPBX, I found it necessary to put together a simple Dumb-Me guide for those deciding to give this new distribution a test run.

As always, this guide is not meant to be a detailed manual; instead it is just a quick and dirty guide and as a resource to assist you get started with minimum pain. Use this document only as a guide, not as a Gospel. It does not cover every detail and every function found in Elastix and neither will it provide you with a complete solution or spiritual guidance.

Only a competent VoIP consultant such as <u>VoiceIntegrity</u> can provide you with the necessary know-how for a complete solution, and as for spiritual guidance, I will leave that entirely to you.

This document was compiled mainly based on Australian condition (because Australia is where I live in – those kind enough to fly me around the world are most welcomed send me an air ticket) through my experience going through the pain and gain, supplemented by inputs from dozens of participants in the various Elastix and Asterisk related forums. These people are mentioned in the bibliography at the end of the document. They have helped me whenever I get into difficulties, and I still seek guidance (the non-divine type) from them.

I take no primary credit for this document (I am only the compiler of the not so divine revelation). I invite others to participate in improving this document. Please send me your feedbacks; so together we can make this document a valuable resource for our peers in the Elastix community, experienced and new alike.

It is also not my intention to violate any copyright or trademarks in preparing this document. I ask those who think that their copyright or trade marks have been violated to inform me so it can be remedied.

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- Ben Sharif November 11, 2007.

The usual disclaimer: (Don't you just hate it, having to say this.)



In its Basic form, all that I have outlined worked for me but it may contain many errors – typographical, omissions and even misguidance that may require a lot of extra corrections. While I have taken due care, however, it may not necessarily be 100% accurate and suitable for your individual configuration. I do not take; neither will I accept any responsibility for any losses incurred due to actions or inaction conducted as a result of methods or advice found in this document.



I am not an expert and therefore I do not have most if not all the answers. I do not provide Elastix support and neither do DIGIUM. If assistance is required, please post in the forum <u>http://www.elastix.org/</u> Participants in the various forums may have a number of solutions and at the same time providing the opportunity for others with similar questions to benefit from the solutions provided. Hep may also be obtained from <u>Australian Elastix Forum at Elastix</u> <u>Connection</u>

# I INTRODUCTION

Asterisk<sup>™</sup> is a Linux based IPBX application developed by Mark Spencer of Digium<sup>™</sup>, the company behind Asterisk. Elastix<sup>™</sup> 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.

# I.I IS VOIP FOR YOU?

Whether VOIP is for you or not depends on a number of or combinations of factors. Some economic and quality requirements should be considered.

#### I.I.I What is it going to cost?

Assuming that you already have a broadband service, a router, and a Windows PC to run the softphone, the cost will be minimal.

If you already have a spare computer to dedicate to this task, then the cost is almost nothing unless you need to buy an audio headset (\$15.00 from your local electronic store) for the softphone. If you do not have a spare PC with the above specification, then you may be able to buy one from your local swap meets for under \$200.00, which may include a monitor. Ensure that the PC has an Ethernet NIC for connecting to your home network.

Your only other initial cost will be the \$20.00 or so activation fee to Oztell (or other VSP of your choice). In fact many VSPs, the like of Pennytel, Primus etc., will register you for free if you want to restrict your calls to VoIP only within the same VSPs. If you want the ability to make to PSTN or between different VSPs, you may have to pay for the calls.

All these "Major Expenses" will be recovered when you receive your monthly Telephony bills from your local Telcos.

#### I.I.II What will the quality of the phone calls be?

If you are expecting the quality to be consistently as good as your existing PSTN calls, there may be room for debate, but if you will be happy with a quality that is comparable to your mobile phone, not quite but very close to your existing PSTN calls and also be prepared for the whining by the other half, the phone calls are not as good and she cant enjoy her mom's singing opn the other end, while you are leaching off the P2P, you might be in luck.

VoIP transport via the Public Internet is very much dependant on a number of factors – available bandwidth not withstanding, your usage habit of the internet and LAN traffic and equipment quality, amongst others, also play very important roles.

For further information please refer to the link below:

http://forums.whirlpool.net.au/index.cfm?a=wiki&tag=VOIP Quality

# I.II WHAT IS 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**<sup>®</sup> and **SugarCRM**<sup>®</sup>, CRM systems
- **A2Billing**<sup>®</sup> Calling Card platform and billing application for Asterisk.
- Flash Operator Panel, a screen-based operator's console
- Hylafax<sup>®</sup> a software based FAX System
- **Openfire**<sup>®</sup> Jabber Compliant Server for Instant messaging, presence management, SIP Phone
- **Conferencing** control application
- **freePBX**<sup>®</sup> (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**<sup>®</sup>, a well known mail server.
- Round Cube webmail Webmail Interface
- **CentOS**<sup>®</sup>, 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.

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.

# I.III THE IP PBX COMPONENTS

Four main components needed to set up VoIP PBX:

- Elastix, the Asterisk powered IP PBX
- The phones (or softphones)
- The VOIP gateway services that let you call other VOIP users and people on the PSTN.
- Have a home network and broadband access with a router and hubs/switches if needed. If you are restricting the usage to PSTN only, you will not need the broadband, but then it will defeat the whole purpose.

I will attempt to describe the set-up of each of these. In most cases it does not matter if you are behind a firewall provided that you have all the appropriate ports forwarded correctly. You are not required to do anything special like running the IP PBX outside the NAT or in the DMZ. However, it is important to ensure that you have enough bandwidth (upstream and downstream) to carry voice traffic.

#### I.III.I The IP PBX

You will need a computer to run the IP PBX. It is recommended that you run a dedicated PC for this purpose. The PC described below (minimum) will be sufficient to power the IP PBX in a small office or home environment. Therefore don't throw away that old Pentium II clunker you have in the attic.

- 500Mhz Pentium II PC or better (PIII 800 will give extra comfort).
- 312MB RAM the more the better
- 8GB Hard disk space minimum (dependant on your usage of MOH, announcements, voice recording etc).
- 10/100 NIC
- CD-ROM Drive
- 10/100 4 or 8 ports Ethernet hub/switch (not required if your router has spare ports)

Naturally if you are running Elastix in a heavy environment, you will need heavier duty and better specification system.



If you are using Telstra Alcatel SpeedTouch 510/530 ADSL Router, you will need to unbind port 5060. To do this, please refer to this link (or see <u>Appendix D</u>). http://www.sipgate.co.uk/faq/index.php?aktion=artikel&type=faq&rubrik=024&id=398&lang=de

When you install Elastix on this "old" computer, it will take it over - it starts by formatting all the hard disks (if you happen to have more than one), so make sure there is nothing on the machine that you want to keep.

#### I.III.II Phones

You can buy SIP phones from Grandstream, SNOM, and Cisco etc. Or you can buy an ATA (Analogue Telephone Adapter) produced by Linksys, Netcomm or Sipura and a few others, which enables you to use an analogue phone (the Linksys PAP2 will allow you to hang two analogue phones to it).

Register with a VSP such as Faktortel, Oztell, Pennytel and several other VOIP Service Providers (VSPs). The choice is yours.

To get started, it is easiest to get a softphone and run it on another computer. See the section on installation of Softphone later in this document.

#### I.III.III 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.

#### I.III.IV 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.

www.hellodigi.ir

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## I.III.V VOIP Service Providers

If you are in Australia, the Whirlpool VOIP forum has a list of VOIP providers in Australia at the following link: <u>http://forums.whirlpool.net.au/forum-replies.cfm?t=370536</u>

# **1 INITIAL PREPARATION AND PLANNING**

## **1.1 FAMILIARISATION**

Before setting up your IP PBX, it is recommended that you familiar yourself with the Elastix GUI and make notes of all the default passwords.

#### 1.1.1 Default passwords

Interface	Login	Password
Elastix	admin	palosanto
freePBX	admin	admin
FOP	admin	eLaStIx.2007
Calling Cards (A2Billing)	admin	mypassword
MySQL	mysql –p (from root)	eLaStIx.2007
SugarCRM	admin	password
vTiger	admin	admin
Openfire	admin	Whatever set at
		install

All other Elastix related password can be found in /etc/amportal.conf

## **1.2 MAP YOUR REQUIREMENTS**

Map out what your home (or small office) telephony requirements are.

Compare the requirements with what facility you already have. You may find that you may need to provide extra facilities that you do not already have e.g. Network switches, router, ATAs or IP Phones and needless to say a suitable broadband service.

For the purpose of this guide, I have determined that we will start with the following:

- 2 extensions with voicemail using a Linksys PAP2 ATA
- 1 extension using a Linksys SPA3000 ATA
- 2 extension using Softphone (we will use these 2 extension for initial testing)
- 3 extension using IP Phones
- 2 VoIP service providers
- 2 PSTN trunks for the purpose of this exercise, I will be using a Linksys SPA3000 for one PSTN trunk and a X100P for the other PSTN trunk.
- IVR
- Music on hold
- A number of customised voice prompts and announcements

Once the above are set-up, I will customise it further with the following;

- Time conditions
- Ring groups
- Follow me
- Conference facility
- DISA etc

In this exercise we will cover all of the above (not necessarily in the stated sequence) and we may also look at other possibilities as we go along, if time and/or resources permit.



This is not an industrial grade installation. This installation is for taining purposes only and to assist new hopefull to get started with Elastix and nothing more.

Once you are clear on what you want, you can start the installation in earnest but before that, here is a brief course on Elastix Network design.

# 2 A BRIEF COURSE IN NETWORK DESIGN (Using Elastix)

One of the most common issues that appears on the forums, are issues that users are having with Asterisk based distributions, which includes Elastix.

Elastix have done a great job of improving how this is setup on your system, but it still needs some thought put into the process.

This chapter is by no means the "be all and end all" in Network theory, and in fact many network aficionados would probably cringe with some terminology and explanations in this chapter, but the purpose is for you to gain a basic understanding of what you are typing in as it will and does have an impact on how your system behaves, and how software interacts with your Elastix system.

First of all, when your Elastix box is finally built, it comes up on the network using a DHCP Address. For 99% of users this will be true. You either have a Dedicated DHCP Server on the Network or your Router is acting as a DHCP Server. What this means is that your Elastix system is assigned a dynamic IP address on your Network, which will last about 7-8 Days (normally). After that time period, it will request a new address. In most cases it is allocated the same address, unless you have a busy network, where in some cases it may allocate that address to another device.

So the first thing we need to do is to allocate and set a static IP address for your Elastix System. But before we can do this, we need to understand how your Network is setup, which includes your Domain Name Service (DNS) entries, the domain name of your Elastix System, whether you are going to make your Elastix system available from the Web (e.g. externally connected Phones).

Let's ignore the Elastix system for the time being and assume that you have a basic Network system. You have one or two Windows workstations connected to a router and this router connects to the Internet. This router has probably allocated IP addresses to the workstations, and it would be useful to obtain the information from one of these workstations to allow us to correctly fill in the Elastix Network Details.

On one of the Windows workstations, click on the Green START button, and from the Menu select **RUN.** Now in the box type the letters *cmd* and click on OK.

Run	? 🗙
-	Type the name of a program, folder, document, or Internet resource, and Windows will open it for you.
Open:	Emd 💌
	OK Cancel Browse

Next a black box will come up. Don't worry if the directory is not the same, as long as it comes up with a prompt.

🛯 C:\WINDOWS\system32\cmd.exe	
Z:\>ipconfig /all	
Windows IP Configuration	
Host Name : WSXP1 Primary Dns Suffix : netintegrity.local Node Type : Broadcast IP Routing Enabled : No WINS Proxy Enabled : No DNS Suffix Search List : netintegrity.local	
Ethernet adapter Local Area Connection:	
Connection-specific DNS Suffix .: Description Intel(R) PRO/100 VE Network Connection Physical Address	ti
Z:\>	-

Type *ipconfig /all* and press Enter.

The information that we want here is:

Primary DNS Suffix: IP Address: Subnet Mask: Default Gateway: DNS Servers:

Write this information down, and let's talk about each one and what impact it has when you implement them on your Elastix System.

## 2.1 IP ADDRESS

Basically this address must be unique on your Network. If you put two devices on your Network, that have the same IP address, you are bound to have issues, some of them very hard to diagnose.

One thing to note is that DHCP does not, by default, know what static addresses that you have issued, and it is possible, sometime in the future, that DHCP allocates the same address as your static address to another device, causing untold issues.

For this reason, we need to understand what addresses your router is giving out. If you have Web access to your router, you can look at the DHCP Server setup and find what addresses that it allowed to give out (Usually called a Scope). Quite often, you will find most routers by default are setup to provide either 50 to 90 addresses. On some routers, you can set exact ranges, on others; it just allows you to select a number of addresses, usually starting from the first available number in the range of IP addresses. If you don't have access to your router, then don't panic.

As a general rule of thumb, we normally choose 100 as the IP address for the Elastix System, as it normally avoids IP address clashes, and is a nice round number to remember.

So based on the information that we gained from our IPCONFIG, we are going to set the IP address of our Elastix System to **172.22.22.100** e.g. If your address of workstation is 192.168.1.2 then you would make Elastix 192.168.1.100

Note that the first three blocks of numbers, must match the same address scheme as our workstation, otherwise (unless you know network design well), you will not be able to talk to your Elastix box from your workstation.

# 2.2 SUBNET MASK

Subnet masks are a complete subject on their own, and this is definitely out of the scope of this document. However, again, to talk to our Elastix System, we need to use the same Subnet Mask as our workstation, so the subnet mask for our Elastix System will be **255.255.255.0** 

## 2.3 DEFAULT GATEWAY

The gateway address is important. In most cases, it will be the address of your router. Basically the Gateway address tells the device or system, where to go if it needs to go to an address or location that does not reside on your network. Your router is basically a traffic cop that knows where network traffic needs to go to get somewhere. In the example provided, it shows our Gateway address as **172.22.25**, which is the address we would use on the Elastix system.

# 2.4 DNS SERVERS

Generally good network design dictates that two DNS Server addresses are used. In the IPCONFIG above, you will note two separate addresses. 172.22.22.2 and 172.22.22.5. On the lab network, it contains a Microsoft Server that is running a DNS Server at 172.22.22.2, so we set that as the primary DNS address. The second address is the router (or Gateway), which is 172.22.22.5.

Your system may be different and contain the addresses that have nothing to do with your normal network address range such as 203.21.20.20 & 203.10.1.9. These are external DNS Addresses and are normally supplied by your Internet ISP. Each ISP will have different addresses for their DNS Servers. If you have a workstation or device using a DHCP Address, then as part of that DHCP address, it passes these DNS Addresses down from the router. The router gets these when it gets its IP address from the ISP.

Now here comes the important part, which forms good network design even for the small network:

If you have an internal DNS Server (e.g. Microsoft Server) on your Network, then it should be the first DNS entry. The reason for this is speed. If a program needs to communicate with a device (even a phone), and especially if it is on the Internal Network, why should it go outside of the Network to interrogate a DNS Server that is going to respond generally 50x slower.

If you don't have a second DNS Server on the Internal Network, then set the second address to the router. Many routers can act as a DNS Proxy, meaning that they forward your DNS requests to the DNS Servers that they know about (remember the router received the DNS Server information when it made a successful connection to the ISP).

Now for many of you, you wil not have a DNS server on your local Network, and your first thought would be to use the ISP's two DNS addresses. This will work, but it may still have some issues.

As we are about to set some static addresses on your Elastix system, it means that we are hard coding these addresses and 99% of the time, this will be fine, except when your ISP decides to change their IP addresses of their DNS Server. In general they don't change that often, but for a period of time, there was one very prominent ISP that changed their address several times over a period of several months.

They work on the 99% rule, that everyone is using DHCP addresses from their router, and when they change the addresses of the DNS servers, then your router will get the updates and naturally filter down to the rest of your Network. If you have the two DNS addresses hard coded into your Elastix System, and your ISP changes the available DNS Server addresses, all of a sudden, your Elastix System cannot find the outside world,

even attempting to access devices on this internal network slows down and appears to hang at times.

So the normal recommended solution for a small network with no internal DNS Server is to set the first DNS Address to your router, and the second address to the Primary DNS address of your ISP. In this case, you are actually improving the redundancy on your Network as the Router interrogates two DNS Servers, and the Elastix System has one more DNS Server to try if the first two do not respond. Furthermore, if the ISP changes the DNS Server addresses, your router would know about these new addresses, keeping your system still running well even though a DNS Server change has occurred.

You might also be thinking that all you need is the router address as it has access to two DNS Servers, and you don't need to add the ISP's Primary DNS Server address. The reason why we do this, is that many consumer based routers, are not the most reliable devices, and it has been seen on many occasions, that the DNS proxy function fails (but usually corrected by a reboot), even though the router is still connected to the Internet and everything else seems to function. By using the ISP's Primary DNS Server address, it means that our Elastix System, can still interrogate a DNS Server, directly through the router, even though the DNS proxy capability has failed.

So if we were using the IPCONFIG example above, we would set the DNS Entries in Elastix to

172.22.22.2 (Our internal DNS Server) 172.22.22.5 (Our Router/Gateway)

If you are running a simple network, then you might be setup using

**192.168.1.1** (your Router/Gateway) **203.21.20.20** (your ISP Primary DNS Server)

It might sound like a lot of trouble, but you are talking about a Phone system, which should be reliable. Especially in a business, how hard is it to go to your manager and tell him that the Phone system failed or suffered some issues because the ISP changed their DNS Server address.

# 2.5 PRIMARY DNS SUFFIX

This item is not absolutely critical, but again good Network design dictates that we should be using a DNS name for the Elastix System

As we are already running a domain on our Lab Network, it shows up under IPCONFIG. This allows us to use this domain name for Elastix.

If you are running a simple Network, you will probably see this entry as empty, so what it means is that you can setup a new domain name for your Internal Network.

If we were to use the details as provided by IPCONFIG, then we would setup the Domain Name as **elastix.netintegrity.local** matching the domain name structure already employed on our network. Netintegrity.local is the domain name. The name before it is whatever you wish to call your Elastix system.

If you have no DNS Structure, as mentioned you set your own up. An example might be **elastix.myhouse.local.** You might be wondering, why the use of the word local instead of **.com** or **.net.** Again this is just good practice. There is no root domain for .local and is regularly used for domain names they are mainly used for Internet networks. Microsoft Servers also use this structure for Internal Networks. Agreed, the world does not live by Microsoft's word, but this one is simple and has no real impact.

# **3 INSTALLATION**

A PC is a prerequisite. For stability, I will suggest that Elastix be installed on a dedicated machine. This is what we are setting out to do.

## **3.1 INITIAL INSTALLATION**

The Elastix version that we will be installing is the latest current version available today – Elastix 1.0. You will need to download the ISO from the Elastix website below. *http://www.elastix.org* 

After you have completed the download,

- Burn the ISO image that you have just downloaded to a blank CD.
- Ensure that your PC will boot from the CD. If necessary change the BIOS settings to enabled this.
- \*\*NOTE: This will erase all data on the hard drives of the PC.
   If you have 2 drives, both may be blown away as well beware.
- Boot your Elastix box with the CD in the CD Drive and press enter. After a few seconds, the following screen will appear. You press F2 to see the various options. However, it is not really necessary. Just press [Enter] to start the installation

CUSTOM MADE TELEPHONY
- To install Elastix, press the <b><enter></enter></b> key.
WARNING: This will format your hard drive and destroy all existing data on your computer!!
<ul> <li>This version doesn't support i586 CPU</li> <li>Use the function keys listed below for more information.</li> </ul>
[F1-Main] [F2-Options] [F3-General] [F4-Kernel] [F5-Rescue] boot: _

- Press the Enter key to start the installation.
- After initial system detection, you will be asked to select the type of keyboard that you will be using. Use your cursor keys to navigate the keyboard types and use the Tab key to move the focus to the OK or Back option. Select the appropriate keyboard. Selected us and move the focus to OK and hit [Enter]

Keyboard Typ	
What type of keyboard d	lo you have?
sg-latin1 sk-qwerty slovene sv-latin1 trq ua-utf uk us	#
	Back

• After system hardware probing is completed, you will be asked to select the Time Zone you are in.

Time Zone Selection	
What time zone are you located in	?
[]] System clock uses UTC	
America/Montevideo America/Montreal America/Montserrat America/Nassau America/New_York	#
OK Back	

- Use your cursor arrow keys to navigate to the appropriate time zone. In our case, we will selected Australia/NSW. Use the TAB key to move the focus to OK and hit [Enter] to accept (see cheat sheet to set NTP time manually).
- Next you will be asked to enter your root password (remember this password).



- After entering and confirming your password, installation will now commence by first formatting the Hard Disk/s.
- From this point it will take about 30-45 minutes for the installation to be complete and ready for the configuration stage.
- During this stage, you will see screens similar to the following. Linux and the required files are being installed. All you do is, wait for it to finish.

Name : filesyste Size : 460k Summary: The basic	Package Installat em-2.3.0-1-i386 : directory layout	for a Lind	ıx system.
	15%		
	Packages	Bytes	Time
Total :	355	1158M	0:17:19
Completed:	Б	9M	0:00:08
Remaining:	349	1149M	0:17:10
	0×		

- After Linux is loaded the CD will eject. Take the CD out and wait for the system to reboot.
- When it reboots, you will be presented with the Elastix splash screen.
- After a moment, it will continue and you will see lots of lines of codes. This process will take a while because it is building Asterisk.
- When Asterisk build is complete, it will reboot itself. Make sure that you remove the CD from the CD Drive before the PC reboots itself.

• Once rebooted, you will see the following screen where you can select the Elastix distribution versions. In this case we shall leave it at the default.



 After going through its initial startup script, Elastix is ready for you to configuring and make changes to the system default.

Once Elastix has been installed, you may log in to Elastix if you need to do any command line tasks.

Log in to your new Elastix (user: root, password: The one you gave earlier)





Elastix uses embedded as well as stand-alone freePBX. To take full advantage of freePBX features, you may want to add extra modules that are not included in the embedded freepbx.

# 4 SET-UP ELASTIX

Using your browser, you can connect to *https://ipaddress/* (e.g. https://192.168.1.100) to configure Elastix

You will be presented with the Elastix initial Admin web login screen as illustrated below.

NHONY B
» Welcome to Flastix
Please enter your username and password Username: Password: Submit
Copyright © 2006 by <u>PaloSanto Solutions</u>

#### Initial Admin Web Login Screen v0.90

The initial Username is *admin* and the default password is *palosanto*. You can change this password later on using that User Management facility provided by Elastix.



To log in to Elastix Web GUI, use user: admin, password: palosanto unless you have changed the user admin password, whereas the default user/password for unembedded FreePBX is admin/admin.

Once logged in, the System Status screen will appear. This screen is the control centre.

OUBTOM MADE TEL	System PBX	Fax Email	IM Reports	Extras					About	Elastix	<u>J.9.0-14</u>		Log
em Info Network	User Management Load M	1odule Shutdo	own Hardware (	Detection Upd	ates B	ackup/R	estore	Prefere	ences				2
System Resou	rces												
U Info:	GenuineIntel Intel(R) Per	ntium(R) 4 CPU 2	.00GHz			Si	multar	neous c	alls,	memo	ry and	CPU	
time:	1 hour(s), 01 minute(s)			1.0							10 15	3 200	
U usage:	2.16% u	sed of 2,003.56	MHz	0.5							10	180	Sim. calls
mory usage:	42.93%	used of 471.77 N	4b	0.0 E							<u>↓</u> 5	170 160	— Mem. usage (MB)
/ap usage:	0.00% u	sed of 760.88 Mt	5	10:45	13:31	19:05	21:51	00:38	06:11	8:58	11:45		
Hard Drives													
Partition Name:	HDA2							Di	sk us	age			
Capacity:	36.27GB						/		5%				Used space Free space
Usage:	5%					(						-	
Mount point:	/							_					
							95%						

#### The Admin Dashboard – The Elastix control centre

**Elastix Without Tears** 

At a glance, the system status screen tells you the existing resources and Elastix's activities.

As it stands, Elastix need to be configured before any of the facilities offered are operable by users. In this guide, we will only be dealing with the basic essentials to get Elastix working.

On the top of the screen, you will notice several tabs arranged by groups (the top most tabs) followed by the sub tabs (the line of tabs underneath). Currently we are looking at the System Info tab under the System group tab.

Under the system group tab are sub-tabs that allow us to conduct system type tasks such as changing Network settings, detecting new hardware etc. The functions of the tabs are self-explanatory.

## 4.1 CHANGE IP ADDRESS TO STATIC INSTEAD OF DHCP

We will start with the network tab because we need to change the DHCP address to a static address.

The network parameters are:

Host	Your Server name, eg: mypbx.netvoice.com
Default Gateway	IP address of your gateway – usually your router (eg. 192.168.1.1)
Primary DNS	The IP address of your Primary Domain Name Server.
Secondary DNS	The IP address of your Secondary Domain Name Server.

	System	РВХ	Fax	Email	IM	Reports	Extras			About Elastix 0.9.2-4	Logout
System Info Network User Ma	nagement	Load Mo	odule	Shutdo	wn	Hardware [	etection	Updates	Backup/Restore	Preferences	0 0
🚸 System Resources											

Selecting the Network Tab will display the following screen where you should select the network interface.

<b>0</b> <u>e</u>			item PBX Fax	Emall IM Reports	Extras	About Elastix 0.9.2-4	Logout
System Info	Network	User Manageme	ent Load Module	Shutdown Hardware	Detection Updates Backup/	Restore Preferences	0 0
Netw	rork Para	meters					
Edit Netwo	rk Paramet	ers					* Required field
Host (Ex. host.examp	ple.com): *	elastix.exan	nple.com		Primary DNS: *	192.168.1.10	
Default Gat	eway: *	192.168.1.1			Secondary DNS:		
Ka Ethe	rnet Inte	rfaces List				147 Start - Previous	(1 - 1 of 1) Next > End >>
Device	Туре	IP	Mask	MAC Address	HW Info		Status
Ethernet 0	DHCP	192.168.1.37	255.255.255.0	00:04:E2:24:5E:59	ADMtek Comet rev 17 at d8838	8000, 00:04:E2:24:5E:59, IRQ 11.	Connected
×						14 Start / Previous	(1 - 1 of 1) Next 🗅 End 🕅
				Elastix is licensed under	r <u>GPL</u> by <u>PaloSanto Solutions</u> . 2006	6, 2007.	

The following screen will appear.

elast	Suctom DBX FA	y Email IN	f Reports Extras			About Elastix 0.9.2-4	Logout
System Info Notwork	User Management Load Modu	e Shutdown	Hardware Detection	Updates	Backup/Restore	Preferences	0 0
📑 Edit Interface	"Ethernet 0"						
Apply changes Can	cel						<ul> <li>Required field</li> </ul>
Interface Type	O Static O DHCP						
IP Address: *	192.168.1.37						
Network Mask: *	255.255.255.0						
		Elastix is lic	censed under <u>GPL</u> by <u>Pa</u>	oSanto Soli	utions. 2006, 2007.		

This is where you will make the necessary changes.

- Change the IP address to the one you want to use
- Click on the Static Radio Button
- Click on apply changes

Click yes when the "Are you sure you want to apply changes" dialog box appears and you are done.

Next we will go to the User Management Tab and change the default password for the user Admin (the only user at the moment).

## 4.2 CHANGE DEFAULT ADMIN PASSWORD

Select the User Maintenance tab as illustrated below.

Qe	last	IX®	System	РВХ	Fax	Email	IM	Reports	Extras			About Elastix 0.9.0-14	Logout
System Info	Network	User Mana	gement	Load M	odule	Shutdo	wn	Hardware D	Detection	Updates	Backup/Restore	Preferences	0 0
		R	$\setminus$										

Click on the User Management tab and the following screen will appear.

elast	System	PBX Fax	Email IM	Reports Extras			<u>About El</u>	astix 0.9.0-14	Logout
System Info Network	User Management	Load Module	Shutdown	Hardware Detection	Updates	Backup/Restore	Preferences		0 (
Jsers	🙉 User Li	ist							
Group Permission	Create New	User							
	Login	Real	Name		Group			Extension	1 of 1) Next p end p
	admin				Administ	trator		No Extension	
	K								1 of 1) Next 卜 End 🗅
	ì			Elastix is licensed	under <u>GPL</u> b	y <u>PaloSanto Solutio</u>	ns. 2006, 2007.		

Click on the admin link (see illustration above) to open up the admin user detail screen below.

	PHONY	System P	BX Fax	Email I	M Reports	Extras			About Ela	stix 0.9.0-14	Logout
System Info Network	User	Management Loa	d Module	Shutdown	Hardware	Detection	Updates	Backup/Restore	Preferences		0 (
Jsers		🙉 View User									
roup Permission		Edit Delete									* Required field
	1	Login: *		admin				Name (E:	k. John Doe): *		
	/	Password: *		****				Retype p	assword: *	****	
		Group: *		Administ	rator			Extensio	n: *		
					Elastix	is licensed u	under <u>GPL</u> b	y PaloSanto Solution	<u>15</u> . 2006, 2007.		

We need to edit the Admin details by changing the default password and other relevant information. At the very least, the default password needs to be changed especially if there is more than one person having access to the system (which is very likely as anyone who knows the box address can get to the web login screen).

To change the details, click on the Edit tab as illustrated above.

	System PBX Fa:	: Email IM Reports E	dras	<u>About Elasi</u>	tix 0.9.0-14	Logout
System Info Network	User Management Load Module	Shutdown Hardware Dete	ection Updates Ba	ackup/Restore Preferences		0 0
Users	🔗 Edit User "admin"					
Group Permission	Apply changes Cancel					* Required field
	Login: *	admin		Name (Ex. John Doe): *	Ben Sharif	
	Password: *	*****		Retype password: *	******	
	Group: *	Administrator		Extension: *	<b>×</b>	
		Elastix is lice	ensed under <u>GPL</u> by <u>P</u> 3	aloSanto Solutions. 2006, 2007.		

In the ensuing screen, enter the name of the admin and change the password followed by a retype of the password in the appropriate fields. Since an extension number has not been allocated as yet, simply leave the field blank.

Once done, click on Apply changes.

# 4.3 HARDWARE DETECTION

Our next stop is hardware detection. This is important if you are using interface cards such as the A100P, X100P, A400P, A1200P, TDM400 etc. As per the illustration below, you need to click on the Hardware Detection tab.

	X®	DRY	Eav	Email	TM	Reports	Extear			About Elastix 0.9.0-14	Logout
System Info Network	User Management	Load M	odule	Shutdo	wn	Hardware [	Detection	Updates	Backup/Restore	Preferences	0 0
Users	User I	ist				1					
Group Permission	Create No.	utteen l			/						

When the following screen appears, click on the Detect New Hardware button. In 99% of the time, your interface cards will be detected. This is the part I love most about Elastix. In my case, my A100P clone card was detected straight away.

	System	PBX Fax	Email IM	Reports	Extras			About Elastix 0.9.0-14	Logout
System Info Network User Ma	nagement L	.oad Module	Shutdown	Hardware	Detection	Updates	Backup/Restore	Preferences	0 0
Hardware Detection									
Detect New Hardware									
PCI Slot # 1: WCFXO/0 "Ge	neric Clone	Board 1"							
Port # : FXO	ſ								

All I have to do is add the following 2 lines at the end of zapata.conf

#include zapata-channels.conf
#include zapata\_additional.conf

(See <u>Get Under The Bonnet</u> to see how the config file can be edited).

**Restart Asterisk** 

And my A100P is ready for use.

Once done, we are now ready to configure the PBX part of Elastix.

# **5 CONFIGURE ELASTIX PBX**

To start configuring the PBX portion of Elastix, select and click on the PBX group tab and we will be presented with the PBX Configuration screen featuring embedded freePBX.

	System PBX Fax Email IM Reports Extras	About Elastix 0.9.0-14 Logout
PBX Configuration Asterisk	Cli Flash Operator Panel Voicemails Monitoring File Editor Echo Cancelle	er 📀 📀
Option Unembedded freePBX	Add an Extension	Add Extension
Basic	Please select your Device below then click Submit	
Extensions	15	
Feature Codes	Device	
General Settings		
Outbound Routes		
Trunks	Device Generic STP Device	
Inbound Call Control		
Inbound Routes		
Announcements	Submit	
Follow Me	Sabilite	
IVR		
Misc Destinations		

From here, those who are familiar with freePBX will have no difficulties in configuring the PBX portion of Elastix. The choice to do the task using the embedded freePBX or the unembedded version is totally on the users. The functionalities are similar either way. Naturally not all the modules available for freePBX are available in the embedded version but what were included were the ones that are considered essential.

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To log in to the Unembedded freePBX, use **user**: admin, **password**: admin. Why we need a login and password is beyond me.

All of the modules necessary to get Elastix operational are already pre-installed and ready for the individual administrator to start setting-up. However there are other optional modules that can be installed if necessary.

For this installation, we will simply use the standard default modules using embedded freePBX.

# 5.1 GENERAL SETTINGS

The first thing you will need to do is select General Setting.

	About Elastix 0.9.0-14     System PBX Fax Email IM Reports Extras	Logout
PBX Configuration Asterisk-C	II Flash Operator Panel Voicemails Monitoring File Editor Echo Canceller	0 0
Option Unembedded freePBX	Add an Extension	Add Extension
Basic Extensions	Please select your Device below then click Submit	
Feature Codes	Device	
General Settings		
Trunks	Device Generic SIP Device	

When presented with the General Setting screen, for the moment, it is wise to leave the default setting as provided except of a few of the fields that needs changing to suit local conditions.

Those fields are as illustrated below.

International Settings
Country Indications Australia
24-hour format 📧
Security Settings
Allow Anonymous Inbound SIP Calls? yes 💌
Online Updates
Check for Updates Update Email bensharif@gmail.com
Submit Changes

#### Country Indications: Australia

Allow Anonymous Inbound SIP Calls? Yes (if this is set to 'No', all inbound unidentified SIP calls will not be accepted).

It is self-explanatory so I will not try to explain, as it is quite minimal and nothing substantial that warrants explanation.

Apart from what I have indicated above, you may need to customised it to your own specific needs such as:

#### Asterisk Outbound Dial command option:

"r" which generate the ring when you dial out, or "m" if you want music instead. There are other options of course – refer to the sub-chapter on *Dial Command Options*.

However, for now, I recommend that we leave them at the default values.

Generally the default values are sufficient to get you started. These values can be changed later to suit your purposes. However the value I have as example above is required to ensure that Elastix will function as expected – at the very least, during the testing stages.

Naturally, help is available. Hovering your mouse on the corresponding field description with a dotted underline (e.g. Asterisk Outbound Dial Command options) will display the purpose of the fields and the various switches related to it.

PBX Configuration       Asterisk: Gli       Flash Operator Panel       Voicemails       Honitoring       File Editor       Echo Canceller         Option       Unembedded freePBX       Basic       Dialing Options         Basic       Asterisk Dial command options: tr       File Editor       File Editor       Image: Command options: tr         Feature Codes       Asterisk Outbound Dial command options: in phones before sending callers to voicemail: 15       Image: Code in phones before sending callers to voicemail: 15       Image: Code in phones before sending callers to voicemail: 15       Image: Code in phones before sending callers to voicemail: 15         Inbound Call Connel by hitting =       Winder the called user to start recording in phones before sending callers to voicemail: 15       Image: Code in phones before sending callers to voicemail: 15         Follow Me       Platering voice this cording in message type: Unevailable       Image: Code in phones before sending callers in phones in phones in phones before sending callers in phones in pho	ela	System PBX	Fax Email IM Reports Extras	About Elastix 0.9.0-14	Logout
Depton     Dialing Options       Basic     Asterisk Dial command options: tr       Extensions     Asterisk Dubbound Dial command options: tr       General Setting     Asterisk Outbound Dial command options: tr       Outbound Rou     tr Allow the called user to transfer the call by       Trunks     Tr. Allow the called user to transfer the call by       Inbound Call Control by htting *     Tr. Allow the called user to transfer the call by       Inbound Call Control by htting *     Tr. Allow the called user to transfer the call by       Inbound Call Control by htting *     Tr. Allow the called user to transfer the call by       Inbound Call Control by htting *     Tr. Allow the called user to transfer the call by       Inbound Call Control by htting *     Tr. Allow the called user to transfer the call by       Indown Call Control by htting *     Tr. Allow the called user to transfer the call by       The present **     Tr. Allow the called user to transfer the call by       The present **     Tr. Allow the called user to transfer the call by       The present **     Tr. Allow the called user to transfer the call by       The present **     Tr. Allow the called user to transfer the call by       The present **     Tr. Allow the called user to transfer the call by       The present **     Tr. Allow the called user to transfer the call by       The present **     Tr. Allow the callew trallow transfer	PBX Configuration	Asterisk-Cli Flash Operator Panel	Voicemails Monitoring File Editor Echo Cancelle	1F.	0 0
Basic     Asterisk Dial command options: br       Extensions     Asterisk Outbound Dial command options: br       General Setting     Asterisk Outbound Dial command options: br       Outbound Rout     t Allow the called user to transfer the call by       Trunks     Trunks     Trunks       Inbound Call Control by bitting =     Trunks       Inbound Call Control by bitting =     Trunks       Inbound Call Control by bitting =     Trunks       Inbound Rout     Vir Allow the called user to start recording fing direct to voicemail: =       Announcement     Vir Allow the calling user to the start recording offer the calling user to the start recording fing direct to voicemail: =       Follow Me     r. You SHOUD hoff use this option on il message type: Unevailable	Option Unembedded fre	eePBX Dialing Options			
Feature Codes     Asterisk Outbound Dial command options:       General Setting     Outbound Rout       Trunks     It Allow the called user to transfer the call by Trunks       Inbound Call Control     by htting *       Trunks     Trunk the to allow the control by htting *       Inbound Call Control     by htting *       Inbound Call Control     by htting *       Inbound Call Control     by htting *       Indownd Call Control     by htting *       Indownd Call Control     by htting *       Indownd Call Control     by htting *       Ing direct to voicemail:     15       Ing direct to voicemail:     *       Follow Me     r, You SHOULD Hort use this option on ill message type:       IVB     outbound trunks	Basic Extensions	Asterisk Dial comm	and options: tr		
Outbound Roult allow the called user to transfer the call by Trunks       hitting =         Trunks       nitting =         Inbound Control by hitting =       ing phones before sending callers to voicemail: 15         Inbound Routt =       Allow the called user to tast recording after prescription = 1 (Attractic V.2.)         Announcement W:       Allow the called user to tast recording after prescription = 1 (Attractic V.2.)         Follow Me       r, You SHOULD NOT use this option on Il message type: Unavailable	Feature Codes General Setting	Asterisk Outbound	Dial command options:		
	Outbound Rou Trunks Inbound Call Control Inbound Route Announcement Follow Me IVR	b: Allow the called user to transfer the call by hitting # T: Allow the calling user to transfer the call by hitting # w: Allow the called user to start recording after possing "1 (Atarcisk v1.2) W: Allow the calling user to start recording after possing "1 (Atarcisk v1.2) r: You SHOULD NOT use this option on outbound trunks	ring phones before sending callers to voicemail: 15 ling direct to voicemail: * il message type: Unaveilable		

Once done, click on the Submit Change button followed by the "Apply Configuration Changes Here" red bar under the sub-tabs for the changes to take effect.

elastix*	System PBX Fax Email IM Reports Extras	About Elastix 0.9.0-14	Logout
PBX Configuration Asterisk-Cli	Flash Operator Panel Voicemails Monitoring File Editor Echo Canceller		0 0
	Apply Configuration Changes Here		
Option	K		

Theoretically, this should apply the changes but unfortunately, there is a bug that will return the following error (this has since been rectified):

*The requested URL /config.php&handler=reload was not found on this server.* 

To overcome this, exit the General Setting screen and go to Extension screen, then click on the aforesaid red bar (things do go wrong from time to time but that is life in the software world).

#### 5.1.1 Dial Command Options

In the **Asterisk Dial command option**, you may customise your preference to the way asterisk behave e.g. if you want the caller to hear music instead of the standard ringing sound, you may replace the "**r**" with an "**m**" (not particularly recommended). For further options, hover your mouse on the label and you will be informed of the other options.

The following are the dial command options available to you but for now, lets leave it alone:

<b>A 1</b>	
Ontione	
options.	

A(x)	Play an announcement to the called party, using 'x' as the file
С	Reset the CDR for this call
d	Allow the calling user to dial a 1 digit extension while waiting for a call to be answered. Exit to that extension if it exists in the current context, or the context defined in the EXITCONTEXT variable, if it exists.
D([called][:calling])	Send the specified DTMF strings *after* the called party has answered, but before the call gets bridged. The 'called' DTMF string is sent to the called party, and the 'calling' DTMF string is sent to the calling party. Both parameters can be used alone.
f	Force the callerid of the *calling* channel to be set as the extension associated with the channel using a dialplan 'hint'. For example, some PSTNs do not allow CallerID to be set to anything other than the number assigned to the caller.
g	Proceed with dialplan execution at the current extension if the destination channel hangs up
G(context^exten^pri)	If the call is answered, transfer both parties to the specified priority. Optionally, an extension, or extension and context may be specified. Otherwise, the current extension is used.
h	Allow the called party to hang up by sending the '*' DTMF digit.
Н	Allow the calling party to hang up by hitting the '*' DTMF digit.
i	Jump to priority n+101 if all of the requested channels were busy.
L(x[:y][:z])	<ul> <li>Limit the call to x' ms. Play a warning when y' ms are left.</li> <li>Repeat the warning every 'z' ms. The following special variables can be used with this option: <ul> <li>LIMIT_PLAYAUDIO_CALLER yes no (default yes)</li> <li>Play sounds to the caller.</li> </ul> </li> <li>LIMIT_PLAYAUDIO_CALLEE yes no - Play sounds to the callee.</li> <li>LIMIT_TIMEOUT_FILE File to play when time is up.</li> <li>LIMIT_CONNECT_FILE File to play when call begins.</li> </ul>

	I MIT WADNING FILE. File to play as warning if
	<ul> <li>LIMIT_WARNING_FILE File to play as warning if but is defined. The default is to set the time.</li> </ul>
	y is defined. The default is to say the time
	Provide hold music to the colling party until a
m/[class])	requested channel answers. A specific MusicOnHold
m([class])	class can be specified
	Execute the Macro for the *called* channel before connecting to
	the calling channel. Arguments can be specified to the Macro
	using 'A' as a delimiter. The Macro can set the variable
	MACRO RESULT to specify the following actions after the
	Macro is finished executing
	<ul> <li>ABORT Hangup both legs of the call.</li> </ul>
	<ul> <li>CONGESTION Behave as if line congestion was</li> </ul>
	encountered.
M(x[^arq])	<ul> <li>BUSY Behave as if a busy signal was</li> </ul>
	encountered. This will also have the application
	jump to priority n+101 if the 'j' option is set.
	<ul> <li>CONTINUE Hangup the called party and allow the</li> </ul>
	calling party to continue dialplan execution at the
	next priority.
	<ul> <li>GOTO:<context>^<exten>^<priority> - Transfer</priority></exten></context></li> </ul>
	the call to the specified priority. Optionally, an
	extension, or extension and priority can be
	specified.
	This option is a modifier for the screen/privacy mode.
n	It specifies that no introductions are to be saved in the
	priv-callerintros directory.
N	This option is a modifier for the screen/privacy mode. It
	specifies that it callerID is present, do not screen the call.
	specify that the Callerid that was present on the
0	channel. This was the behaviour of Asterisk 1.0 and
	earlier
	This option enables screening mode. This is basically
р	Privacy mode without memory.
	Enable privacy mode. Use 'x' as the family/key in the
P([x])	database if it is provided. The current extension is used
	if a database family/key is not specified.
-	Indicate ringing to the calling party. Pass no audio to
	the calling party until the called channel has answered.
S(x)	Hang up the call after 'x' seconds *after* the called
S(x)	party has answered the call.
t	Allow the called party to transfer the calling party by sending the
-	DTMF sequence defined in features.conf.
Т	Allow the calling party to transfer the called party by sending the
	DINF sequence defined in features.conf.
	Allow the called party to enable recording of the call by sending
w	the DTIME sequence defined for One-touch recording in
	Allow the colling porty to enable recording of the coll by conding
W	Anow the caning party to enable recording of the call by sending the DTME sequence defined for one touch recording in
**	features conf (default *1 in Astarick v1.2)

# 5.2 EXTENSIONS

The number of extensions to be set up depends on you. You can have soft phones installed in 4 or 5 computers or mixture of ATAs and SIP SoftPhones. In this case, we have 8 extensions to be set up using ATA as well as softphone and IP phone as we have planned.

#### It's best to avoid the following extension numbers:

Ĵ	200 300-399 666 70-79 700-799	- - - -	Park Notify Reserved for speed dial (Backward compatible) Reserved for FAX testing Reserved for calls on hold (Definable) Reserved for calls on hold (Definable)
	7777	-	Reserved extension for incoming calls simulation

#### 5.2.1 Create Extensions

To create extensions, type of trunk e.g. SIP, IAX2, ZAP or Custom, is done from the Create Extension menu illustrated below:

	System PBX Fax Email IM Reports Extras	About Elastix 0.9.0-14	Logout
PBX Configuration Asterisk	Cli Flash Operator Panel Voicemails Monitoring File Editor Echo Canceller		0 0
Option Unembedded freePBX	Add an Extension		Add Extension
Basic Extensions	Please select your Device below then click Submit		
Feature Codes General Settings	Device		
Outbound Routes Trunks	Device Generic SIP Device		
Inbound Call Control			
Inbound Routes	<u>`2</u>		
Announcements	Submit		
Follow Me	3		
Outbound Routes Trunks Inbound Call Control Inbound Routes Announcements Follow Me DVR	Device Generic SIP Device 2		

There are a number of device types for the extensions. This we need to determine prior to creating the extensions.

- Generic SIP Device : SIP is the Standard protocol for VoIP handsets and ATA's.
- **Generic IAX2 Device** : IAX is 'Inter Asterisk Protocol', a newer protocol supported by a few devices (eg, PA1688 based phones, and the IAXy ATA).
- **Generic ZAP Device** : ZAP is a hardware device connected to your Asterisk machine eg, X100P TDM400, TE110P etc.
- Other (Custom) Device : Custom is a general term used for any non standard device, such as H323. It can also be used for "mapping" an extension to an external number (eg: to route extension 2010 to 1300151151, we can create a custom extension 2010 and in the "dial" text box enter: Local/1300151151@outbound-allroutes.

We need to create a few SIP extensions. Therefore you should select Generic SIP Device from the device drop-down list then click "Submit".

The illustration below is where you create the extension.

You will notice a few fields that you will need to populate. The important ones are:

• User Extension: 3000 (that's the extension number I gave for reception)

- Secret: 3000 (for simplicity, I used the exten number as the secret password)
- Disabled the FAX extension as there will be no FAX attached to this extension.
- Voicemail is not required.
- No Privacy manager
- Neither do we require Dictation service.

I left the rest of the fields at their default values.

5	System PBX Fa:	K Email IM Re	Extras	
Configuration Asterisk-	Cli Flash Operator Panel	Voicemails Monito	File Editor Echo Canceller	
on	Add SIP Exten	sion		
nembedded freePBX	Add OIF Exten	51011		Add
c rtensions				
ature Codes	Add Extension			
neral Settings				
tbound Routes	User Extension	2000		
inks	Display Name	Reception		
nd Call Control	CID Num Alias			
ound Routes	SIP Alias			
ow Me	Extension Options			
	2			
Destinations	Direct DID			
eues	DID Alert Info			
g Groups	Music on Hold	default 💌		
al Options & Configuration	Outbound CID	B ( 1) 11		
nferences	Call Waiting	Disable V		
c Applications	Emergency CID			
sic on Hold				
Sets	Device Options			
jing and Intercom				
rking Lot	This device uses sip tec	hnology.		
ote Access	secret	2000		
llback	dtmfmode	rfc2833		
	Fax Detection Type Pause after answer	None 💌		
	Privacy			
	Privacy Privacy Manager	No 💌		
	Privacy Privacy Manager Dictation Services	No 💌		
	Privacy Privacy Manager Dictation Services	No V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format	No V Disabled V Ogg Vorbis V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address	No V Disabled V Ogg Vorbis V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Ontions	No V Disabled V Ogg Vorbis V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Options	Ito V Disabled V Ogg Vorbis V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Options	No V Disabled V Ogg Vorbis V		
	Privacy Privacy Manager Dictation Services Dictation Format Email Address Recording Options Record Incoming Record Outgoing	Disbled V Ogg Vorbis V On Demand V On Demand V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Options Record Incoming Record Outgoing	No V Disabled V Ogg Vorbis V On Demand V On Demand V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Options Record Incoming Record Outgoing Voicemail & Directory	No V Disabled V Ogg Vorbis V On Demand V On Demand V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Options Record Incoming Record Outgoing Voicemail & Directory Status	Diabled V Ogg Vorbis V On Demand V On Demand V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Options Record Incoming Record Outgoing Voicemail & Directory Status Voicemail Password	III IIIIIIIIIIIIIIIIIIIIIIIIIIIIIIIIII		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Options Record Incoming Record Outgoing Voicemail & Directory Status Voicemail Address	No V Disabled V Og Vorbis V On Demand V On Demand V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Record Incoming Record Outgoing Voicemail & Directory Status Voicemail Password Email Address Pager Email Address	To V Diabled V Ogg Vorbis V On Demand V On Demand V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Record Incoming Record Outgoing Volcemail & Directory Status Volcemail Address Email Address Email Address Email Address Email Attachment	To V Disabled V Ogg Vorbis V On Demand V On Demand V Disabled V		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Record Incoming Record Outgoing Voicemail & Directory Status Voicemail & Directory Status Voicemail Address Pager Email Address Pager Email Address Pager Email Address Pager Email Address	To V Disabled V Cog Vorbis V On Demand V Disabled V Disabled V Oyes Ono Oyes Ono Oyes Ono		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Record Incoming Record Outgoing Voicemail & Directory Status Voicemail Address Pager Email Address Email Address Email Address Pager Lop Play Envelope Deleta Vmail	No     ▼       Disabled     ▼       Org Vorbin     ▼       On Demand     ▼       Disabled     ▼       Oves     ⊙ no       ○yes     ⊙ no		
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	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Record Incoming Record Outgoing Voicemail & Directory Status Voicemail Address Pager Email Address Pager Email Address Email Attachment Play CID Play Envelope Delete Vmail VM Options VM Options	Ite       Disabled       Ogg Vorbis       On Demand       On Demand <td></td> <td></td>		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Format Email Address Recording Options Record Incoming Record Outgoing Voicemail & Directory Status Voicemail & Directory Status Voicemail Address Pager Email Address Pager Em	To V Disabled V Cgg Vorbis V On Demand V Disabled V Oyes Ono Oyes Ono Oyes Ono Oyes Ono Oyes Ono Oyes Ono		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Service Dictation Format Email Address Record Incoming Record Outgoing Voicemail & Directory Voicemail & Directory Status Voicemail & Directory Status Voicemail Address Pager Email Address Pager Email Address Email Attachment Play CID Play Envelope Delete Vmail VM Options VM Context VmX Locater <sup>***</sup>	Ito       Disabled       Org Vorbis       On Demand       On Demand       On Demand       On Demand       Oyes       Ono       Oyes		
	Privacy Privacy Manager Dictation Services Dictation Service Dictation Service Dictation Format Email Address Record Incoming Record Outgoing Voicemail & Directory Status Voicemail Address Pager Email Address Pager Email Address Email Address Pager CID Play Envelope Delete Vmail VM Options VM Context VmX Locater**	Ito     Ito       Disabled     Ito       Orgg Vorbis     Ito       On Demand     Ito       On Demand     Ito       On Demand     Ito       Orges     On O       Oyes     On O		
	Privacy Privacy Manager Dictation Services Dictation Services Dictation Format Email Address Recording Options Record Incoming Record Outgoing Voicemail & Directory Status Voicemail Address Pager Email Address Email Address Email Address Email Address Email Address Email Address Dayer CID Play Envelope Delete Vmail VM Options VM Context VmX Locater™ Submit	Ito     ▼       Ogg Vorbis     ▼       On Demand     ♥       On Demand     ♥       On Demand     ♥       Oyes     On O       Oyes     ● no		

#### Add Extension Screen

Submit when done and don't forget the red bar to apply the changes.

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On the existing Elastix distribution 1.3, upgrading to freepbx 2.5 will break the Extensions GUI where all the fields will be blank. To fix this problem, Saleh, an Elastix user came up with the fix. <u>See Users' Suggestions</u>.

	System PBX Fax Email IM Reports Extras	About Elastix 0.9.0-14	Logout		
PBX Configuration Asterisk-Cli	Flash Operator Panel Voicemails Monitoring File Editor Echo Canceller		0 0		
Apply Configuration Changes Here					
Option					

Click on the Add Extension button to add more extensions.

We will create 8 extensions;

- 3000 Main Reception
- 3001 Medical Centre Manager
- 3002 Dr. Stan Roberts
- 3003 Dr. Susan Forbes
- 3004 Dr. Francis Wong
- 3005 Administration Office
- 3006 Lab and X-Ray
- 3007 Courtesy Phone.

For simplicity, we will allocate passwords to be the same as the extension numbers. Where Voicemail is enabled, we will allocate the same password as well (but you don't have to. You may also nominate an email address for Voicemail Email Notification – it's up to you). Voicemail will be covered in more detail a little later in this document.

#### 5.2.2 ZAP Extension

Creating a ZAP extension (for those who have FXS modules) is no different from creating a SIP or IAX2 extension except that you have to choose the appropriate technology from the Device dropdown list and select **Generic ZAP device**.

When creating a ZAP extension the following must be done otherwise you run the risk of getting a one way audio only. This also affects any Fax machines attached to the FXS ports; you will not receive faxes at all unless you do as below

ntercom	Device Options		
rdings	This device uses zap technology.		
	channel	з 🔶	<ul> <li>Zaptel Channel of your FXS module</li> </ul>
	context	from-internal	
	immediate	no	
	signalling	fxo_ks	
	echocancel	yes	
	echocancelwhenbridged	no	
	echotraining	←	— Echo training should be left blank
	busydetect	no	
	busycount	7	
	callprogress	no	
	dial	ZAP/3	
	accountcode		
	mailbox	2030@device	
	Custom Context	×	]
As illustrated above, after having created your Zap Extension, go back in and remove anything in echo training and submit

Now you should have audio both ways and faxing inbound and out should work Note: Make sure if you are sending over SIP that the codec is ulaw or alaw or the fax will fail with any compression codec like g729

#### 5.2.3 Extensions Batch (Adding a large amount of Extensions Quickly)

It will be necessary to re-submit all extensions created this way from within Elastix/freePBX to enabled freePBX to correctly register these extensions in extensions\_additional.conf, otherwise you may not be able to ring those extensions.

As part of Elastix 1.0, the developers have added a new tool called Extensions Batch



This tool allows you to use a spreadsheet to setup all your extensions and users on a spreadsheet, and then upload them to Elastix. This will then create all the extensions in FreePBX.

To start and learn how to use this tool, it is recommended that you setup two extensions in FreePBX as shown elsewhere in this guide. Set them up with all the options that you would normally set in an extension.

Next, click on Extensions Batch, and click on the **Download Extensions.** This will download a CSV file down to your local hard drive (you normally can choose where to save it). You can now open/import it with your favourite spreadsheet.

You should see a spreadsheet with headings and hopefully the two extensions that you have already setup like the following

2	A	В	C	D	E	F	G	н	1	1	К	L	М	N
1	Display Nam	User Extension	Direct DID	Call Waiting	Secret	Voicemail Status	Voicemail Password	VM Email Address	VM Pager Emai	VM Options	VM Email Attachmen	VM Play CID	VM Play Envelope	VM Delete Vmail
2	Ext 201	201		DISABLED	201	enabled	201	John@mjdomain.com			no	no	no	no
3	Ext 202	202		DISABLED	202	enabled	202	bil@mgdomain.com			no	no	no	no

Now you have a guide, you can setup the rest of the extensions on the spreadsheet. Using the tools that are in your spreadsheet application, you can copy most of the fields that are common for all users, and with the rest, do a select and drop, and in the case of Excel (and probably others), as you drag and drop, it will increment the numbers for you.

Once you are satisfied, that the content is correct, save the file back out as a CSV file. You can call it whatever you like, maybe extensions\_companyname.csv or

extensions\_15042008.csv. Just make sure that when you save it, that your spreadsheet program is selected to save it as a CSV Text file (.CSV).

Now we need to upload you completed CSV file back to Elastix, so click on the browse button, and the following dialog box will appear

DX Configuration Flash Operator	iystem PBX Fax Email IM Papel Voicemails Monitoring E	Reports	Extras	nfiguration Con	ference Extensions Bate	th Tools	
Extensions Batch		File Upload				2 🛛	
Save		Look in:	i extensionsis	st	V 0 0 P 🖽	-	<ul> <li>Required field</li> </ul>
File (file.csv):*	Browse_		Nome -	mycompany	Size Type 1 KB Microsoft Office I	Date Mo Exc 16/04/20	
The header of the file must be in the first line and it must have at minimu If you load a file with extensions already created, these extensions will b		Documents Desktop					
	Elastix is licer	My Documents					
		My Computer	<			<b>x</b>	
			File name:	extensions_mycomp	any 💌	Open	
		Ma Natural	Files of tuner	Al Files		Cancel	

Hopefully it should default the directory where you saved your extensions to, select the Extensions file, and click on **Open** and you should return to the following screen.

2	System	PBX F	ex Email	IM Reports	Extras		
X Configuration Flash Oper	ator Panel	Voicemails	Monitoring	Echo Canceller	Endpoint Configuration	Conference Extensions Batch To	ols 🔍 🔍 🔍
Extensions Batch							
Save							* Required field
File (file.csv):*	5	Asteriskielas	ixlexte Browse	1		Download Extensions	

As you can see, it has filled in the file name and directory in the input box. Now click on **Save.** 

That's it!!! Now go into Freepbx and check that your extensions are there, which they should be.

On a 20 user system, your time to implement extensions, can come down from 30-40 minutes, to around 5-10 minutes, and the more extensions, the more time you will save

Combined with the Endpoint Manager, this tool can literally save you hours in larger system setups.

Again another great development from the Elastix Development team!!

### 5.2.4 Enable Voicemail

To enable voicemail on an extension simply "enable" it when you create the extensions or it may be added later on when you decide that the extension requires a voicemail. If you require email notification of your voicemail, you may enter your email address in the email address field.

In this case let us enable voicemail on extension 3001 (the Manager's extension).

Go back to the Extension Option of Elastix and click on extension 3001. In the Device Option's Mailbox field, change the **3001@device** to read **3001@default** (this has since been retified). Scroll down to the Voicemail and directory section and do the following:

- **Status**: Enabled (use the drop down selection to select it)
- Voicemail password: 3001 (for simplicity we shall use the extension number)
- Email address: Enter an email address of the person (only if you want email notification)
- Email attachment: yes (mark the radio button)
- Play CID: yes
- Play envelope: yes
- Vm context: default

Status	Enabled	*
Voicemail Password	3001	
Email Address	benshari	f@gmail.com
Pager Email Address		
Email Attachment	⊙yes	Ono
Play CID	⊙yes	⊙no
Play Envelope	⊙ yes	⊙no
Delete Vmail	⊖yes	💿 no
VM Options		
VM Context	default	
VmX Locater™	Disabled	*

Click submit when done followed by the red bar on top of the screen.

# 5.3 FOLLOW ME

After setting up the extensions, you need to decide if you want Elastix to call another prearranged extension, if the extensions called do not answer. This is where we will do it as per the illustration below:

### Select the PBX -> PBX Configuration -> Follow Me.

You will be presented with the following screen:

PBX Configuration Asteris	System PBX Fax Email IM Reports Extras	About Elastix 0.9.0-17 Log	out
	Apply Configuration Changes Here		
Option Unembedded freePBX Basic	Follow Me	Main Reception <3000> (add) Medical Centre Manager <3001>	×
Extensions Feature Codes		(add) Dr Stan Roberts <3002> (edit)	
General Settings Outbound Routes	Choose a user/extension:	Dr Susan Forbes <3003> (add) Dr Francis Wong <3004> (add)	
Trunks Inbound Call Control		Admin Office <3005> (add) Lab and X-Ray <3006> (add)	
Announcements		Courtesy Phone <3007> (add) Staff Room <3009> (add)	
IVR		Pathologist (Kuala Lumpur) <3011> (add)	
Misc Destinations Queues		Dental Technician (Kuala Lumpu <3012> (add)	r)
Ring Groups		Sydney Head Office <3060> (ad	(bt

Select the extensions that you want to define (the extension selection is on the right of the screen). In this case, we will pick the Medical Centre Manager's extension – 'Manager <3001> add'

	System PBX Fax	Email IM Reports Extras	About Elastix 0.9.0-17	Logout
PBX Configuration Asterisk	-Cli Flash Operator Panel V	ocemails Monitoring File Editor Echo Canceller Apply Configuration Changes Here		00
PBX Configuration Asterisk Option Unembedded freePBX Basic Extensions Feature Codes General Settings Outbound Routes Trunks Inbound Routes Announcements Follow Me TVR Misc Destinations Queues Ring Groups Time Conditions Internal Options & Configuration Conferences Misc Applications Music on Hold PIN Sets Paging and Intercom Parking Lot System Recordings Remote Access Callback DISA	Cil Plash Operator Panel V Follow Me: 3007 Edit Extension 3001 Delete Entries Edit Follow Me Disable as Default: Initial Ring Time: Ring Strategy: Alert Info: Confirm Calls: Remote Announce: Too-Late Announce: Follow-Me List: CID Name Prefix: Ring Time (max 60 sec): Announcement: Play Music On Hold? Destination if no answer: O Queues: General 4000 O Ring Groups: ALE Pen- Miss Destinations: O DISA: [patrivaje M O Phonebook Directory: Conferences: General O Time Conditions: Ciec VIR: MainCinit/K M Custom App:	oteemails Hontoring File Editor Echo Canceller Apply Configuration Changes Here  Apply Configuration Changes Here  i  i  i  i  i  i  i  i  i  i  i  i  i	Main Reception <3000> Medical Centre Manager (add) Dr Stan Roberts <3002 Dr Susan Forbes <3003 Dr Francis Wong <3004: Admin Office <3005> (ad Lab and X-Ray <3006> Courtesy Phone <3007> Staff Room <3003> (ad) Pathologist (Kuala Lump <3011> (add) Dental Technician (Kuala <3012> (add) Sydney Head Office <300	(add) <3001> > (edt) > (add) > (add) d) (add) d) Lumpur) 60> (add)
	Submit Changes			
		Elastix is licensed under GPL by PaloSanto Solutions. 20	006, 2007.	

In the screen that follows (see the illustration above), enter the following information.

- **Ring strategy:** ringallv2 (to call the main numbers then the other numbers)
- **Extension List:** 2001 (Manager's extension) followed by Administration's Office and the Reception (The reception will take down messages for the Manager).
- Ring time: 20 seconds
- **Destination if no answer**: Core Hangup

Every time the Manager's extension is called, Asterisk will try to connect to extension 3001 and if no answer, it will call Ext. 3005 and 3000 (Reception). If still no answer, it will simply hang up. During business hours, the reception or the Admin Office will take down messages for the manager or the Doctors.

	System	РВХ	Fax	Email	IM	Reports	Extras	Ab	out Elastix 0.9.0-14	Logout
PBX Configuration Asterisk-Cli	Flash Opera	tor Pane	el Vo	picemails	Ma	onitoring	File Editor	Echo Canceller		0 0
					A	pply Cor	figuration	Changes Here		
Option								×		

Click "submit" followed by clicking the red bar on top of the screen to finalise this selection.

## 5.4 RING GROUPS

A ring group is a group of extensions that will ring when there is an external incoming call. You can even put your Mobile Phone number in the ring group if you want to. The 0400123456# is the Medical Centre's Manager phone (see illustration below). *For mobile phone to work, you must have the appropriate route and trunk set up.* 

You may not want a ring group - it's entirely up to you. If you don't require a ring group, you may ignore this section.

When there is an incoming call to the ring group, the phones nominated in the selected group will ring. You may select different ring group for each of the incoming trunk or you may nominate the same group for all the trunks, in which case you will only need to define only one ring group.

For simplicity, we will only defined 1 ring group for all incoming calls from all trunks – at this stage, let's not get too fancy.

We will create a ring group 11 for this purpose. Let's call it AdminGroup as it rings all the phones related to the Administration of the Medical Centre. A single digit ring group is not recommended.

We have included the Manager's Mobile Phone number in as the last one in the group. If no one answers the call, Asterisk will simply hang up. Since the last number in the group is the Manager's mobile phone, generally, the Mobile phone voicemail will take the message if the phone is not answered.

The ring group screen is illustrated below:

	System PBX Fa	K Email IM Reports Extras	About Elastix 0.9.2-4	Logout
PBX Configuration Asterisk-	Cli Flash Operator Panel	Voicemails Monitoring File Editor Echo (	Canceller	0 0
Option	Din a Casa a d			Add Ring Group
Basic Basic	Ring Group: 1			Ring ALL (10)
Extensions	Delete Group			Admin Group (11)
Feature Codes	Edit Ring Group			
General Settings				
Outbound Routes	Crown Description :	Admin Occurs		
Trunks	Group Description: :	Admin Group		
Tobourd Call Control	Evtoncion list:			
Annound Routes	Extension list.	3005		
Follow Ma		0400123456#		
Follow Me		Class 9 Remain dualization		
Mice Dectinations	CID name prefix:	Clean or Kemove objinates		
Queues	Ring time (may 60 sec):	30		
Ring Groups	Announcement:	None		
Time Conditions	Play Music On Hold?	acc 1 💌		
Internal Options & Configuration	Alert Info:			
Conferences	Confirm Calls:			
Misc Applications	Remote Announce:	Default 💌		
Music on Hold	Too-Late Announce:	Default 💌		
PIN Sets				
Paging and Intercom				
Parking Lot	Destination if no answer:			
Contract Descendance				

Next we will set up a couple of softphones to test what we have done.

# 6 SET UP SOFTPHONES.

There are a number of softphones available for use with Elastix. Some of them are free. Following are 2 examples of the free softphones.

# 6.1 COUNTERPATH X-LITE SOFTPHONE

X-Lite Softphones can be downloaded here:

http://www.counterpath.com/xlitedownload.html





After installing the softphone (assuming that users can install the softphone), we need to configure the softphone.

When you start X-Lite 3 for the first time, you will see the following screen.

Click the Add button.

You will then get the following screen.

This is where you will enter your credential

At the various fields, add the following:

- Display Name: Your Name not Ben Sharif @
- User Name: Your extension number
- **Password:** The password of the extension when you created it in Elastix
- Authorization User name: The same as your User Name or extension
- Domain: Your Elastix IP address

Put a check mark in the Register with domain and receive incoming calls

Make sure the Target Domain Radio Button is marked.

For this purpose, that's all you need to do and Click apply, OK and close at the next screen. You are now ready to use the X-Lite softphone.



perties of Account1		
court Voicemail Topolo	gr Presence Advanced	
User Dotals		Fatore
Display Name	Ben Sharif	Properties
User name	2001	Male Default
Password		
Authorization user name	2001	
Domain	192.168 1.100	
Donain Proxy Pagister with donain an Send outbound vis: O donain O proxy Address O target donain	d receive incoming calls	000
Challing plan	#1\aja.7;match=1;preatrip=2;	The second

Sometime it is necessary to change the Honor setting in X-Lite for it to work correctly. To do this, you would need to go to the advance setting of X-lite.

Those using X-Lite having this problem may have to make the following correction in the advance setting of X-Lite.

How to get to the advance setting?

In X-lite:

Dial \*\*\*7469 (SEND)

This will bring up the advanced settings window

Filter for honor

Double click on the honor entry and change the value to 1

## 6.2 BOL SIPPHONE

I have found the BOL SIPPhone extremely simple to set up for use with Asterisk and it also has a call forward facility that I use from time to time.

You may obtain a copy of the BOL 2000 sipphone from the link below. http://www.bol2000.com/website\_c/download/sip

phone/BOL%20SIPPhone\_EN.msi

SIP: Remote Party: Please press right button of mouse to select a profile. 1 2 3 4 5 1 2 ABC 3 DEF 4 GHI 5 JKL 6 MINO 7 PORS 8 TUV 9 WV72 X 0 X 0 X

SIP Phone DUWERED BY

After downloading and set up you will see the following when it is run:

To configure the softphone, click on the hammer icon and you will see the following.

#### Profile Tab

This is the only screen that is required to be filled in.

These are the only information required: **Account**: <enter the extension number e.g. 3001>

**Password:** 3001 (I use the same number as password for simplicity. I use the same password when I set up the extension in Asterisk).

Preferences	? 🔉	<
Profile Audio & V	/ideo Call Forward Network STUN	
Profile List:	Remove	
Account:	201 Add	
Password:	···· Clear Update	
Domain/Realm:		
Proxy:	192.168.1.101 Port: 5060	
Authentication T	ype: DIGEST 🛛 Transport Type: UDP 💙	
📃 Default Profil	e 🗹 Auto Logon 🗹 Keep Password	
	OK Cancel Apply Help	)

Domain/Realm: <leave it blank> Proxy: Your Asterisk network address e.g. 192.168.1.100 Port: 5060

Check the Auto Login and Keep Password.

Then click OK.

### Audio & Video Tab

Next Click on the Audio & Video Tab to ensure that audio properties set is consistent with the Audio card installed in your PC/Notebook.



The illustration above depicts the sound device installed in my Notebook.

- Click on the Tuning Wizard to tune your audio input and output.
- Check Auto Send Video (if you are using Video). I checked it anyway.
- Check Auto Receive Video (if you are using Video). I checked it anyway.

Click OK

### **Network Tab**

Ensure that your Internet Connection Type is set to LAN.

You may ignore the Information of Network field.

Click OK

Preferences	?×
Profile Audio & Video Call Forward Network STUN	
Internet Connection Type: LAN	
OK Cancel Apply	Help

#### STUN Tab

You may or may not need to use a STUN server. If you need to use it, click on the STUN tab and enter the STUN server you want to use. If you do, a list of publicly available STUN server is listed in the section referring to <u>STUN Servers</u> later in this document.

Preferences ?	×
Profile Audio & Video Call Forward Network STUN	_
STUN Server: Stun.fwdnet.net	
STUN Port: 3478	
Enable STUN	
OK Cancel Apply Help	5

In my case, I do not use STUN and therefore I left that section blank and Enable STUN check box un-ticked.

Click OK to close.

#### **Call Forward**

This is pretty simple to set-up. To forward an unanswered call to this extension, all you need to do is click on the Call Forward tab and enter the telephone number you want to forward your incoming calls to. You have 3 options of call forwarding – Always, On Busy or On No Answer. However this facility is only available if your PC is on and the softphone is active.

Click OK when done.

You might want to set-up a couple of PCs with the softphone after which you may start testing your brand new Phone System by dialling each extension in turn.

If you use one of the softphone and dial **7777**, Asterisk will simulate an incoming call. Make sure that in the general setting you have allowed for anonymous sip calls and you also have an inbound route for "any DID/any CID" otherwise the 7777 will not work.

Once done, you may test your softphone connection to Asterisk.

If everything has been done as explained above, we should be able to make and receive calls between our internal extensions. If not, it is time to re-inspect what we have done above and make the necessary correction before attempting to go any further.

Let's take a break and test the soft phone extensions by making calls to each extension that we have created.

# 7 ELASTIX ENDPOINT MANAGER

One of the new features in Elastix is the Endpoint Manager. This allows you to use Elastix to configure the phone with the correct Extension and the other details needed for the phone to register with Elastix. It only covers the popular phone models, but I am sure more will be added as time goes by. Also it is dependent on the model of phone as not all models support this feature.

It assumes that the phone is set as default (e.g. in the state it was when removed from the box).

Before you use the Endpoint Manager, you should have setup your Extensions in Freepbx, as it will provide these as a dropdown box so that you can assign each Extension to a phone.

Under the PBX tab in Elastix, click on Endpoint Configuration and the following screen will appear.

elas REEDEM TO COM	Syster	n PBX	Fax Email	IM Repor	Extras		About Elas	<u>ix 1.0-14</u>	Logout
PBX Configuration   F	lash Operator Pane	Voicema	ils Monitoring	Echo Cance	ller Endpoint Con	figuration Confer	ence Extensions Batch	Tools	0 0
a Endpoint Co	onfiguration								
Endnoint Scan	192 168 1 0/2								
Enapoine scarr [1	192.100.1.0/2								
								Start / Previou:	) (0 - 0 of 0) Next [> End [)]
Set Ur	nset M	AC Adress	IF	Adress	Vendor	Phone Type	User Exte	nsion	Status
								🗸 Start 🧹 Previou:	; (0 - 0 of 0) Next 🗅 End 🕅
			Elastix	is licensed und	ier <u>GPL</u> by <u>PaloSant</u>	<u>o Solutions</u> . 2006, 2	007.		

You will notice the input box beside the ENDPOINT SCAN button. Here we will enter, the address of our network, however, it differs slightly from what we may have entered into Elastix. The reason for this is that the subnet mask uses a different method of describing it.

If you are running a simple network, you probably have a subnet mask set as **255.255.255.0**. This can also be written as **/24**. For most network technicians, this is common knowledge, and also know how the similar numbers for other subnets and again it requires a subject on its own, so for the time being accept that **255.255.255.0** equals **/24** 

The reason for the change is that Elastix makes use of a product called NMAP which performs a search on your Network, and with the Elastix implementation, it requires the subnet to be entered in the **/24** format.

One other item, if you are not aware off, is that if we are specifying the whole subnet, we use a zero (**0**) at the end of the ip address, e.g. your Elastix Server might be 192.168.1.100, and your address for the entire subnet would be 192.168.1.0.

So, having explained that, we have an Elastix Server at 172.22.22.100, and our subnet is 255.255.255.0, so the information we would add to that box is 172.22.22.0/24.

Once you have entered that, press SCAN and if successful, and you have some compatible phones, you should see the following screen.

6	FREEDON TO D	System	PBX Fax	Email	IM Reports	Extras		BROOM LEADIN		
BX Config	uration	Flash Operator Panel	Voicemails	Monitoring	Echo Canceller	Endpoint Configuration	Conference	Extensions Batch	Tools	
🧶 En	dpoint (	onfiguration								
Endpoi	int Scan	172.22.22.0/2								
								8	Start < Previous (1 - 2 of	2) Next > End
Set	Unset	MAC Adress	IP /	Adress	Vendor		Pho	пе Туре	User Extension	Status
		00:08:02:0D:9E:	19 <u>172</u>	22.22.31	Grandstream / G	randstream Networks	~ <b>1</b>	Inselected 💌	Unselected 💓	Not Set
		00:0E:08:DE:1A:	IC 172	22.22.45	Linksys / Sipura	Technology	6	Inselected 🐱	Unselected 💌	Not Set
									Course of Admitance in . A at	ON MALINE & REAL

This is basically the result of its findings. We had a Grandstream GXP2000 and a Linksys on the Lab Network. We can now auto-configure these. Under the Phone type, drop down the box and it should show the models. Select the correct model. Next select the user extension that this phone will handle.

0	elas	System	PBX Fax Email	IM Reports	Extras		About Ela	stix 1.0-14	Logout
BX Confi	iguration	Flash Operator Panel	Voicemails   Monitoring	Echo Canceller	Endpoint Configuration	Conference	Extensions Bate	h Tools	•
A Er	ndpoint (	Configuration							
Endpo	oint Scan	172.22.22.0/2							
								Previous (1 - 2	of 2) Next In End I
Set.	Unset	MAC Adress	IP Adress	Vendor		Pho	ле Туре	User Extension	Status
		00:08:82:00:05:	172 22 22 31	Grandstream / G	randstream Networks	( ave	2000 0005	Verity <202>	Not Set
		00.00.02.00.02.0	A.C.R.M.M.M.M.C.A		and a solution of the	L devite		A rest of the second	
		00:0E:08:DE:1A:3	C <u>172.22.22.45</u>	Linksys / Sipura	Technology	SPA	942 💌	Bob <201>	Nut Set

If done correctly, you should end up with something similar to above. Once you are happy with the settings click on all the boxes underneath the SET button and click on the SET Button.

If successful, you should end up with the screen similar to below.

10									
Endp	oint Scan	72.22.22.0/2							
								/ Start	Previous (1 - 2 of 2) Next   End
Set	Unset	MAC Adress	IP Adress	Vendor	Phone Type		User Extensio	n	Status
E		00:08:82:0D:9E:79	172.22.22.31	Grandstream / Grandstream Networks	GKP2000		Verity ≪202≥	×.	Configured without incident.
		00:0E:08:DE:1A:3C	172.22.22.45	Linksys / Sipura Technology	SPA942	*	Bob <201>		Configured without incident.

On occasions we have found it necessary to reboot the phones afterwards, but it appears to depend on the state of the phone before the Endpoint Discovery.

If you get another phone, then just run the Endpoint Scan again, and it should show the new phone. Make the changes to the phone type, and extension and click only the box relating to that phone and press SET. As simple as that...

# 8 TRUNKS TO THE OUTSIDE WORLD

Making and receiving calls between internal extensions is one thing, but a telephony installation isn't really an installation if you can't receive and make calls to the outside worlds.

To be able to do these, you will need at least one trunk to the outside world.

This is a simplified guide to give the budding Elastix users some understanding in setting up Trunks.

#### Points to keep in mind.



**Beware!!!** If you are using an ATA, please allow for what the ATA will be sending to asterisk when you are preparing you rules and patterns. If you want asterisk to take care of all the rules, it is best to ensure that your ATA is sending exactly what you dialled to Asterisk – just delete your dialling rule from your ATA and replace it with (\*\*|\*x.)

### 8.1 WHAT IS A DIAL PATTERN?

Dial patterns are part of Outbound Routing. They act like a filter for matching numbers dialled with trunks.

The various patterns you 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') –*Avoid using this notation in your Trunk Dial Rule as it does not seem to work. Use this only in your Outgoing Route Dial Pattern.*
- . 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**.

### 8.2 WHAT IS A TRUNK?

A trunk is the telephony service line that you will be using to make an external call on, e.g. the Telstra telephone line that you have at home is a trunk (or it can be used as a trunk if you wish but not necessarily so). A VOIP service provider (VSP) that you have signed up with is also a trunk.

If you have paid for VOIP service to enabled you to make calls out through PSTN, you can use this as a trunk for that purpose otherwise, you can only use it for making calls using VOIP between subscribers of the VSP only (unless there are peering arrangements with other VSPs).

You can have several trunks if you want to. You can have Telstra and/or Optus telephone lines and use them as trunks, and you can also have a few VSPs that you subscribed to as additional trunks.

To make external, PSTN or VOIP calls; you must have at least one trunk.

The reason why asterisk users have several trunks can be explained as follows:

- Trunk (a) charges the best rate for Local calls but is expensive for Mobile calls and not so great for international calls.
- Trunk (b) has great rate for international call.
- Trunk (c) has good mobile rate and will allow 1300 numbers while the other 2 do not.

A properly planned route will direct the phone calls you make to the appropriate trunk that will provide you with maximum effectiveness and savings.

E.g. When you make a call to a Mobile phone, asterisk will route your call via trunk (c) while it will route your call to trunk (a) if you make a local call. Similarly, when you make an international call, asterisk will route your outbound call via trunk (b).

## 8.3 TRUNKS AND OUTBOUND ROUTES – A CIRCUS OF CONFUSION

To the uninitiated, *Outbound Route Dial Patterns* and *Trunk Outgoing Dial Rules* are 2 areas of confusion. I will try to explain this in the simplest possible term.

Each of these two works in conjunction with each other – they compliment each other when an outgoing call is being made.

#### 8.3.1 Outgoing Route Dial Pattern

This is the pattern you normally will dial when you want to make a phone call. As an example, if you are living in Brisbane, and you are calling a friend who lives in Sydney, you will normally dial a number such as 02 91234567, however if that friend is in Victoria, you will need to dial 03 91234567.

Since your VOIP provider does not discriminate where you are calling from or where you are calling to, it requires a universal type number to call so your call can be directed to the correct recipient.

In most cases, the VOIP provider will need to dial the universal **e164** format e.g. **61291234567 for Sydney** or **61391234567 for Melbourne**. (**61** being the code for Australia, **2 or 3** being the code for Sydney or Melbourne respectively, followed by the telephone number of your friend which is **91234567**).

Since you are dialling 03 91234567, asterisk only need to send 391234567 (without the 0 prefix). Therefore in your Outbound Route Dial Pattern (lets call this Outgoing Route **Domestic**) you will need the following pattern:

#### 0 | NXXXXXXX

That pattern tells Asterisk to send only 9 digits out of 10 dialled by not sending the first digit 0. Thus, if you dialled 0291234567, the number that will be sent to the Trunk will be **291234567**.

How will this then be translated to 61291234567? This is where the Trunk Outgoing Dial Rule comes in.

#### 8.3.2 Trunk Outgoing Dial Rules

This is what the trunk is required to send to the VSP to make a successful call to the number you dialled.

Using the above example, assuming that the VOIP service provider requires the full e164 format, Asterisk will need to add the missing Australian code 61.

In your Trunk Outgoing Dial Rule, you will need the following pattern.

61+NXXXXXXXX

That pattern tells Asterisk to add 61 in front of the 291234567 before dialling the number via the trunk. For variations of patterns, see <u>What is a Dial Pattern</u> in the previous chapter earlier on in this document.

## 8.4 LET'S CREATE THE VOIP TRUNKS

To create a new trunk, select the **Trunks** option from the vertical menu on the left.

	System PBX Fax Email IM Reports Extras	About Elastix 0.9.2-4
PBX Configuration Asterisk-Cli	Flash Operator Panel Voicemails Monitoring File Editor Echo Canceller	
Option Unembedded freePBX	Add an Extension	
Basic	Please select your Davice below then click Submit	
Extensions	Please select your Device below then click Submit	
Feature Codes	Device	
General Settings		
Outbound Routes		
Trunks		
Inbound Californitrol		
Inbound Routes		
Announcements	Submit	
Follow Me		

For this example, let's create 2 SIP trunks; Pennytel and Oztell.

<u>Pennytel</u> will be used for all International and mobile calls Oztell will be used for all local and mobile calls.

Since we are creating SIP trunks, click on the Add SIP Trunk option as illustrated below.

	System PBX Fax Email IM Reports Extras	About Elastix 0.9.2-4	Logout
PBX Configuration Asterisk	-Cli Flash Operator Panel Voicemails Monitoring File Editor Echo Canceller		0 0
Option			
Unembedded freePBX	Add a Trunk	Add Tru	nk
Basic		Trunk ZA	AP/1
Extensions	Add ZAP Trunk	Trunk 2/	NP/2
Feature Codes	A STATE AND A STATE OF A	Trunk Et	IUM/
General Settings	Add IAX2 Trunk	Trunk SI	P/SPA3K-1
Outbound Routes	Add SIP Trunk	Trunk St	P/SPA3K-2
Trunks		Trunk IA	X2/MalaccaPBX
Inbound Call Control	Add ENUM Trunk	Trank IO	V2/MicroStyle
Inbound Routes	Add Custom Truck	Truck C	0/Delesius
Announcements	Add Casconn marik	Trunk SI	P/IPTITIUS
Follow Me		Trunk SI	P/Oztell

General Setting (This applies to all the trunks):

Outbound Caller ID: "Insert Your Name" <Your Number> (Must include the " " and <>)

Maximum channels: Leave blank for maximum.

In this example we will create a Pennytel and Oztell SIP Trunks, which we will call *pennytel* and *oztell* respectively. We will also define their dialling patterns.

#### 8.4.1 Pennytel (SIP)

In this example, we will use Pennytel for all international and mobile calls only.

Elastix Without Tears

**Pennytel** requires numbers dialled to follow the international format *Country code* + *area code* + *phone number*. To call an overseas destination you would normally dial 0011+country code + *phone number*. E.g. to call *New Zealand is 0011 64 <phone number*>, and to dial *France is 0011 33 <phone number*>. Since Pennytel only requires the country code and the phone number but does not require the 0011, the route (that we will create later) will not pass the 0011 to the trunk.

We will require the following Outgoing Dialling Rules for this trunk, which instruct the trunk to dial whatever is sent by the appropriate route. This example is for dialling Malaysia with a country code of 60, and Australian Mobile phones.

Pattern	Explanation	
60177777777	This pattern tells Pennytel to dial all Malaysian Mobile numbers starts	
00177777777	with 601 followed by 8 more digits.	
CONVVVVVVV	This pattern dials numbers start with 60 followed by a one-digit	
0000	number in the range of 2-9 followed by 7 more digits.	
61.477777777	For dialling Australian Mobile phone. The 61 is the Australian Country	
01747777777	code that will be added to the mobile phone number.	

In your **Outgoing Dial Rules** box, we will enter the following;

601XXXXXXXX 60NXXXXXXX 61+4XXXXXXXX

Next we need to create the Outgoing Setting, Incoming Settings and Registration for Pennytel.

### **Outgoing Settings**

In the Trunk Name field we will enter the name of this trunk: pennytel

In the **Peer Details** enter the following; *Pennytel preferred codecs are: g729, gsm and g723 but g729 and g723 are not installed as default in ELASTIX* 

allow=alaw&ulaw&gsm canredirect=no canreinvite=no disallow=all host=sip.pennytel.com insecure=very secret= Password type=peer username=8880XXXX

### Incoming Settings

In the User Context, enter the Pennytel User ID: 888xxxxx

In the User Details we will have the following:

canreinvite=no
context=from-trunk
fromuser=888XXXXX
qualify=no
secret= <i>Password</i>
type=user
username=888XXXXX

### **Registration**

In the registration field, enter the following:

888XXXXX: Password@sip.pennytel.com/888XXXXX

After all these have been done, click on the *Submit Changes* button and click on the red bar across the top of the screen.

It seems that in some cases, Pennytel is rejecting Asterisk. To overcome this problem, it is necessary to add the following in the general section of your sip\_custom.conf.

$\equiv$
----------

useragent = Elastix

The useragent can be any name, as long as it is **NOT Asterisk**.

#### My sip\_custom.conf looks like this;

language=au videosupport=yes allow=g729 allow=g723 allow=h261 allow=h263 allow=h263p

> ; \*\*Engin users: include this if necessary ; \*\*Engin & BBP Global users: include this if necessary

maxexpirey=3600 rtptimeout=60 rtpholdtimeout=120

defaultexpirey=600

useragent = Elastix

### 8.4.2 Oztell (SIP)

Oztell requires the numbers dialled to follow international format *Country code* + *area* code + *phone number*. To call Sydney 67541234, Oztell requires it in the following format 61 267541234. To use Oztell family cheap rate you can also prefix it with 09. Therefore the dialling rule will be in this form:

We need various patterns here:

Pattern	Explanation
899060X.	This pattern tells Oztell to dial the inter Oztell's numbers
60ZXX.	This pattern dials numbers start with 60 followed by a one digit number 1-9 followed by other numbers minimum of 2 digits from 0 - 9. This can be an international number with a Country code of 60 followed by a number between 1 and 9 inclusive.
06612+NXXXXXXX	Using the Oztell 06 rate to dial any NSW number.
0661+NXXXXXXXX	Using Oztell 06 rate to dial any Australian number.
61+1300XXXXXX	Dials 1300 numbers
61+13ZXXX	Dials 13 numbers
61+1800XXXXXX	Dials 1800 numbers

In your **Outgoing Dial Rules** box, it will look like this: – this is simply an example based on my setting; yours will no doubt be different.

899060X.	
60ZXX.	
06612+NXXXXXXX	
0661+NXXXXXXXX	
61+1300XXXXXX	
61+13ZXXX	
61+1800XXXXXX	

Next we need to create the Outgoing Setting, Incoming Settings and Registration for Oztell.

### **Outgoing Settings**

In the Trunk Name, enter the name of this trunk: oztellsip (as an example)

In the *Peer Details* we will need the following;

You must enable all the codecs that you want to use through Oztell's web page.

```
allow=ulaw&alaw
canredirect=no
disallow=all
dtmfmode=rfc2833
fromuser=899060xxxxx
host= sip2.syd.net2max.com
insecure=very
secret=Password
type=peer
username=899060xxxxx
```

### **Incoming Settings**

In the User Context, enter the Oztell User ID: 899060XXXXX

In the User Details enter the following:

```
canreinvite=no
context=from-trunk
fromuser=899060XXXXX
insecure=very
qualify=no
secret=Password
type=user
username=899060XXXXX
```

### **Registration**

In the registration field, enter the following:

899060XXXXX:Password@sip2.syd.net2max.com/899060XXXXX

After all have been done, click on the **Submit Changes** button and click on the red bar across the top of the screen.

### 8.4.3 Oztell (IAX)

Since I have IAX trunk with Oztell, I might as well include my Oztell IAX trunk setting here for general information only just in case you also want to set up an IAX trunk.

Note\* Don't forget to Forward UDP Port 4569 to your Asterisk IP address and edit your *iax.conf* file.

### **Outgoing Settings**

In the Trunk Name, enter the name of this trunk: oztelliax (as an example)

In the *Peer Details* we will need the following;

Note: You must enable all the codecs that you want to use through Oztell's web page.

```
allow=ulaw&alaw
disallow=all
host=iax2.syd.net2max.com
qualify=yes
secret=Password
type=peer
username=899060XXXXX
```

### **Incoming Settings**

In the User Context, enter the Oztell User ID: 899060XXXXX

In the User Details enter the following:

```
context=from-trunk
secret=Password
type=user
```

### **Registration**

In the registration field, enter the following:

```
899060XXXXX: Password@iax2.syd.net2max.com
```

After all have been done, click on the **Submit Changes** button and click on the red bar across the top of the screen.

The SIP Trunks screen in freePBX will look like the following illustration below:

elastix	System PBX Fax Email IM Reports Extras	About Elastix 0.9.2-4 Logout
Configuration Asterisk	Ell Flash Operator Panel Voicemails Monitoring File Editor Echo Canceller	•
on nembedded freePBX	Edit SIP Trunk	Add Trunk
16 -		Trunk ZAP/1
(tensions	Delete Trunk Pennytel	Trunk ZAP/2
eneral Settinos	In use by 2 routes	Trunk ENUM/
tbound Routes	General Settings	Trunk SIP/SPA3K-1
unks	General Settings	Trunk IAX2/MalaccaPB
und Call Control	Outbound Caller ID: "Kuala Lumpur" <603	Trunk IAX2/MicroStyle
bound Routes	Never Override CallerID: 🔲	Trunk StP/iPrimus
slow Me	Maximum channels:	Trunk SIP/Oztell
R	Disable Trunk: Disable	Trunk SIP/Pennytel
sc Destinations	Monitor Hunk Pallares.	Trunk SIP/Pennytel-2
Jeues		Trunk SIP/MyNetFone
ng Groups	Outgoing Dial Rules	Trunk SIP/FWD
ne Conditions	Did Dulas	Trunk SIP/SipPhone
onferences	Lordi Fundis. 777 A68222.	Trunk SIP/VoIPBuster-
ise Applications	60320X. 02+4NXXXXXX	
usic on Hold	O2+6XXXXXXX	
N Sets	O2+9XXXXXXX	
aging and Intercom	O+NXXXXXXX OO11+NXXXXXXXX	
vstem Recordings	0011+EXXXXXXXXXX	
note Access	VV a s TRANSANANANA	
aliback	Clean & Remove duringter	
ISA	Dial rules wizards: [pick one]	
	Outbound Dial Prefix	
	Outgoing Settings	
	Trunk Name: Pennytel	
	PEER Details	
	carredirect=no	
	canreinvite=no disallow=all	
	dtmfmode=rfc2833	
	insecure-very	
	secret=xxxxxx type=peer	
	username=6032056nnnn	
	Incoming Settings	
	USER Context: 60320560000	
	USER Details	
	canreinvite=no	
	context=from-trunk dtmfmode=rfc2033	
	fromuser=6032056nnn	
	qualify=no secret=xxxxxx	
	type=user	
	Registration	
	Register String:	
	6032056nnnn:xxxxxx@sip.pennytel.com	
	Colorado Characteria	
	Submit Changes	

I already have a few trunks setup so please ignore all the other trunks in the illustration.

\* For more trunk settings, please refer to <u>APPENDIX B</u>

Now that you know what the trunk will do, you may proceed to the next chapters to create the Inbound and outbound routes.

# 8.5 CREATE A ZAP TRUNK

A ZAP trunk is just another trunk very similar to IAX or SIP trunk except that the telephone calls (traffic) comes and goes through your PSTN line. Normally you will only create ZAP trunks if you are using FXO cards like the TDM400/A400, A1200, X100P/A100P etc.

This time when you chose to create a trunk, you will select ZAP trunk.

	About Elastix 1.0-16 System PBX Fax Emeil IM Reports Extras	Logout
PBX Configuration Flash Oper	ator Panel Yoicemails Monitoring Echo Canceller Endpoint Configuration Conference Extensions Batch Tools	0 0
Option		
Unembedded freePBX	Add a Trunk	Add Trunk
Basic		Trunk ZAP/1
Extensions	Add ZAP Trunk	Trunk ZAP/2
Feature Codes		Trunk ENUM/
General Settings	Add IAX2 Trunk	Trunk SIP/SPA3K
Outbound Routes	Add SIP Trunk	Trunk IAX2/MalaccaPBX
Trunks		Trunk IAX2/MicroStyle
Inbound Call Control	Add ENUM Trunk	Trunk CID/Drimun
Inbound Routes	Add Custom Tauli	Trunk Str/Initids
Announcements	Add Custom Frunk	Trunk StP/Oztell
Follow Me		Trunk SIP/Pennytel

You will then be presented with the following screen.

	System PAX Fax Fmail	IM Penorts	Extras		About Elas	tix 1.0-16	Logou
BX Configuration Flash Ope	rator Panel Voicemails Monitorin	ng Echo Cancelle	r Endpoint Configuration	Conference	Extensions Batch	Tools	•
ption							
Unembedded freePBX	Add ZAP Trunk						Add Thunk
anic							Trunk ZAP/1
Extensions	General Settings						Trunk ZAP/2
Feature Codes			/				Trunk ENUM/
General Settings	Outbound Caller ID:						Trunk SIP/SPA3K
Outbound Routes	Never Override CallerID.						Trunk IAX2/MalaccaP8>
Trunks	Maximum channels:	1					Trunk IAX2/MicroStyle
bound Call Control	Disable Trunk:	Disable					Trunk SIP/IPrimus
Inbound Routes	Monitor Trunk Failures:		Enable				Trunk SIP/Oztell
Announcements							Trunk SIP/Panovtal
Follow Me							Trunk CID/Depositel-2
IVR	Outgoing Dial Rules						Thurk StryPennyter-2
Misc Destinations							Trunk Str/MyNeu one
Queues	Dial Rules:	XX.					Trunk SIP/FWD
Ring Groups							Trunk SIP/SipPhone
Time Conditions							Trunk SIP/VoIPBuster-
ternal options & comiguration							Trunk IAX2/InternalLI
Conferences		Clean & Remove	duplicates				
Misc Applications	Dial rules wizards:	(pick one)		M			
Music on Hold	Outbound Dial Prefix:						
Pily Sets							
Paging and intercom							
Forking Loc	Outgoing Settings						
system Recordings							
Callback	Zap Identifier (trunk name	2): 1					
- ailuath	Submit Changes		~				

For simplicity, simply do the following in the related fields:

Maximum Channels: 1 Dial Rules: XX. ZIP Identifier (Trunk name): 1 (replace the default g0 with 1, or 2 if it is the second channel)

Click Submit changes followed by the confirm bar on top and you are done.

# 9 CALLS HANDLING

# 9.1 INBOUND ROUTES (INCOMING CALLS)

This is where the behaviour of incoming calls from all trunks is being handled.

When an incoming call from PSTN or VoIP trunk is received, asterisk needs to know where to direct it. It can be directed to a ring group, an extension, Digital Receptionist (IVR) or Queue. For this purpose, Inbound Route needs to be set up.

Select the Inbound Routes selection in the left bar of the screen.

Initially, (since we have not created an IVR yet) we will set all incoming calls to be directed to Ring Group 600 as illustrated below (later on in this guide we will create a digital receptionist for this purpose and we will redirect all the incoming calls to the IVR).



As a matter of fact, this can be the only Inbound Route that you will need if you deploy IVR (to be discussed later on). Besides, without this inbound route created, your 7777 simulated calls will not work.

elastix	System PBX Fax Email IM Reports Extras	About Elastix 0.9.2-4 Logout
PBX Configuration Asterisk-	Cli Flash Operator Panel Yoicemails Monitoring File Editor	Echo Canceller 📀 📀
Option Unembedded freePBX Basic Extensions	Route: //	Add Incoming Route any DID / any CID
Feature Codes General Settings	Edit Incoming Route	You must enter your DID or your
Outbound Routes Trunks Inbound Call Control Inbound Routes Announcements Follow Me	Description: DID Number. Caller ID Number. OR	route to be handled individually. This applies to SIP trunks as well.
IVR Misc Destinations Queues	Zaptel Channel	
Ring Groups Time Conditions Internal Options & Configuration Conferences Misc Applications Music on Hold	Fax Extension: FreePax default 💌 Fax Email: Fax Detection Type: Hene 🔍 Pause after answer: 0	CARDS
PIN Sets Paging and Intercom	Privacy	
Parking Lot System Recordings	Privacy Manager. 🛛 💌	
Rémote Access Callback	Options	

If you are using freePBX v2.4, you will not have the Zaptel Channel (see arrow in the above illustration). To handle calls coming in through the ZAP channel you will have to do define the DID of the channel through the **ZAP Channel DID** option in freePBX as described in **9.1.1** below.

For our purpose, only put in the following data. The rest are optional.

- **DID Number:** Leave blank
- CID Number: Leave blank
- Fax Extension: Disabled (I do not require fax facility here)
- Privacy Manager: No
- In the radio button, enable Ring Group 11 (AdminGroup). We want all incoming calls to ring the phones in the Admin group (see illustration below).

Permite Access       Options         Caliback       Options         DISA       Alert Info:         CID name prefix::			
Calback DISA Alert Info CID name prefix:: Music On Hold? Signal RINGING. CID Lookup Source Source: Neme Set Destination DISA: Particust? DISA: Particust? DISA: Particust? DISA: Particust? DISA: Particust? DISA: Particust? Other Call Hangup Cidential Source Call Hangup Cidential Call Hangup Cidential Sources: Promote Support Call Hangup Cidential Call Hangup Cidential Call Hangup Cidential Call Hangup Cidential Sources: Promote Support Cidential Call Hangup C	Temote Access	Ontions	
Alert Info: CID name prefix: Music On Hold? Signal RINGING: CID Leakup Source Source: Set Destination ODISA: Particover Time Conditions: Network Miner Original RING City Settlewidth Time Conditions: Network Miner Original Cold Network Starf (hurry) Oucemail: ColdOn Rehard Starf (hurry)	Callback		
Aller Linio. CID name prefix; Music On Hold? Signal RINGING; CID Lookup Source Source: Set Destination O DISA: PETHEVUEP O Time Conditions; Neurose O Time Conditions; Neurose O Time Conditions; Neurose O Eddensions; Neurose Statu O Terminate Call: Hangup O Eddensions; Scoop Rehard Baarf O Voicemail: Scoop Rehard Baarf O Phonebook Directory; Phonebook Directory M O Confferences; Sanaud Goodo M O Misc Destinations; VoiceMail M O Misc Destinations; VoiceMail M	DISA	Alast Jofa	
CLD name prenz: Music On Hold? Signal RINGING: CID Leakup Source Source: Set Destination O DISA: Pathatowate O Time Conditions: Normal @ O Time Conditions: Normal @ O Ring Groups: Indemn drage <12> @ O Terminate Call Hangup O Extensions: <a href="mailto:sease">Sease</a> O Disa: Satura drage <12> @ O Terminate Call Hangup O Queues: Internate thand thand (bury) @ O Queues: Internate thand (bury) @ O Conferences: Seased tooloo @ O Misc Destinations: <a href="mailto:sease">Sease</a> O Conferences: Seased tooloo @ O Misc Destinations: <a href="mailto:sease">Sease</a> O Custom App		CID same scaling	
Multic On Hold?       Inc.1         Signal R1NGING:       ♥         CID Lookup Source       ■         Source:       ■         Set Destination       ●         O DISA:       PSTR2Note         ♥ Ring Groups:       Mema         ♥ Ring Groups:       Mema         ♥ Ring Groups:       Mema         ♥ Disa:       [25009 Robani Bhatf         ♥ Voicemail:       [20009 Robani Bhatf         ♥ Onnebook Directory:       [Robabok Duedny ♥         ♥ Nonebook Directory:       [Robabok Duedny ♥         ♥ Misc Destinations:       [wommail ♥         ● Custom App:       ●		CLD name prenx:	
Signal KINGING: CID Leakup Source Source Set Destination O DISA: [#STR2Volp M O Time Conditions: [remain M O Ring Groups: famma Group <12.** O Terminate Call Hangup O Xuccernitic Call Hangup O Volcernitic Coolor Anhani Sharff O King Groups: [ameral 40000 M O King Isonations [volcernitic M O K King Isonations [volcernitic M		Music Un Hold?	
CID Lookup Source Source:  Set Destination  DISA: particular  Rang  Finance Conditions:  Rang Groups:  Rang Groups		Signal RINGING.	
Source: Set Destination O DISA: Patricivate & O Time Conditions: [normal @ O Ring Groups: Indering along along O Territinate Call [Hangue @ O Extensions: [acooon Rehard Bhard [Burg] O Voicemail: [acooon Rehard Bhard [Burg] O Queues: [manual acodo: ] O Conferences: [manual acodo: ] O Misc Destinations: [voiceMail @ O Misc Destinations: [voiceMail @		CID Lookup Source	
Set Destination DISA: particular  Find Conditions: [itermal  Find Conditions: [itermal  Find Course: Antimi Brand  Find Course: Find		Source Mone M	
ODISA (Particivate		Set Destination	
Control Conditions   terminal M Conditions   terminal Call   terminal terminal   terminal			
Contractional General Sectors     Contractions     C		Ime Conditions: Normal      Pipe Groups: Normal	
Caderisons (<2000 - Rohani Sharif ) Caderues: General <600 - Monis Sharif (bury) / Caderues: General <600 - M Conferences: General <6000 - M Conferences: General <6000 - M Caderues: General <6000 -		C Terminate Call Junear	
○ Voicemail <a>&gt;</a>		O Extensions (20002 Bohani Sharif	
© Queues: Gaenari codo: ₩ ○ Phonebook Directory: [Phonebook Directory: [M ○ Conferences: Gaenari elicono: ♥ ○ IVR: Isomativn: ♥ ○ Misc Destinations: [Voicentail ♥ ○ Custom App:		O Voicemail (<2000> Rohani Sharif (busy)	
Phonebook Directory [Phonebook Directory ]     Conferences		O Queues: deneral <600> 👻	
© Conferences: [canaral <8000> ♥ ○ IVR: [tiomativn ♥ ○ Misc Destinations: [voicettail ♥ ○ Custom App:		O Phonebook Directory Phonebook Directory	
		O Conferences: General <0000> 💌	
O Misc Destinations (voicemail x)		O IVR: Homesve 💌	
O Custom App		O Misc Destinations: VoiceMail M	
		O Costom App.	
Submit		Submit	

For each extra trunk, a corresponding inbound route may be created (you don't have to). All incoming calls from all the different trunks that you may have can have their own individual characteristic by judicious deployment of the time condition facility available in FreePBX (to be discussed later on).

However, at least one Inbound Route must be created for Elastix to answer incoming calls. This includes calls from PSTN trunk as well. The route that we have created above is precisely for this purpose.

This route will handle the calls that come in without Trunk ID and calls from trunks that do not have an Inbound Route created (which includes all calls from PSTN and incorrectly created incoming route). We call this a "Catch-All" route, an **Inbound Route with the DID Number and Calling ID Number fields left blank**.

As usual, you will need to click on the red bar on top of the screen after each time you submit a new Inbound Route.



#### 9.1.1 Inbound Route from ZAP Trunk

To handle inbound calls from a ZAP trunk, in *freePBX 2.4* is a little fiddly because you need to define your DID (incoming phone number) for each ZAP channel. If there are 4 FXO modules in a TDM400 card, the zap channels will be 1, 2, 3 and 4. Each FXO is a channel (this can be confirmed in Zapata-channels.conf file) and each channel can have a DID assigned to it.

Admin	Edit Zap Channel: 2	English M
FreePBX System Status		
Module Admin	Zap Channel DIDs allow you to assign a DID to specific Zap Channels. You can supply the same DID to multiple channels.	dd Channel
Advanced	This would be a common scenario if you have multiple POTS lines that are on a hunt group from your provider. You MUST	Netvoice Solutions
Custom Contexts	assign the channel's context to from-zaptel for these settings to have effect. It will be a line that looks like:	
Custom Contexts Times	2	Home
Basic	context = from-zaptel	
Administrators		
Extensions	in your zapata.conf configuration effecting the specified channel(s). Once you have assigned DIDs you can use standard Inbound	I Routes with the
Feature Codes	specified DiDs to Foure your cans.	
General Settings		L
Outbound Routes /	Edit Channel: 2	2
Trunks		
Inbound Call Control		
Inbound Routes	Description. Home number	
Zap Channel DIDs	DID: 61247364780	
Announcements		
Blacklist	Submit Unanges Delete	

This has to be done through the unembedded freePBX (**ZAP Channel DIDs** option) as Elastix does not provide this facility.

The DID that you have defined is the DID number you should use in the DID number field when defining your inbound route.

# 9.2 TIME CONDITIONDS

You can create various time conditions and use these time conditions in conjunction with your Inbound Route to individualise each of the incoming trunk's behaviour.

	System PBX Fax Email IM Reports Extras	About Elastix 1.0-12	Logout
PBX Configuration Flash Open	ator Panel   Voicemails   Monitoring   Echo Canceller   Endpoint Configuration   Conference   Exte	nsions Batch   Tools	0 0
Option Unembedded freePBX Basic Extensions Feature Codes General Settings Outbound Routes Trunks Inbound Routes Announcements Follow Me UR Misc Destinations Queues Ring Groups Time Conditions Internal Options & Configuration Conferences Misc Applications	Add Time Condition Add Time Condition Time Condition name: Time to match: Time to match: Time to finish: Week Day Start Week D	Server time: 15:46:33	Add Time Condition
PIN Sets Paging and Intercom Parking Lot System Recordings Remote Access Callback DISA	Terminate Call (Finance)     Voicemail: [sagops Resettion W     Voicemail: [sagops Resettion Work]     Zork (Read this link before downloading): [sate (sak) w     Phonebook Directory [Phanabask Directory] w     Announcements: [sitest w     Vice Trainoffice w  Destination if time does not match:      Misc Dostinations: [sat Voicemail M     Terminate Call [Finance		

You may create several time conditions and give each of the time condition a Short Name to identify it. These time conditions can then be assigned to each individual Inbound Route if you choose to do so or they can be nested.

#### 9.2.1 Simple Time Condition

Unlike businesses, most private householders, ignore the time condition, instead would prefer to direct all calls to an extension or selected extensions regardless whether it is day or night. If this is the case, you may ignore time condition altogether, instead, use the inbound route to direct your incoming calls to individual extensions or to the digital receptionist. However, if you want to accept calls during the day but prefer callers to simply leave voice messages during the night, below is an example of a simple time condition that you can adopt where the time condition will send the calls to IVR during the hour of 8:00am and 8:00pm and anything outside the stipulated hours, the calls will be directed to your voicemail.

- Add a new time condition by selecting PBX Configuration > Time Condition - > Add time condition.
- In the *Time condition name* field, give it a name eg: DayNight
- Under *Time to match*, do the following:
  - o Time to start: 08:00
  - Time to finish: 20:00
  - Week Day start: Monday
  - Week Day finish: Sunday
- Under *Destination if time matches*, select IVR (or Core extension or ring group)
- Under *Destination if time does not match*, select Core mailbox: eg voicemail **box 2001** (if you have voicemail enabled for ext. 2001).

• Submit and click on the red bar on top of the page.

Once the above is done, go to Inbound Route, select the appropriate inbound route and under Set Destination, select Time Condition **DayNight**, submit and click the red bar on top of the screen.

### 9.2.2 Complex Time Condition

A slightly more complex one can also be crafted e.g: You want o accept calls only between 8:00am till 1:00pm and between 2:00pm and 5:30pm Monday to Friday and any other time you want to direct all the calls to voicemail.

- 1. Add a morning time condition.
  - Add a new time condition by selecting *PBX Configuration > Time Condition - > Add time condition*.
  - In the *Time condition name* field, give it a name eg: OfficeMorning
  - Under *Time to match*, do the following:
    - Time to start: 08:00
    - o Time to finish: 13:00
    - Week Day start: Monday
    - Week Day finish: Friday
  - Under *Destination if time matches*, select IVR (or extension or ring group)
  - Under *Destination if time does not match*, select an extension for the moment. We will need to change this later.
  - Submit and click on the red bar on top of the page.
- 2. Add an afternoon time condition
  - Add a new time condition by selecting *PBX Configuration > Time Condition - > Add time condition*.
  - In the *Time condition name* field, give it a name eg: OfficeAfternoon
  - Under *Time to match*, do the following:
    - o Time to start: 14:00
    - o Time to finish: 17:30
    - Week Day start: Monday
    - Week Day finish: Friday
  - Under *Destination if time matches*, select IVR (or extension or ring group)
  - Under *Destination if time does not match*, select mailbox: eg voicemail box 2001 (if you have voicemail enabled for ext. 2001).
  - Submit and click on the red bar on top of the page.
- 3. Go back to time condition *OfficeMorning* and change the *Destination if time does not match* to: Time Condition *OfficeAfternoon.* Submit and click on the red bar on top of the screen.
- 4. Once the above is done, go to Inbound Route, select the appropriate inbound route and under **Set Destination**, select Time Condition **OfficeMorning**, submit and click the red bar on top of the screen.

What the inbound route will do is, as soon as an incoming call is accepted, it will be forwarded to the **OfficeMorning** time condition.

If the time condition is met, it will then be forwarded to the IVR. On the other hand, if it does not match, it will be forwarded to time condition **OfficeAfternoon**.

If it matches the **OfficeAfternoon** time condition, the call will be forwarded to IVR, if not, it will be forwarded to the voicemail box that you have determined.

Following the above example, time conditions can be nested to multiple levels deep allowing you to manage your time condition and make them as complex or as simple as the situation demands – imagination is the only restricting factor.

# 9.3 OUTBOUND ROUTES

### 9.3.1 What is an Outbound Route?

An **outbound route** works like a traffic cop giving directions to road users to use a predefined route to reach a predefined destination.

### 9.3.2 How does it work?

Every time you dial a number, asterisk will do the following in strict order:

- Examine the number you dialled.
- Compare the number with the pattern that you have defined in your route 1. If matches, it will initiate the call using that trunk. If it does not, it will compare the number with the pattern you have defined with route 2 and so on.
- Pass the number to the appropriate trunk to make the call.



To make an external call (except inter extension calls), you will need at least one trunk and one route.

In this case we will create 5 routes:

- International
- Domestic
- MobileAust
- OztellOnly
- > The International Route will be used for all international phone calls.
- > The Domestic Route will take care of the Australian phone and mobile.
- > The other two will do as the names suggest.

To create a new route using **Elastix**, select **PBX** tab and then select the **Outbound Routes** option from the vertical menu on the left.

C elastix						About Elastix 1.0-12	Logout
	System PBX Fax	Email IM F	teports E	xtras			
PBX Configuration Flash Ope	rator Panel Voicemails Mon	nitoring   Echo C	anceller E	ndpoint Configuration	Conference	Extensions Batch Tools	0 0
Option							
Unembedded freePBX	Edit Route						Add Route
Basic							0 9_outside
Extensions	Delete Route Internati	ional					
Feature Codes							1 International
General Settings	Route Name:	International	ename				-
Outbound Routes	Route Password:						
Trunks	PIN Set:	None 💙					
Inbound Call Control	Emergency Dialing						
Inbound Routes	Intra Company Route:						
Announcements	Music On Hold?	default 💌					
Follow Me	Dial Patterns						
IVR.		OO11  GONXX.					
Misc Destinations		0011  [7-9] XX.					
Queues		1933 B.B. + 1947 B.					
Ring Groups							
Time Conditions		Clean & Remove	duplicates				
Internal Options & Configuration	Dial patterns wizards:	(pick one)	~				
Conferences	Trunk Sequence						
Misc Applications	0	SIP/Dennytel					
Music on Hold	1	SIP/Ortall	- 8.				
PIN Sets		1	~				
Paging and Intercom		Add					
Parking Lot	Submit Changes						
System Recordings	Contraction of the Article of the Ar						
Pernote Access							
Callback							
DISA							
		Elastix is license	d under GPL	by PaloSanto Solutions	. 2006, 2007.		

Click on the Add Route in the menu on the right of the screen.

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### 9.3.3 International

You will need the following pattern to match international dialing.

Pattern	Explanation
0011 60NXX.	Most of our International calls are to Malaysia, I have defined the Malaysian PSTN pattern here.
0011 [7-9]XX.	This pattern dials all international numbers by prefixing it with 0011 and any digit between 7 and nine inclusive. Before the number is sent to the trunk the 0011 will be stripped. If required by the trunk, the trunk will insert the 0011.
0011 NXX.	This is the catchall international dialling including International Mobile (excluding USA which is a 1).

In your Dial Patterns box, you will see something like this.

#### **Dial Patterns**

0011 60NXX.
0011 [7-9]XX.
0011 NXX.

You will then need to pick the Trunk Sequence. In this case we will use Pennytel as the trunk for this route and if it fails, we will use Oztell as a second choice/backup.

Trunk Sequence

0	Pennytel
1	Oztell

Any number dialled with the 0011 prefix will be routed through the Astratel trunk and should it fail, it will be routed to Pennytel.

#### 9.3.4 Domestic

You need the following pattern to match the Australian Telephone numbers and the Australian Mobile numbers.

Pattern	Explanation
0 [23478]XXXXXXXX	You will dial Australian telephone number as normal using 03, 02 prefix but the route will strip the leading 0 before it is sent to the trunk. If the trunk requires the 0, the trunk will insert it.
[2-9]XXXXXX	If you dial local number and not use the STD prefix, it will send to the trunk as is and the trunk will insert the 02 if required (this is a NSW pattern. If you are in Victoria you will need to modify the trunk).
ZXX.	This is the catchall dialling for numbers that does not start with a 0 such as 1300, 1800, 13 etc.

In your Dial Patterns box, you will see something like this.

**Dial Patterns** 

0 [2378]XXXXXXXX	
[4689]XXXXXXX	
ZXX.	

You will then need to pick the Trunk Sequence. In this case we need Oztell as the trunk for this route and if it fails, we want Pennytel to take over as a second choice/backup.

**Trunk Sequence** 

0	Oztell
1	Pennytel

All numbers dialled matching the above pattern will be routed through Oztell and if it failed, it will be routed via Pennytel.

### 9.3.5 MobileAust

There is only a single pattern for calling Australian mobile – illustrated below.

**Dial Patterns** 

0 4XXXXXXX	

We want all my domestic mobile calls to be routed via Pennytel and we have not nominated a second trunk. If it fails then we will not be able to make a Mobile call, as we don't have a second trunk nominated.

Trunk Sequence

0	Pennytel

#### 9.3.6 Oztellonly

The following pattern is in the Oztellonly route. We will use Oztell for a number of call types therefore some explanation may be in order.

Pattern	Explanation
0011 601XX.	This pattern will allow me to dial a Malaysian Mobile number. The 0011 will be stripped before the number is sent to Oztell. Oztell does not require the 0011. It only requires the Country code.
0 [378]NXXXXXX	You will dial Australian telephone number as normal using 03, 02 prefix but the route will strip the leading 0 before it is sent to the trunk. If the trunk requires the 0, the trunk will insert it.
1[38]XX.	This is for dialling 1300, 1800 and 13 numbers.
899060XX.	For dialling Oztell inter Oztell users
NXXXXXXX	Catchall for dialling a local number that doesn't start with 1

In the dial pattern box you will see something like the illustration below.

**Dial Patterns** 

0011 601XX.	
0 [378]NXXXXXXX	
1[38]XX.	
NXXXXXXX	

Since the above pattern is for Oztell only, in my trunk sequence, Oztell was chosen to be the only trunk for any of these calls.

**Trunk Sequence** 

0	Oztell

## 9.4 ENUM

ENUM is a way for us to use VOIP to establish calls between VOIP systems without using PSTN even though the numbers called are PSTN numbers provided that the numbers called are registered with ENUM. In short, ENUM offers you the ability to call other telephones directly over the Internet without involving a third party.

I will not enter into any elaborate explanation how enum works, as there are already a number of articles describing this. In essence, your phone number and server information are stored in a special DNS zone (very similar to a reverse lookup zone). When an ENUM enabled Asterisk user calls your phone number, it will first attempt to find your phone number in the ENUM DNS zone. If it finds your phone number it will use the information to setup a call directly between his Asterisk server and yours, resorting to PSTN only if your number is not found.

**Note:** ENUM only works between ENUM enabled Asterisk (or other ENUM enabled VOIP) users.

### 9.4.1 Setting up ENUM Trunk

- 1. Click the Add Enum Trunk link.
- In the Outbound Caller ID field enter your callerid information in the format of "Your Name" <61246123456> (Including the quotes), assuming your telephone number in Sydney is 0246123456.
- 3. On the outgoing Dial Rule enter 612+NXXXXXX (for local calls without having to dial the 02)
- 4. On the second line enter 61+NXXXXXXX (for dialling Australian numbers where you need to dial the prefix (eg. 03)
- 5. On the next line enter ZXXXXXXX. (Including the dot) for dialling International numbers see example below:

elastix	System PBX Fax 1	mail IM Reports Extras	About Elastix 0.9.2-4	Logout
PBX Configuration Asterisk	Cli Flash Operator Panel Voic	emails Monitoring File Editor Echo Canceller		0 0
Ceten Unembedded freeFBX Dear Extensions Freature Codes General Settings Outbound Routes Trunks Inbound Routes Announcements Follow Me IVR Misc Destinations Queues Ring Groups Time Conditions Immar Others & Configuration	Edit ENUM Trunk Delete Trunk In use by 1 route General Settings Outbound Caller ID Never Override CallerID Maximum channels: Disable Trunk: Monitor Trunk Fallures: Outgoing Dial Rules Dial Rules:	Tietroico" <6124736		Add Trunk Trunk ZAP/1 Trunk ZAP/2 Trunk ZAP/2 Trunk SIP/SPA3K-1 Trunk SIP/SPA3K-2 Trunk SIP/SPA3K-2 Trunk SIP/SPA3K-2 Trunk SIP/Pennytel Trunk SIP/Pennytel Trunk SIP/Pennytel-2 Trunk SIP/FwD Trunk SIP/FWD Trunk SIP/FWD
Conferences Misc Applications Music on Hold Phi Sets Paging and Intercom Parking Lot System Recordings Remote Access Caliback DISA	Dial Rules Dial rules wizards Outbound Dial Prefix Submit Changes	6 12 - 875 XXXXXXX 1 - NXXXXXXXXX 2 XXXXXXXXX Clean & Remove displicates (pick one)		Trunk SIP/VoIPBuster-

- 6. Click the Submit Changes button at the bottom of the page
- 7. Click the Apply Configuration Change at the top of the page.

### 9.4.2 Setting up ENUM Outbound Route

- 1. Click the **Outbound Routes** link on the left hand side.
- 2. To create a new route, click on Add Route the top option on the right side of the screen.

- 3. Name the route **eNUM**.
- 4. We will to use the prefix 164 to force dial through ENUM
- 5. In the Dial Pattern, on the first line, we insert 1640 [378] NXX. This will allow us to dial an STD number such as 0732123456 by dialling 1640732123456 where only the 732123456 will be sent to the ENUM trunk.
- 6. On the second line we insert 1640011|NXX. This will allow us to call an International number eg. To call Malaysia we will dial 16400116062861234 and only the 6062861234 will be sent to the ENUM trunk.
- 7. On the third line I have 164|NXX. This enables me to dial NSW local number without having to dial 02.
- 8. Make the ENUM trunk the first route in your **Trunk Sequence**. You may also add other trunks in the sequence in case ENUM failed.
- 9. Click the **Submit Changes** button at the bottom of the page.
- 10. Click the Apply Configuration Change at the top of the page.
- 11. You may try calling someone while monitoring Asterisk CLI and you should see the Enumlookup scroll by.

	System PBX Fax Email IM Reports Extras	About Elastix 0.9.2-4	Logo
PBX Configuration Asterisk-	Cli Flash Operator Panel Yoicemails Monitoring File Editor Echo Canceller		0
Dption			
Unembedded freePBX	Edit Route		Add Route
Jasic	Delete Deute (MIM		3 U ZAPI
Extensions Each re Codec	Delete Route eNUM		1 ZAP2
Feature Codes	Route Name: eNUM Rename		23
General Settings	Route Password:		2 SPA3K
Trunke	PIN Set:		3.2
nbound Call Control	Emergency Dialing:		3 eNUM
Inbound Routes	Intra Company Route:		d Malagaa
Announcements	Music On Hold?		3 Malacca
Follow Me			5 MicroStyle
IVR	Dial Patterns		23
Misc Destinations			6 Oztell
Queues	1640011 NXX. 1640 [378]NXX.		33
Ring Groups	1641NXX.		7 Pennytel
Time Conditions			P iPrimus
nternal Options & Configuration			23
Conferences	Clean & Remove duplicates		9 MyNetFone
Misc Applications	Dial patterns wizards: (pick one)		23
Music on Hold			10 Domestic
PIN Sets	To all Company		23
Paging and Intercom	Trunk sequence		11 AustMobile
Parking Lot	0 Enum/		12 International
System Recordings	×		33
Remote Access	Add		
Callback			
DISA			

Click here for a Step-by-Step instruction on how to set up an ENUM account.

# **10 CUSTOM CONTEXTS MODULE**

The custom contexts module currently is a beta module that must be manually installed.

First a brief description of what custom contexts does (version 0.3.3):

Custom Contexts creates contexts that can be used to allow limited access to dial-plan applications.

Features:

- Allows for time restrictions on any dialplan access.
- Allows for pattern matching to allow/deny.
- Allows for failover destinations, and PIN protected failover.
- This can be very useful for multi-tennant systems.
- Inbound routing can be done using DID or zap channel routing, this module allows for selective outbound routing.
- Phones can be placed in a restricted context allowing them only internal calls.

More info: <u>http://aussievoip.com.au/wiki/freePBX-CustomContexts</u>

### **10.1** DOWNLOADING AND UNPACKING MODULE.

Login on the Elastix box manually as user root and do the following:

```
cd /var/www/html/admin/modules
wget http://www.freepbx.org/trac/attachment/ticket/1447/customcontexts-
0.3.3.tgz?format=raw
mv customcontexts-0.3.3.tgz?format=raw customcontexts-0.3.3.tgz
tar -xzvf customcontexts-0.3.3.tgz
rm -f customcontexts-0.3.3.tgz
```

# **10.2 INSTALLING CUSTOM CONTEXT MODULE**

On the Elastix web interface, click on the PBX, Unembedded freePBX.

	System PBX Fax Email JM Reports Extras	About Elastix 1.0-13	Logout
PBX Configuration Flash Opera	tor Panel Voicemails Monitoring Echo Canceller Endpoint Configuration	Conference Extensions Batch Tools	00
Option	Add an Extension		Add Extension
Basic Extensions	Please select your Device below then click Submit		Reception <2000> Ben Sharif <2001>

#### Default Login/Password is admin/admin

0	Enter username and password for "EreePBX Administration" at https://192.168.1
50	User Name:
	admin
	Password:
	*****
	Use Password Manager to remember this password.

On entering freePBX, select the following:

Admin Tab  $\rightarrow$  Module admin  $\rightarrow$  Custom Contexts (as per the illustration below). At this stage, you will notice that Custom Contexts is not installed.

FreePBX 24.0.0 m 192 188 1.37	Admin Reports F	Panel Help			
Setup Tools Admin FreePBX System Status	1 Module Admini	stration			
Module Admin Basic Administrators	Check for updates online	Upload module	e		
Extensions	Module	Туре	Version		
Feature Codes General Settings	Advanced				0
Outbound Routes	Custom Contexts	setup		Not Installed (Locally available)	
Trunks Inbound Call Control Inbound Routes	Basic	3		(	33
Zap Channel DIDs	Builtin	setup		Enabled	
Announcements	Core	setup	2.4.0.1	Enabled	

Select Install and click on Process.

FreePBX 2.4.0.0 on 192.160.1.37	Admin Reports	Panel Help			Logged in: admirr (Logeut)
Setup Tools Admin FreePBX System Status	Module Adm	inistration			English 💌
Module Admin Basic	Check for updates or	nline   Upload module			Deres Deres
Administrators Extensions	Module	Туре	Version		Heset Process
Feature Codes General Settings	Advanced				/
Outbound Routes Trunks Inbound Call Control Inbound Routes Zap Channel DIDs	Custom Contexts Action Description Changelog	Setup O No Action Install		Not Installed (Locally available)	2
Announcements Blacklist CallerID Lookup Sources Davidijett Control	Basic	1			
Follow Me	Builtin	eatin		Enabled	

Confirm you action by clicking on the confirm button



Click on the orange bar to complete the installation.

O Elastix	📑 🦂 FreePBX adm	inistration 📮	l.		
FreePBX 2.3.1.3 on 192.168.101.1	Admin Reports	Panel Help			Logged In: admin (Logoti)
Setup Tools	Module Adm	have made changes t nut will been epoled ig all changes, click o	o the configuration th When you are finish in Apply Configuration	at nd	English 💌
FreePBX System Status Module Admin	Check for updates onlin	e   Upload mod	ule		
Custom Contexts Custom Contexts Times	Module	Туре	Version		Reset Process
Administrators Extensions	Advanced				
Feature Codes	Custom Contexts	setup	0.3.3	Enabled	

# **10.3 ADD AND CONFIGURE NEW CUSTOM CONTEXT**

Add New custom Context: Admin  $\rightarrow$  Setup  $\rightarrow$  Custom Context

	FreePBX 2.3.1.3 on 192.168.101.134	Admin Reports Panel Help
	Setup Tools Admin FreePBX System Status Module Admin	Add Context
To edit the custom-context, only select: Allow or Deny each context.	Advanced Custom Contexts Custom Contexts Times Basic	Context
	Administrators	Context cell-calls
	Extensions	Description allow cell calls
	General Settinas Outbound Routes Trunks Inbound Call Control	Custom Contexts v0.3.2
	Inbound Houtes Announcements CallerID Lookup Sources Follow Me	Submit
	Outbound Routes	
To control any outbound call, select the route and from thes	outrt-001-local	Allow Priority 51
selection, select Allow or	outrt-002-internation	al Deny 🔽 Priority 52 🔽
Deny.	outrt-003-cell-phone	Allow  Priority 53

# **10.4 APPLY A CUSTOM CONTEXT TO EXTENSION.**

	This device uses s	ip technology.
	secret	100
	dtmfmode	rfc2833
	canreinvite	no
	context	from-internal
	host	dynamic
	type	friend
	nat	yes
custom context on	port	5060
ce Options.	qualify	yes
	callgroup	
	pickupgroup	
	disallow	
	allow	
	dial	SIP/100
	accountcode	
	mailbox	100@device
	Custom Context	

### To finish, you use th Edit Extensions, De

# Reference

http://aussievoip.com.au/wiki/freePBX-CustomContexts

http://www.elastix.org/index.php?option=com\_openwiki&Itemid=27&id=installation#first\_time\_access\_to\_the\_web\_interface

www.hellodigi.ir

# **11 SYSTEM RECORDINGS**

System recording is a facility available under freePBX to enable the recording of customised voice prompts etc.

By now we should be able to navigate ourselves through the freePBX menu. To get to the system recordings, select the **PBX Configuration** tab and pick **System Recordings** as per the illustration below.



### **11.1 DIRECT RECORDING**

Enter the extension number of the mouthpiece that you will be using to record your message in the appropriate field.

Step 1: Record or upload If you wish to make and verify recordings from your phone, please enter your extension number here: 3000 Go

You will be making your recording using the telephone on this extension (or the microphone of the softphone).

Click Go.

Follow the prompt on the screen and dial \*77 on your phone to do your recording now.

Step 1: Record or upload	BenClinicClosed
Using your phone, dial *77 and speak the message you wish to record.	HazaMenu1

In my message, I have asked one of the receptionists to record the following:

Elastix Without Tears

"Hi, you have reached the Family Health Centre. Press 1 for Reception, 2 for the Centre's Manager, 3 for Dr. Roberts, 4 for Dr. Forbes, 5 for Dr. Wong, 6 for the Admin Office, 7 for Lab and X-Ray or you may simply hold and your call will be answered by one of our available receptionists. If no one answers your call, please leave a message". - Then hang up.

After recording or uploading, <i>dial *99</i> to listen to your recording.	Step 2: Verify		
If you wish to re-record your message, dial *77	After recording or uploading, dial *99 to listen to your recording.		
	If you wish to re-record your message, dial *77		

Verify the recording by dialling \*99. Go to step 1 again if you are not happy with the recording. Keep doing this till you are satisfied.

Step 3: Name		
Name this Recording:		
Click "SAVE" when you are satisfied with your recording Save		

Once you are happy with the recording, give the recording a descriptive name e.g. "*MainMenu*" so you will know what the recording is for.

Click Save.

## 11.2 UPLOAD PRE-RECORDED MATERIAL

If you have a recording that you have prepared in a .wav format you can simply upload that recording by browsing you local hard drive, locating the file that you want to upload and upload it to Elastix.

	About Elastii System PBX Fax Email IM Reports Extras
PBX Configuration Asterisk-Cli	i Flash Operator Panel Voicemails Monitoring File Editor Echo Canceller
Option Unembedded freePBX Basic	System Recordings
Extensions	Add Recording
Feature Codes	
General Settings	Step 1: Record or upload
Outbound Routes	
Trunks	If you wish to make and verify recordings from your phone, please enter your extension number here:
Inbound Call Control	Alternatively, upload a recording in any supported asterisk format. Note that if you're using .way, (eg, recorded
Inbound Routes	be PCM Encoded, 16 Bits, at 8000Hz:
Announcements	Browse Upload
Follow Me	
IVR	Step 2: Name
Misc Destinations	Name this Departing:
Queues	
Ring Groups	
Time Conditions	Click "SAVE" when you are satisfied with your recording Save
Internal Options & Configuration	
Conferences	
Misc Applications	

Follow the prompt for the upload and once again when the upload is completed, give the file a descriptive name for you to know what that recording is for.

The above recorded files either recorded manually or uploaded using the system recording facility will be stored in the */var/lib/asterisk/sounds/custom* directory.

# **12 DIGITAL RECEPTIONIST**

Like most organisation, where possible, we would like to automate the redirecting of all incoming calls. The digital receptionist is very handy for these sorts of things. Unless the calls are non-specific and will need the assistance of the receptionist (in our case also the operator), the system should allow callers to make the selection.

## 12.1 SETTING UP DIGITAL RECEPTIONIST

In the PBX Configuration tab, click on the IVR option and you will get the following screen.



Click on the *Add IVR* option on the top right on the screen (see arrow) and the new Unnamed IVR configuration screen will be presented to you.

elastix	System PBX Fax Email IM Reports Extras	Logout
PBX Configuration Flash Ope	rator Panel Voicemails Monitoring Echo Canceller Endpoint Configuration Conference Extensions Batch Tools	• •
	Apply Configuration Changes Here	
Option Unembedded freePBX Basic Extensions Feature Codes General Settings Outbound Routes Trunks Inbound Call Control Inbound Routes Announcements Follow Me IVR Misc Destinations Internal Options & Configuration Conferences Misc Applications Musc on Hold	Digital Receptionist Edit Menu Unnamed Delete Digital Receptionist Unnamed Change Name Unnamed Timeout Enable Directory  Directory Context Enable Direct Dial  Announcement None  Increase Options Save Decrease Options  Return to IVR. Directory Phaneboch Directory  Phaneboch Directory  Networks of the Save Decrease Options  Phonebook Directory Phaneboch Directory  Directory  Directory Phaneboch Directory  Directory	Add IVR Unnamed
PIN Sets Paging and Intercom Parking Lot System Recordings Remote Access Callback	Terminate Call. <u>Hangup</u> Return to IVR.     Catched the Sink before downloading). <u>SolP (Zork)</u> O Phonebook Directory. <u>Phonebook Directory</u> IVR: <u>Linnamed</u>	
DISA	O Terminate Call: Hangup	

We will then fill in the following fields:

Name (Change Name): MainOffice (that's what we shall called the main menu) Timeout: 10 (leave it at default) Enable Directory: Tick Directory Context: Leave it at Default Enable Direct Dial: Tick Announcement: MainOfficeMenu (That's what we called the IVR recording that was made for this purpose)

(See illustration below)

Page 72 of 257
	System PBX Fax Email IM Reports Extras	Logout
PBX Configuration Flash Ope	rator Panel Voicemails Nonitoring Echo Canceller Endpoint Configuration Conference Extensions Batch Tools	• •
	Apply Configuration Changes Here	
Option Unembedded freePBX Basic	Digital Receptionist	Add IVR Unnamed
Extensions Feature Codes	Edit Menu Unnamed	
General Settings Outbound Routes Trunks Inbound Call Control Inbound Call Control Announcements Follow Me IVR	Celete Digkal Receptionist Unnamed Change Name Information Information Directory Directory Directory Context Information Infor	
Misc Destinations Queues Ring Groups	Increase Options Save Decrease Options	
Internal Options & Configuration Conferences Misc Applications Music on Hold	Return to IVR	
PIN Sets Paging and Intercom	O Terminate Call: Hangup	

Then proceed to enter the selection options that you require for this IVR when the caller presses the appropriate option.

If you need more choice options in the IVR, click on the *Increase Options* button at the bottom of the screen.

In this IVR, we have defined our requirement as outlined below:

For number 1 - 7, we select *Extension*, and for 8, which we have changed to the letter "t", we also selected *Extension*.

For 1, we will select the Reception (Extension 3000).

For 2, select 3001, which is the Manager's extension.

For 3, select Dr. Roberts (Extension 3002)

For 4, select Dr. Forbes

For 5, select Dr. Wong

For 6, select Admin Office

For 7, select Lab and X-Ray

For 8, (which we shall change to "t") select extension 3000, which is the reception.

Parking Lot System Recordings Remote Access Callback	Z	Cork Read this link before downloading): [zetP (zerk)      Phonebook Directory: [Phonebook Directory      NR; Unnamed      NR; Unnamed	
DISA	Return to IVR	Terminate Call [Hangup     Detensions [=2006 #arceptaion M     Zork [Read this link before downloading]; [zate (zerk)      Phonebook Directory; [Phonebook Directory; W     (MR; [Linnaued M	
	Increase Options 5	Ve Decrease Options Elastix is licensed under GPL by PaloSanto Solutions. 2006, 2	007.

The last extension (which we have changed to the letter "t") is to capture timeout, if the caller does not press any option or, in case the DTMF tone does not register, the call will be sent to the receptionist.

Once finished, click **Save** and also click on the **Apply Configuration Changes** on the top of the screen.

Oela	stix <sup>,</sup>							1944 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 - 1949 -		About Elastix 0.9.0-14	Logout
Сцетон на	OF TELEPHONY	System	РВХ	Fax	Email	IM	Reports	Extras			
PBX Configuration	Asterisk-Cli	Flash Opera	tor Pan	el V	oicemails	Mc	onitoring	File Editor	Echo Canceller		• •
1						A	pply Cor	figuration	Changes Here		
Option									K		

Test it by dialling **7777**, (Asterisk will simulate an incoming call) and you will hear your Digital receptionist in action.

## 12.2 MULTI-LANGUAGE IVR

(This is included in this section as an example only and is not part of this tutorial. However you can try this if you want to).

Elastix allows multi languages IVR handling by simply telling Elastix what language to use. Naturally you will need to install all the language sets that you wish to use.

See <u>Installing Other Languages</u> in the chapter referring to <u>Customised Voice</u> in later part of this document.

## 12.2.1 Create Multi Languages IVR

I am assuming that you already have your *MainMenu IVR* in English set up for selecting various options. If you havnt done so, lets rename this IVR to *MainMenu\_EN*.

You will also need to create an IVR called *MainMenu\_ES* with options to be presented in Spanish. This is similar to your *MainMenu\_EN* except that it is presented to the caller in Spanish

## 12.2.2 Define Language Codes

Before you go any further, you will have to defive the various languages that you will be using. To do this you will need to choose the following options from Elastix PBX TAB:

PBX Configuration - - > Unembedded freePBX

Once you are in freePBX, select the Languages option from the vertical menu on the lafthand side. You will enter the following screen.

FreePBX	Admin Reports Panel Recordings Help	S. C.
FreePBX 2.4.1.2 on 192.168.1.100	Logged	in: admin (Logieut)
Setup Tools	Edit: Spanish (es)	English 💌
FreePBX System Status		
Module Admin	Languages allow you to change the language of the call flow and then continue on to the desired destination. For example, you may	Add Language
Basic	have an IVR option that says "For French Press 5 now". You would then create a French language instance and point it's	English
Administrators	destination at a French IVR. The language of the call's channel will now be in French. This will result in French sounds being chosen	English
Extensions	if installed.	Spanish
Feature Codes		
General Settings	Edit Language Instance	
Outbound Routes		
Trunks	Description: Spanish	
Inbound Call Control	Lannuare Code: as	
Inbound Routes		
Zap Channel DIDs		
Announcements	Destination:	
Blockdiet		

This is where you define your languages codes. The example above shows that I have defined the Spanish code where the following fields were populated:

Description: **Spanish** Language Code: **es** 

The destination for this code will be IVR: MainMenu\_ES

You will need to do the same for the English language code.

Description: **English** Language Code: **en** 

The destination for this code will be IVR: MainMenu\_EN

## 12.2.3 Create Language Choice IVR

You will need to create another IVR for selecting the language options that you want to present to the callers. This will be the IVR that will greet all callers with Select 1 for English, 2 for Italian etc. In this example, we will have 2 language sets, Australian and Spanish. Let us name this second IVR *LanguageChoice*.

This will be the first IVR that will greet all callers.

In *LanguageChoice*, we will have 3 options.

- Select 1 for English (presented in English)
- Select 2 for Spanish (preferably presented in Spanish so the Spanish speaking caller can understand it).
- The third option is unannounced as it will be the time out option represented by the letter "t". This option is the option the IVR will default to if no choice is made or if the caller's DTMF tone is not recognised by Elastix.

In choice 1, select the Languages radio button and from the dropdown choice, select *English*.

In choice 2, select the Languages radio button and from the dropdown choice, select *Spanish*.

The third choice, instead of a number, you will enter the letter "t" in the option box, select IVR radio button and from the dropdown choice, select **MainMenu\_EN** (if your callers are mainly English speaking or you can select MainMenu\_ES if the majority of your callers are Spanish speaking).

## 12.2.4 Change inbound route

Now you will need to change your inbound route to point to the *LanguageChoice* IVR. All calls will then be greeted by the LanguageChoice IVR and when the caller select 1 (or 2), the IVR will call the appropriate exten. If the caller selected 1, point it to the Language option English and the *MainMenu\_EN* will be selected as defined in the Define Language Code section above and the language will be set to English or Spanish if you pointed to MainMenu\_ES (Naturally, you must also have recorded the call options announcement in the various languages).

Save it and re-read the configuration change and you are done.

# 12.3 MULTI-TENANTS IVR

(This is included in this section as an example only and is not part of this tutorial. However you can use this if you want to).

While freePBX is a very versatile GUI front-end for Elastix it does not have the facility to configure a PBX for multi-Tenants application. However, Elastix (Asterisk) has the ability to do it with a little creative manipulation.

In the chapter above we covered multi languages IVR handling by simply telling Elastix what language to use. We are going to use the same principal to handle the Multi-Tenants requirement.

In this example, I will use a Medical Centre in Malaysia, that I have the privilege of installing the Asterisk PABX for. There are 3 tenants involved – a General Practitioner, a Dental Practitioner and a Pathology representative. Each has separate incoming telephone numbers with their own receptionist but using a common PABX.

I have picked this organisation as an example because it is small enough to write about. If I choose one of my bigger installations, this book will never see the light of day <sup>(2)</sup>

The main requirements are pretty straightforward:

- Calls that come in through the different tenants should be directed to the appropriate IVR. One IVR for each tenant.
- 12 extensions 4 for Dental Practitioner, 3 for Radiologist and 5 for General Practitioners.
- Voicemail for each receptionist.

There are other requirements such as multi language prompts, after-hours ring groups, time conditions, Interlinking 2 Elastix boxes between the Medical Centre and the Pathology Main Office in Kuala Lumpur etc., but I have only listed the major requirements.

In this example I will only highlight the Multi-Tenants incoming route handling and restricting outgoing calls to the appropriate telephone lines only.

My first task was to install the base Elastix as per the chapter on Installation. In this installation, there is no VOIP trunk involved. All telephone communications are through PSTN lines apart from the Elastix inter-connect between the Medical Centre and the Pathology Main Office in Kuala Lumpur, which will be done via a DSL connection.

## **PSTN Interface**

2 x TDM400 were installed in the PC designated for the PBX

- 3 ports allocated to the General Practitioners (ports 1-3)
- 3 ports allocated to Dental (ports 4-6) and
- 2 ports allocated for Pathology (ports 7-8).

#### **IVR Prompts**

3 x IVR were created, one for each of the different tenant.

- MedicalIVR
- DentallVR
- PatholVR

Each IVR will act as a traffic cop to direct calls to the appropriate extensions belonging to each tenant.

## Inbound Routings

- ZAP channels 1,2 and 3 (3 PSTN numbers for GP) routed to MedicalIVR
- ZAP channels 4,5 and 6 (3 PSTN numbers for Dentist) routed to DentalIVR
- ZAP channel 7 and 8 (2 PSTN numbers for the Pathology representative) routed to PatholVR

## Extensions

- 1. Extensions 2000 to 2004 allocated to GP
  - 2000 for Receptionist/Clerk with voicemail
  - 2001 for Doctor A
  - 2002 for Doctor B
  - 2003 for Surgery Room
  - 2004 for Dispensary

Change the context of these extensions, from [from-internal] to [from-internal-medical]

- 2. Extensions 3000 to 3003 allocated to Dental
  - 3000 for receptionist/clerk with voicemail
  - 3001 for Dentist A
  - 3002 for Dentist B
  - 3003 Spare extension

Change the context of these extensions, from [from-internal] to [from-internal-dental]

- 3. Extensions 4000 to 4002 allocated to Pathology
  - 4000 for receptionist/clerk with voicemail
  - 4001 for Radiographer
  - 4002 for Assistant

Change the context of these extensions, from [from-internal] to [from-internal-pathology]

## Outbound Calls Handling context

Create 3 different call handling contexts in the **extensions\_custom.conf** configuration file.

- [from-Internal-medical]
- [from-internal-dental]
- [from-internal-pathology]

The purpose of these contexts is to restrict extensions belonging to each tenant to be confined to telephone lines belonging to that particular tenant when making external calls. See <u>How to restrict outgoing calls</u> in the Calls Handling Notes chapter of this document.

You also install the <u>Custom Context Module</u> and use this module to do this. Actually it is much simpler using the custom context module.

All of the basic set up requirements above are explained in the various chapters of this book and should not present any difficulty to complete.

Once the basic is completed, the rest of the variations and extras can be handled as per the individual requirement.

# **13 CHECK YOUR SYSTEM STATUS**

The following are some basic steps for you to do – at least to ensure that your installation is running correctly and is registered to the appropriate VSP before you start doing anything more adventurous.

First you must ensure that asterisk is well and running or you may not be able to make or receive any call.

If all processes are running then your battle is half won.

We can do the initial checking through unembedded freePBX.

Click on the PBX Tab then click on the Unembedded freePBX option as shown in the illustration below.

Elastix - mozina riferox		66
Ele Edit Yew History Bookmark	s Iools Heb	
🚸 • 🔿 • 😴 😳 🏠 🛚	) https://192.168.1.37//menu-phyconfig	e
🕗 Flastis	🔯 👷 The New Strats Times Online	
elastix	1 About Elastix 1.0-12 System PBX Kox Email IM Reports Extras	Logout
PBX Configuration Flash Ope	rator Pagel Voicemails Monitoring Echo Conceller Endopint Configuration Conference Extensions Batch Tools	
Option	Add an Extension	Add Extension
Option Unembedded freePB	Add an Extension	Add Extension Reception <2000>
Option Unembedded freeFBX	Add an Extension 2 Please select your Device below then click Submit	Add Extension Reception <2000>
Option Unembedded freePB Basic Extensions Feature Codes	Add an Extension 2 Please select your Device below then click Submit Device	Add Extension Reception <2000>
Option Unembedded freePB x Basic Extensions Feature Codes General Settings	Add an Extension 2 Please select your Device below then click Submit Device	Add Extension Reception <2000>
Option Unembedded freeFBX Basic Extensions Feature Codes General Settings Outbound Routes	Add an Extension 2 Please select your Device below then click Submit Device Device	Add Extension Reception <2000>

The freePBX system status screen will be displayed.

FreePBX	Admin Reports Panel Help	Logge	In: admin (1) for
Setup Tools Annual FrankPEX System Status	FreePBX System Status		English 💌
Module Admin	FreePBX Notices	System Statistics	
Administrators Extensions Extensions Feature Codes Concernal Settings Outbound Routes Trunks Ubbound Routes Zap Channel D/Ds Announcements	As symithic from modules failed     On ornal address for online update checks     On a faile Address for online update checks     one available     show all     FreeCPEX Statistics     O     Determine case         0     External case         0     External case         0     External case         0	Processor Losd Awrage 011 OPU 0% Memory 6% Dolake 10% End among 10%	
Blacklist CallerID Lookup Sources Day/Night Control Follow Me MR	FreePBX Connections 0 If Phones Online 0 If Trainis 0 If	Networks lette rearies 0.45 ct/vs lette transmit 0.77 kt/vs	
Queues Ring Groups	Uptime	Asteriak Color	
Time Conditions Internat Option & Comparation Callback Conferences	System Uptime: 21 hours, 52 minutes Asterisk Uptime: 17 hours, 5 minutes Last Reload: 12 hours, 58 minutes	Op Panel OR M/SOL OK S Web Server OR SBH Server OK	
DISA Languages Misc Applications	FreePBX* Freedom to Connect®		

The server status should all display OK and green as illustrated above.

If any of the four services aren't running, chances are something is wrong and your Elastix will not be working correctly or not working at all. A probable cause can be wrong configuration of your zaptel drivers. If you don't have a zaptel devise, you should try to disable the related .conf files.

Ensure that the minimum requirements have been met. If the minimum requirements and the right type of hardware requirements have been met, and yet it is still not working, then maybe its time to start the installation again.

Next, while on the freePBX screen, Tools  $\rightarrow$  Asterisk info  $\rightarrow$  Full Report.

Emo	<b>7</b> 8			0-2
LIGGLDV	Admin Reports	Panel Help		Sizar
FreePBX 2.4.0.0 on 192.168.1.37			Log	ged in: admin (Logiou)
Setup Tools	Asterisk (Ver. 1	.4.18): Summary	1	English 💌
Module Admin				Summary
Support	Sum	mary		Registries
Asterisk Logfiles				Channels
Online Support				Charnes
FreePBX Support				Peers
System Administration	Last reload: 13 hours. 12	19 minutes, 13 seconds minutes, 57 seconds		Sip Info
Asterisk API				IAX Info
Asterisk Info	Active Sip Channels: 2	Active IAX2 Channels: 0		Conferences
Asterisk Physebook	SIP Registrations: 2	IAX2 Registrations: 0		Subscriptions
Backup & Restore				Voicemail Users
Custom Destinations	SIP Peers	IAX2 Peers		Full Report
Custom Extensions	Online: 3	Online: 2 Offline: 0	V 5 — P	T diriveport
Java SSH	Offline: 1	Unmonitored: 0		
PHP Info	· · · · · ·			
PHPAGI Config				
Third Party Addon	Refresh			
Customer DB				
Gabcast				
Inventory	FreePBX	reedom to Connect* sePDX is a registered trademark of Aten	ngo, LLC.	
Print Extensions	Fr	sePBX.2.4.0 is licensed under OPL		

When presented with the full report page, scroll down to the SIP (or IAX if you are using IAX) section and ensure that you are registered to your trunks. You will only be able to make and receive external calls – hopefully - if you are registered.

See illustration below.

Sin Remistry
Nost Username Pefresh State Peg.Time sip2.syd.net2max.com/5060 899060194748 105 Registered Tue, 01 Apr 2008 10:46:00 sprimusr15060 AU_6CSYOFWR 105 Registered Tue, 01 Apr 2008 10:45:26
Sip Pens
Name/username         Host         Dyn Nat ACL Port         Status           SPARK         192.166.1.91         5061         OK (0 ms)           Otesil/S90060194748         210.80.176.20         5060         Dimanitored           10.01/A2_CONTOFWR         (Daspecified)         D         0         UMERICUM           2000         (Unspecified)         D         0         0         UMERICUM           4 sip peers [Monitored 1 online, 1 offline Unmonitored: 2 online, 0 offline]         0         0         0         0
IXX2 Registry
Host dnsmgr Username Perceived Refresh State
IAX2 Peers
Hame/Hosrname Host Hack Port Status HisroStyleFEX 122.107.26.143 (3) 255.255.255 4569 OK (19 ms) HalaccaFEX 60.53.77.107 (3) 255.255.255.255 4569 OK (394 ms) 2 iax2 peers [2 online, 0 offline, 0 unmonitored]
Subseribe Notify

If all trunks are registered, you can be almost certain that at the very least, your Elastix installation is working (although not necessarily that you can make or receive calls).

In the majority of cases, if you have done everything as described in the preceding chapters, the chances of your installation working are pretty good. However there are always times when you think that the weather is going to be fine and yet you are pelted by a thunderstorm.

If you are not able to make or receive calls, the most common causes are Trunk Registrations, the choice of Codec, Routings, and Dialling rules errors.

Routings and Dialling rules are something that needs to be thought out logically with no simple way of determining, as different people has different requirements and different VSP have different dialling rules.

# **14 INTERFACING ASTERISK TO PSTN**

Interfacing PSTN to Elastix can be done through various methods, as there are various interfaces available for this purpose depending if you are using POTS or ISDN. For most people, interfacing PSTN to Elastix mean connecting the POTS line to the box therefore in this document we will touch on this method.

Generally there are 2 ways of doing this – using an ATA or using interface cards such as the OpenVox A100P and the A400P (there are other cards out there). If you are using the interface card, it has to be physically installed in the PC.

# 14.1 DETECTING NEW HARDWARE

Once the card is installed, fire up Elastix and make sure the card is detected. To do this, select the following: System  $\rightarrow$  Hardware Detection

elast	tix System PBX Fax Email IM Reports	About El	astix 1.0-13 Locout
System Info Network	k   User Manager ant   Load Module   Shutdown   Hardware De	ction Updates Backup/Restore Preferences	0 0
🚲 System Reso	ources		
CPU Info:	GenuineIntel Pentium III (Katmai)	Simultaneous calls, me	mory and CPU
Uptime:	1 day, 13 hour(s), 13 minute(s) 2	1.0	360
CPU usage:	4.07% used of 601.41 MHz	0.5	8         340         Sin. calls           6         320         CPU usage (Z)           9         Pen, usage (R)
Memory usage:	81.18% used of 376.52 Mb		g 1 <sub>200</sub>
Swap usage:	0.00% used of 760.88 Mb	201 231 232 232 232 232 232 232 232 232 23	ភ

In the following screen, click on Detect New Hardware. Elastix will then detect the hardware installed.

System PBX Fax	Email IM Reports Extras	About Elastix 1.0-13 Logout
System Info   Network   User Management   Load Module	Shutdown Hardware Detection Updates Dackup/Restore Pr	references 💿 📀
Image Hardware Detection		
Replace file zapata.conf     Detect New Hardware		
PCI Slot # 1: WCFXO/0 "Wildcard X100P Board 1"		
Port # 1		
FXS		
Active		
	Elastix is licensed under GPL by PaloSanto Solutions. 2006, 2007.	

## A100P card installed

	STIX	m PBX Fax	Email IM	Reports Extras	1		About Elastix 1.0-13	Logout
System Info Nets	vork User Managen	nent   Load Modu	e Shutdown H.	ardware Detection	Updates	Backup/Restore	Preferences	0 0
📷 Hardware	Detection							
Replace file za	pata.conf Detect	New Hardware						
PCI Slot # 1: W	/CTDM/0 "Wildcar	rd TDM400P RE	V E/F Board 1"					
Port # 1	Port # 2	Port # 3	Port #4					
Active	Active	Active	Unknown					
			Elastix is licens	ed under GPL by Pa	iloSanto So	lutions. 2006, 2007.		

## A400P with 2 FXO and 1 FXS modules installed

Elastix Without Tears

#### 14.1.1 Changes for ZAPTEL users

A new issue has raised its ugly head since Elastix 0.9. While the Hardware Detection will detect your FXO card, asterisk will not be able to use this hardware for incoming and outgoing calls. This is due to the <code>zapata-channels.conf</code> was not included in the <code>zapata.conf</code>.

(Note: Ignore this issue if you are using the latest Elastix or using DAHDI instead of ZAPTEL)

To remedy this, you have to add the following lines at the end of <code>zapata.conf</code>:

```
#include zapata-channels.conf
#include zapata_additional.conf
```

You may also need to edit another file */usr/sbin/genzaptelconf* and add the following line in to overcome the ZAP channel handling difficulties that many people are experiencing.

```
# If set: no context changes are made in zapata-channels.conf
#context_manual=yes
context_lines=from-pstn # context into which PSTN calls go
context_phones=from-internal # context for internal phones calls.
context_lines=from-zaptel <---- add this line</pre>
```

Reboot after this is done.

#### 14.1.2 Changes for DAHDI Users

DAHDI users need to edit the .conf file *etc/chan\_dahdi.conf* and if needed to add the following line to overcome the ZAP channel handling difficulties that many people are experiencing where calls that come through the PSTN line cannot be selectively directed to specific destinations.

[trunkgroups]

```
[channels]
;context=from-pstn <--- comment out this line
context=from-zaptel <--- add this line
signalling=fxs_ks
rxwink=300 ; Atlas seems to use long (250ms) winks
usecallerid=yes
useincommingcalleridonzaptransfer=yes <--- add this line if required
hidecallerid=no
callwaiting=yes
usecallingpres=yes</pre>
```

## 14.2 A100P FXO PCI CARD

Digium does not make the X100P card any longer, but there are many X100P compatible in the market – some good and some not so good. The cards I am using are the A100P from Openvox. The 3 A100Ps I use for testings and POCs (in different boxes of course) have been working flawlessly for the last 7 months; unfortunately, since Asterisk no longer will be supporting this types of card officially, Openvox has stopped its production.

This card allows you to connect a POTS (plain Old Telephone System) line to your Asterisk box.



OpenVox A100P Card

# 14.3 CREATE A ZAP TRUNK

Next, if you have an A100P, A400P or TDM400 with 1 FXO module, go into the freePBX web interface to create a trunk and you will notice that there is already a trunk called ZAP/g0. You need to edit this.

- 1. Enter the **phone number** for you pots line in the **Caller ID field**
- 2. Enter 1 for Maximum channels
- 3. Set a dial rule you want for this trunk
- 4. Select an outbound dial prefix to select this trunk when dialling
- 5. Set the **Zap Identifier** to 1 (replace the default g0 with 1)

Linembedded freePBY	Edit ZAP Trupk			Add Trunk
insic asic	Edit ZAP Trunk			Trunk ZAP/1
Extensions	Delete Trunk 1			Trunk IAX2/NetVoice
Feature Codes				Trunk SIP/ostn-1
General Settings	In use by 2 routes			Trunk SIP/ostn-2
Outbound Routes	General Settings			Trunk SIP/Oztell
Trunks	Serierar Serangs			Touck SIP/Reported
pound Call Centrol	Outbound Caller ID	Family Health Centre		Strong StryPennyter
Inbound Routes	Never Override CallerID			
Announcements	Maximum channels	1		
Follow Me	Disable Trunk	Disable		
	Disable frank.	LD/IS-NDIRE		
VR	Monitor Trunk Failures	DEastile		
IVR Misc Destinations	Monitor Trunk Failures	Enable		
IVR Misc Destinations Queues	Monitor Trunk Failures	Enable		
IVR Misc Destinations Queues Ring Groups	Monitor Trunk Failures	Enable		
IVR Misc Destinations Queues Ring Groups Time Conditions	Monitor Trunk Failures	Enable		
IVR Misc Destinations Queues Ring Groups Time Conditions wroat Options & Configuration	Monitor Trunk Failures Outgoing Dial Rules Dial Rules	Enable		
IVR Misc Destinations Queues Ring Groups Time Conditions ternal Options & Configuration Conferences	Monitor Trunk Failures Outgoing Dial Rules Dial Rules	Enable		
IVR Misc Destinations Queues Ring Groups Time Conditions Inter Conditions Conferences Misc Applications	Monitor Trunk Failures Outgoing Dial Rules Dial Rules	Enable		
IVR Misc Destinations Queues Ring Groups Time Conditions Fine Conditions Conferences Misc Applications Music on Hold	Monitor Trunk Failures Outgoing Dial Rules Dial Rules	Enable		
VR Misc Destinations Queues Ring Groups Time Conditions email options & Cempuration Conferences Misc Applications Music on Hold 7IN Sets	Monitor Trunk Failures Outgoing Dial Rules Dial Rules	Enable		
IVR Misc Destinations Queues Ring Groups Time Conditions Imac Applications Misc Applications Misc Applications Misc Applications PIN Sets Paging and Intercom	Monitor Trunk Failures Outgoing Dial Rules Dial Rules	Ceen & Remove duplicates		
IVR Misc Destinations Queues Ring Groups Time Conditions menal Optimus & Centparable Conferences Misc Applications Music on Hold 1/K Setts Paging and Intercom Parking Lot	Monitor Trunk Failures Outgoing Dial Rules Dial Rules Dial rules wizards	Cean & Remove duplicates		
IVR Misc Destinations Queues Ring Groups Time Conditions main Optime & Comparation Conferences Misc Applications Misc Applications Misc Applications Misc Sets Paging and Intercom Parking Lot System Recordings	Monitor Trunk Failures Outgoing Dial Rules Dial Rules Dial rules wizards Outbound Dial Prefix	Clean & Remove duplicates [pick mea)		
IVR Misc Destinations Queues Ring Groups Time Conditions france Conditions france Conditions Music Conditions Music on Hold Phy Sets Paging and Intercom Parking Lot System Recordings minthe Access	Monitor Trunk Failures Outgoing Dial Rules Dial Rules Dial rules wizards Outbound Dial Prefix	Clean & Aemore duplicates	ı.	
IVR Misc Destinations Queues King Groups Time Conditions mana Quitows & computation Conferences Misc Applications Music on Hold PIN Sets Paping and Intercom Parking Lot System Recordings minute Access Caliback	Monitor Trunk Failures Outgoing Dial Rules Dial Rules Dial rules wizards Outbound Dial Prefix	Clean & Remove duplicates		
IVR Misc Destinations Queues Ring Groups Time Conditions frime Conditions frime Conditions Music Conductors Music on Hold Phy Sets Paging and Intercom Parking Lot System Recordings minter Access Caliback DISA	Monitor Trunk Failures Outgoing Dial Rules Dial Rules Dial rules wizards Outbound Dial Prefix Outgoing Settings	Clean & Remove duplicates		
VR Misc Destinations Queues Ung Groups Time Conditions mail optime a computation conferences Misc Applications Music on Hold VIM Sets Saping and Intercom Parking Lot System Recordings mise Access allback DISA	Monitor Trunk Failures Outgoing Dial Rules Dial Rules Dial rules wizards Outbound Dial Prefix Outgoing Settings	Change the ide	ntifier	

Once the card is configured, you must add a route for **Incoming Calls** or asterisk will not answer this line

Click on **Incoming Calls** in freePBX and set up an incoming route. To make outbound calls you will need to set an outbound route as well.

You may need to edit the following files *zapata.conf*, *zapata-channels.conf*, *zaptel.conf* and *modprobe.conf*. The last 2 files live in the *letc* directory – use a text editor (I use nano) to edit them.

**14.3.1 zapata.conf** (*for ZAPTEL users only*) Under [channels] change the following lines to reflect the following values:

```
[channels]
context=from-zaptel < --- changed from context = from-pstn
busydetect=yes
busycount=6
```

; This is to ensure that the line is hung up if there is no answer or busy after 6 busy tones)

;echotraining = yes <--- make sure this is commented out
;echotraining = 800 <--- make sure this is commented out</pre>

At the bottom of this file ensure that the following 2 lines exist if not add the following (otherwise your ZAP will not work):

#include zapata-channels.conf
#include zapata\_additional.conf

For my A100P (for A400P or TDM400, you may have other values that will work well for your installation) installation to function correctly, I have also changed the following setting to obtain a good compromise on volume/echoing:

rxgain=4(you may have to experiment with this setting for both A100P and A400P)txgain=2(you may have to experiment with this setting for both A100P and A400P)

### **14.3.2** zapata-channels.conf (for ZAPTEL users only)

Ensure the following exist in <code>zapata-channels.conf</code>. It is located at the end of the file.

context=from-zaptel ; (change context=from-pstn to context=from-zaptel)
group=0
channel => 1

Check it for each channel. Leave the rest of the file as it is.

#### **14.3.3 zaptel.conf** (for ZAPTEL users only)

This file is in the /etc directory. Change the loadzone and defaultzone to "au" (*if you are in Australia. Other countries have different identifiers.* 

# Global data
loadzone = au
defaultzone = au

#### 14.3.4 Changes for DAHDI users

For DAHDI users, all the changes that may be required in the sub chapters above may be done in *etc/chan\_dahdi.conf.* 

## 14.4 OPENVOX A400P FXO/FXS CARD

Like the Digium X100P, this card allows you to connect a POTS (plain Old Telephone System) line to your ELASTIX box. Unlike the X100P or OpenVox A100P, this card has 4 module ports that can be loaded with FXS or FXO modules. **Channel 1** is the **top RJ-45** on the back of the TDM400P card. The illustration below is the OpenVox A400P card and 2 fxo modules. This card is 100% compatible with the Digium TDM400 and is very reliable.



## 14.4.1 zapata-channels.conf (for ZAPTEL users only)

Next, using config edit, look in the *zapata-channels.conf* file and you will see a list of all your channels in your A400P. Set up the trunks as trunks and the extensions as extensions in freePBX.

For your information only, when you open the file, it will look something like the illustration below. For **DAHDI** users this information is found in *etc/dahdi-channels.conf.* 

; Autogenerated by /usr/local/sbin/genzaptelconf -- do not hand edit ; Zaptel Channels Configurations (zapata.conf) ; This is not intended to be a complete zapata.conf. Rather, it is intended ; to be #include-d by /etc/zapata.conf that will include the global settings ; ; Span 1: WCTDM/0 "Wildcard TDM400P REV E/F Board 1" (MASTER) ;;; line="1 WCTDM/0/0" signalling=fxs\_ks callerid = asreceivedgroup=0 context=from-zaptel channel => 1context=default ;;; line="2 WCTDM/0/1" signalling=fxs\_ks callerid = asreceivedgroup=0 context=from-zaptel channel => 2context=default ;;; line="3 WCTDM/0/2"

If in the illustration it shows channel 1 is your Zap extension then add a zap extension for channel 1 in Elastix or freePBX and if it shows your Zap trunk is channel 2 you should create a zap trunk for channel 2 in Elastix or freePBX.

Once this is done, reboot your PC and when Asterisk starts, use the GUI to add a route for incoming calls or asterisk will not answer your trunk. Similarly, to make outbound calls you will need an outbound route. Set them up as per setting up routes in the earlier chapters of this document.

If you have this card installed, you may need to edit the following files: *zapata.conf* and *zaptel.conf* as per the A100P card in the previous section.

#### 14.4.2 modprobe.conf

You may need to edit the */etc/modprobe.conf* to add the necessary option for usage in Australia (don't bother doing it for A100P or X100P as it will have no effect).

You may use nano or WinSCP3 to edit /etc/modprobe.conf. Simply add the line in red at the bottom of the file as illustrated below.

alias eth0 natsemi alias snd-card-0 snd-intel8x0 options snd-card-0 index=0 install snd-intel8x0 /sbin/modprobe --ignore-install snd-intel8x0 && /usr/sbin/alsactl restore >/dev/null 2>&1 || : remove snd-intel8x0 { /usr/sbin/alsactl store >/dev/null 2>&1 || : ; }; /sbin/modprobe -r --ignore-remove snd-intel8x0 alias usb-controller uhci-hcd options wctdm opermode=AUSTRALIA

(If you happen to be using the newer crop of Digium TDM410P, the swich will be options wctdm24xxp opermode=AUSTRALIA).

Once this is done, execute the following command.

modprobe wctdm

Reboot if there is no error,

reboot

To ensure that you have the right mode operating, do the following at the command line:

grep "AUTO FXO" /var/log/messages

or

dmesg

You should see something like this:

Mar 10 13:30:09 asterisk1 kernel: Module 0: Installed -- AUTO FXO (AUSTRALIA mode) Mar 10 13:30:09 asterisk1 kernel: Module 1: Installed -- AUTO FXO (AUSTRALIA mode) Mar 10 13:30:09 asterisk1 kernel: Module 2: Installed -- AUTO FXO (AUSTRALIA mode) Mar 10 13:30:09 asterisk1 kernel: Module 3: Installed -- AUTO FXO (AUSTRALIA mode)

If you see an FCC mode then you will have an impedance mismatch.



As of Zaptel Drivers 1.2.4, by selecting **opermode=AUSTRALIA** the zaptel drivers automatically add the **"boostringer=1, fxshonormode=1**".

## 14.5 IS THE A100P/A400P RECOGNISED BY ZAPTEL

At the command line, run lspci and amongst others, you should see the following;

00:09.0 Communication controller: Motorola Wildcard X100P

Determine if there is any conflict with other PCI hardware. If there is, you probably will need to replace or use another motherboard. If there is no conflict, then you will have to re-detect the card.

Once done, run zttool on the command line to ensure the card is functioning.

To confirm that zaptel is actually running, after reboot, log in to Asterisk and at the command prompt issue the following command to configure zaptel.

genzaptelconf -d -s -M -F

Then issue the following command;

modprobe wcfxo

If there is no error then your card has been detected. Prove it by issue the following command:

dmesg

You should see the following in amongst the messages;

wcfxo: DAA mode is 'FCC' Found a Wildcard FXO: Wildcard X100P

After that restart amportal just to be doubly sure

Amportal restart. (genzaptelconf -d - s - M - F above should have restart the card anyway)

Using your browser on another PC, connect to ELASTIX and using the Asterisk Info option on unembedded freePBX, check to see if zaptel is loaded.

Chances are, Zaptel will be found.

# 14.6 CALLER ID (CID) USING ZAP DEVICE

The illusive CID has been a talking point and hair pulling exercise for many users. It has always been the case of "I have my ZAP trunk working fine but it is not sending the CID to my Softphone or SIP Phone".

The simple answers to this:

- In Australia, for you to get Caller ID is actually having to apply to your Telco to have it activated on your line – this is not activated by default by your Telco (Optus or Telstra). If caller ID is not activated on your line, you will not get CID.
- If you already have caller ID activated and still don't get caller ID, it's time to have a look at the ZAP configuration file. Typically you may have to look at **zapata.conf** and/or **zapata-channels.conf**.

Naturally if you are using an X100P clone for your FXO, the following exercise will probably be futile as the X100P, even at the best of time will probably not handle the CID properly. However if you are using a more substantial device such as the A400P or TDM400, then this may possibly help you towards getting the CID passed correctly.

(**caveat:** This solution may not necessarily work for your installation but you have nothing to loose if you try)

A participant (Shane B) in the Whirlpool Forum provided the solution below that will correctly pass the CID to Asterisk and your telephones:

You may need to set the following switches in your **zapata.conf** and/or **zapata-channels.conf**.

**14.6.1** zapata.conf (for ZAPTEL users only)

```
usecallingpres=yes
callwaitingcallerid=yes
threewaycalling=yes
usecallerid=yes
hidecallerid=no
relaxdtmf=yes
adsi=yes
;echotraining = yes <--- need to be commented out
;echotraining = 800 <--- need to be commented out
sendcalleridafter=2 ;you may need to add this switch
hanguponpolarityswitch = yes (may have to ask your telco for the polarity reversal)</pre>
```

## **14.6.2** zapata-channels.conf (for ZAPTEL users only)

The following switches may need to be added to the existing ones.

useincommingcalleridonzaptransfer=yes

After the above are done, restart Elastix:

If you are lucky, it should work. If not, you haven't really lost anything apart from the CID.

## 14.7 SIPURA SPA3000/SPA3102 AS A PSTN INTERFACE

To those new to the SPA3000/SPA3102, there is a <u>simplified installation and</u> <u>configuration instruction</u> by JMG Technology. While it is directed mainly at standalone ATA users, it gives a good insight of the Sipura SPA3000/SPA3102's capabilities.

I have come across a few people in the various forums wanting to use their Sipura SPA-3000s as FXO front-end to their ELASTIX boxes. To help them in their endeavours, I've put the following together, as no one single source of information that I've found so far has a config that would actually work for me.

## 14.7.1 Log in to SPA3000/SPA3102

Login to your SPA-3000/SPA3102 as admin/advanced.

Before you change anything, I'd suggest taking a snapshot (i.e. just save the .html page) of your current SPA3000/SPA3102 configuration, just in case you ever need to refer back to your own customisations.

If you're not already running the latest SPA3000/SPA3102 firmware, then upgrade it to the latest version. Take another snapshot for good measure. Nothing should have changed in your settings, except that you have a few extra options that you didn't have before.

Now reset SPA-3000/SPA3102 back to factory defaults, because I'm only going to list the minimum changes required to keep things simple. Take another snapshot now too, in case you ever want to know what the defaults were.

## 14.7.2 Change the settings

## System tab

DHCP: No Static IP: something on your local subnet e.g.; 192.168.1.200 NetMask: 255.255.255.0 Gateway: your router's IP address e.g.; 192.168.1.254 Primary DNS: your ISP's primary DNS address e.g.; 203.12.160.35 Secondary DNS: your ISP's secondary DNS address e.g.; 203.12.160.36

## **Regional tab**

Dial Tone: 400@-19,425@-19;10(\*/0/1+2) Busy Tone: 425@-10;10(.4/.4/1) Reorder Tone: 425@-10;10(.2/.2/1) Ring Back Tone: 400@-19,425@-19,450@-19;\*(.4/.2/1+2+3,.4/.2/1+2+3,0/2/0) Ring 1 Cadence: 60(1.5/3.4) Ring 3 Cadence: 60(1.5/3.4,.4/.2,.4/2,.4/.2,.4/2,.4/.2,.4/2,.4/.2 Hook Flash Timer Max: .13 FXS Port Impedance: 220+820 | |120nF

Delete all the Vertical Service Activation Codes.

#### Line 1 tab

M

Proxy: *IP address of your Asterisk box* e.g.; 192.168.1.100 Register Expires: 60 Display Name: *Whatever* User ID: Asterisk extension number e.g.; 2000 Password: password for that extension Silence Threshold: medium DTMF Tx Method: INFO Hook Flash Tx Method: INFO Dial Plan: (\*\* |\*xx|000|0011xxxxxxx.|0[23478]xxxxxxx|09xxxxxx|1100 |122[135]|1222xxxxxx|12510[12]|12554|1[38]00xxxxxx|13[1-9]xxx |2xx|393xxxxxx|3xxxx.|[4689]xxxxxxx|899060xxxxx.) for example

(\*\*/\*xx./x.) will work, but I like to do a bit of sanity checking, etc.

## PSTN Line tab (method 1)

Proxy: IP address of your Asterisk box e.g.; 192.168.1.100 Register: no Make Call Without Reg: yes Ans Call Without Reg: yes Display Name: No name User ID: PSTN Password: password Silence Supp Enable: no Echo Canc Enable: no Echo Canc Adapt Enable: no Echo Supp Enable: no FAX CED Detect Enable: yes FAX CNG Detect Enable: yes FAX Passthru Codec: G711u FAX Codec Symmetric: no FAX Passthru Method: None DTMF Tx Method: INFO FAX Process NSE: no Dial Plan 1: (S0<:0298765432>) for example - it could be your phone number VoIP Caller Default DP: none PSTN Ring Thru Line 1: no PSTN CID For VoIP CID: yes PSTN Answer Delay: 2 PSTN Ring Thru Delay: 3 PSTN Ring Timeout: 4 PSTN Hook Flash Len: .1 Disconnect Tone: 425@-30,425@-30;1(.375/.375/1+2) FXO Port Impedance: 220+820 | | 120nF On-Hook Speed: 26ms (Australia)

Using method 1 above, it will be necessary for you to create an inbound route using the 0298765432 (or whatever you have defined) as your inbound route DID number. As this incoming call is identified by the (S0<:0298765432>) in the dial plan 1 setting of your SPA3000, this can be directed to any extension by the inbound route.

### PSTN Line tab (method 2)

Proxy: *IP address of your Asterisk box* e.g.; 192.168.1.100 Register: no Make Call Without Reg: yes Ans Call Without Reg: yes Display Name: No name User ID: PSTN Password: leave blank Silence Supp Enable: no Echo Canc Enable: no Echo Canc Adapt Enable: no Echo Supp Enable: no FAX CED Detect Enable: yes FAX CNG Detect Enable: yes FAX Passthru Codec: G711u FAX Codec Symmetric: no FAX Passthru Method: None DTMF Tx Method: INFO FAX Process NSE: no Dial Plan 1:(S0<:s@YourAsteriskIP>) e.g. (S0<:s@192.168.0.101) VoIP Caller Default DP: none PSTN Ring Thru Line 1: no PSTN CID For VoIP CID: yes **PSTN Answer Delay: 2** PSTN Ring Thru Delay: 3 PSTN Ring Timeout: 4 PSTN Hook Flash Len: .1 Disconnect Tone: 425@-30,425@-30;1(.390/.390/1+2) FXO Port Impedance: 220+820 | | 120nF On-Hook Speed: 26ms (Australia)



*Disconnect Tone setting of* **425@-30,425@-30;1(.375/.375/1+2)** *work well for* **Telstra**, *while* **425@-30,425@-30;1(.390/.390/1+2)** *works better with* **Optus.** 

Using method 2 above, you are not required to create an inbound route as the call is directed directly to asterisk and will be handled by the **any DID** / **any CID** catch all inbound route. Through this route, you can direct it to your IVR or any extension you like. This method is simpler if you only have one SPA3000/SPA3102, however if you are using multiple SPA3000/SPA3102, method 1 is the recommended method.

### User 1 tab

Default Ring: 3 Default CWT: 8

14.7.3 Add SIP Trunk

Then in freePBX, add a SIP trunk.

## **General Settings:**

Outbound Caller ID: <0298765432> (for example) Maximum Channels: 1 Dial Rules: 0+NXXXXXXX (for example) 0011+ZXXXXXXXX. Trunk Name: telstra (for example)

#### **Peer Details:**

canreinvite=no context=from-pstn host=the IP address of your SPA-3000/SPA3102 (for example; 192.168.1.200) insecure=very nat=no port=5061 (for example)

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qualify=yes type=peer username=PSTN

**User Context:** telstra-incoming (for example)

## **User Details:**

```
canreinvite=no
context=from-pstn
host=the IP address of your SPA-3000/SPA3102 (for example; 192.168.1.200)
insecure=very
nat=no
port=5061 for example
type=user
username=PSTN
```

Leave "Register String" empty

## 14.7.4 SPA3000/SPA3102 as an outbound PSTN Trunk

To make PSTN calls using the SPA3000/SPA3102 you simply select the SPA3000/SPA3102 trunk, which you have named *Telstra* (*see trunk name that you have given to this trunk above*) as the first choice in your outbound route.

Also see Eliminating echo problems in Appendix E.1 in Sipura SPA-3000/SPA3102

## 14.8 SPA3000 REMOTE MANAGEMENT

The SPA3000 can be managed remotely. To enabled remote management the following must be set as per the illustration below:

A Division of Cisco Syst	YS <sup>®</sup> ems, Inc.	Linksys Phone Ade	apter Configuration
Info System SIP Provi	isioning Regional Line 1	PSTN Line User 1 PSTN	User User Login basic   advanced
System Configuration			
Enable Web Server:	ves 💙	Web Server Port:	91
Enable Web Admin Access:	ves 💙	Admin Passwd:	mypassword
User Password:	,05		(ii) passing i
Internet Connection Type DHCP:	no 💌		
Static IP:	192.168.1.91	NetMask:	255.255.255.0
Gateway:	192.168.1.1		
Optional Network Configurat	tion		
HostName:	LinkSysSPA3K	Domain:	Mesra.PABX
Primary DNS:	192.168.1.1	Secondary DNS:	
DNS Server Order:	Manual 💌	DNS Query Mode:	Parallel 💌
Syslog Server:		Debug Server:	
Debug Level:	0 💌	Primary NTP Server:	
Secondary NTP Server:			
	Undo All Changes	Submit All Changes	

Naturally you will have to use the appropriate IP addresses, port and your own password etc. The illustration above is for my system.



If you are managing your SPA3000 remotely, you may encounter some difficulties displaying the pages or sometimes it just seems to hang. To solve this problem, you can try changing your TCP setting (I use TCP Optimizer) by setting the RWIN to something smaller eg. 10xMTU, i.e if your MTU is 1500, try setting your RWIN to 15000.

## 14.9 Use STUN Server on SPA3000/SPA3102

To use a STUN server with the SPA3000, simply select the SIP tab on Admin Admanced mode and make the following changes. I use stun.xten.com, you may use any STUN server of your choice.

- a. Select SIP Tab
- b. Handle VIA received: no
- c. Handle VIA rport: no
- d. Insert VIA received: no
- e. Insert VIA rport: no
- f. Substitute VIA Addr: yes
- g. Send Resp To Src Port: yes
- h. STUN Enable: yes
- i. STUN Test Enable: no
- j. STUN Server: *stun.xten.com* (I use stun.xten.com, you may use whichever stun server you like)
- k. NAT Keep Alive: 15

NAT Support Parameters			
Handle VIA received:	no 💌	Handle VIA rport:	no 💌
Insert VIA received:	no 💌	Insert VIA rport:	no 💌
Substitute VIA Addr:	yes 💌	Send Resp To Src Port:	yes 💌
STUN Enable:	yes 💌	STUN Test Enable:	no 💌
STUN Server:	stun.xten.com	EXT IP:	
EXT RTP Port Min:		NAT Keep Alive Intvl:	15

# 15 DISA

## 15.1 DISA - HAND CODING METHOD

I use this function in my PSTN to VOIP gateway above (method prior to the facility being inbuilt into freePBX).

**DISA** (Direct Inward System Access) allows you to dial in from outside to the Asterisk switch (PBX) to obtain an "internal" system dial tone. You can place calls from it as if they were placed from within.

Normally you can do this via the Digital Receptionist and enter the unannounced option number known only to selected callers.

The caller enters his/her password, followed by the pound sign (#). If the password is correct, the user is then given system dial tone on which an external call may be placed.

This type of access has SERIOUS security implications, and GREAT care must be taken NOT to compromise your security.

The example below was how I implemented my DISA in the older versions of freePBX (and AMP) where the built-in function was not available. Now, this method is no longer used as this function is already included within freePBX.

In my "**MyMenu**" that says, "Hi, you have reached the Sharif's household. Press 1 for Ben, 2 for Rohani, 3 for Norsurya etc..." I have created an extra option – option 4 (best if you use a few digits) that has an "Action" of "Custom App", which reads, "custom-pstn2voip,s,1".

Then in *extensions\_custom.conf*, I have added (do this towards the end of the file).

```
[custom-pstn2voip] < -- you can name it to whatever you like
exten => s,1,Answer
exten => s,2,DigitTimeout(5)
exten => s,3,ResponseTimeout(10)
exten => s,4,Authenticate(12345678)
exten => s,5,DISA(no-password|from-internal)
```

Change "12345678" to whatever digits you want your "DISA password" to be.

The above is one way of doing it (the hard way). The easier way of doing it is through Elastix - see below.

# **15.2 DISA – USING ELASTIX**

With the current versions of freePBX, the same thing can be achieved through Elastix, and simpler as illustrated in the screenshot below.

	System PBX Fax	Email IM Reports	Extras	About Elastix 1.0	)-12 Logout
PBX Configuration Flash Ope	rator Panel   Voicemails   Me	onitoring Echo Canceller	Endpoint Configuration	Conference Extensions Batch Too	s 📀 💿
PBX Configuration         Flash Ope           Oplion         Unembedded freePBX           Basic         Extensions           Extensions         Feature Codes           General Settings         Outbound Routes           Trunks         Inbound Routes           Inbound Call Control         Inbound Routes           Announcements         Follow Me           IVR         Misc Destinations           Queues         Ring Groups           Time Conditions         Internol Options & Configuration           Conferences         Misc Applications           Musics on Hold         PIN Sets           Paging and Intercom         Parking Lot	Add DISA Add DISA DISA name: PIN Response Timeout Digit Timeout Digit Timeout Require Confirmation Caller ID Context Submit Changes	PSTN2VoIP 1234 10 5 from-internal	Endpoint Configuration	Conference   Extensions Batch   Too	Add DISA
System Recordings					
Callback					
DISA					

I have populated the various fields thus:

- **DISA name: PSTN2VoIP** (This is the name I gave to this application)
- **PIN:** *1234* (or whatever you think is appropriate. This pin number will be asked when you want to use this service)
- **Response Timeout:** 10 (time needed for a response from the you in seconds)
- Digit Timeout: 5

The rest you can leave at the default value.

When you set up your digital receptionist, in the DISA dropdown selection, simply select PSTN2VOIP as your DISA application.

If you are using this method, there is a slight catch. When you choose the DISA option to call a number, you will be greeted with "Please enter your password followed by the pound key" and immediately you will get a dialtone. You may start dialling the password followed by # followed by the telephone number.

E.g. If you have chosen to have a password e.g. **12345**, and the number you wish to dial is **0292345678**, you must dial the number thus: **12345#0292345678** without waiting for another dialtone, otherwise it will not dial out correctly and you will be wondering what happened. I hope this will be fixed up soon.

# **16 OPERATOR PANEL**

Operator Panel is a switchboard type application for the Asterisk PBX. Typically, it runs on a web browser. It is able to display information about your PBX activity in real time.

The Flash Operator Panel is the one that comes with ELASTIX.

Apart from Flash Operator Panel, There are a number of operator panels available. One that comes to mind can be found at <u>SOFTPEDIA</u> and another is <u>HUDLite</u> from Fonality. HUD will integrate nicely with ELASTIX and the support for HUD is already integrated within ELASTIX.

# 17.1 FLASH OPERATOR PANEL (FOP)

This is the standard Operator Panel that is already part of ELASTIX – some loved it while some hated it.

The layout is configurable (button sizes and colours, icons, etc). You can have more than 100 buttons active per screen. It also supports contexts: you can have one server running and many different client displays (for hosted PBX, different departments, etc).

It can integrate with CRM software, by popping up a web page (and passing the CLID) when a specified button is ringing.

FOP can be activated by clicking on the PBX tab on the top of the screen and selecting Flash Operator Panel.

PBX Configuration Flash Operator Panel Voicemails Monitoring Lcho Canceller	Extras	Abor Extensions 0	ut Elastix 1.0-12 Jatch   Tools	Loaout
Asterisk Flash Operator Panel			No timeout	
Extensions	Queues		No uneout	1000
2000:	Op General	ŧŤ		
2001 : Ben 2				
2003 : Rossii				
	Conferences			
	General			
	Parking lots	8	Parked Parked Parked	a a a
	Trunks	5	MalaccaPB	O
	Pe SPA3K	O	MicroStylePB	O
	iPrimus	Ô		
	Oztell	Ø		
	Pennytel	Ø		
	SPA3K-	O		
	024744395	O		
	028215061	O		
	6032056533	O		
Elastix is licensed under <u>op</u>	by <u>PaloSanto Solutions</u> . 2006, 2007.			

The following information are displayed on FOP:

- Which extensions are busy, ringing or available
- Who is talking and to whom (CLID, context, priority)
- SIP and IAX registration status and reachability
- Meetme room status (number of participants)
- Queue status (number of users waiting)
- Message Waiting Indicator and count

- Parked channels
- Logged in Agents

Functions you can perform on FOP:

- Hang-up a channel
- Using drag-&-drop to transfer a call
- Initiate calls by drag-&-drop
- Barge in on a call using drag-&-drop
- Set the caller id when transferring or originating a call
- Automatically pop up web page with customer details
- Click-to-Dial from a web page
- Mute/Unmute meet-me participants

For more details refer to: <u>http://www.asternic.org/</u>

## 17.1.1 Setting the Admin Password

It is recommended to change the FOP password to something easy and simple to remember such as the number 2 for example. The simple method is by logging in to your asterisk box either remotely using putty or directly on your box console.

In this example, Putty is used to log in remotely to ELASTIX. Once logged in, change the directory to */etc* 

cd /etc

Using nano as the editor, open the configuration file amportal.conf

nano amportal.conf

Go to the line that says FOPPASSWORD=eLaStIx.2007 (in FOP that comes with ELASTIX, the default password is "eLaStlx.2007")

Replace the "eLaStIx.2007" with the password of your choice.

FOPPASSWORD=cowscomehome

Close off nano and putty. Open your web browser and go to FOP. You should now be able to click on the little lock, put in your password and you will see it lock up. Now you can begin to have some fun.

## 17.1.2 Hang-up a Call

You can end a call by clicking the big red button.

## 17.1.3 Transfer a Call

Simply drag the little telephone over to the new extension and it will transfer the call.

## 17.1.4 Initiate a Call

If you are lazy, like me, not bothered to pick up the phone and dial, you can drag and drop your little telephone to another extension it will start by ringing your phone, when you pick up and then it will ring the other extension. Once the destination phone is picked up it will initiate the call.

## 17.1.5 Barge in or Create a Conference

Say there is a call already going and you want to jump in. Well all you have to do is grab your little phone in the flash panel and drop it into the conversation and it will connect you. Great if you want to see if your employees are making lots of private calls.

# **17 REPORTS**

Reports are essential in the operation of a PABX. Elastix provides this facility through the Reports Tab. Under this tab the following can be addressed;

- CDR Report (report regarding calls made and received)
- Channels Usage
- Billing

# 17.1 CDR REPORT

This report is accessed under the Reports Tab.

2 elastix	System Pbx Fex	Email IN Repor	ts Extras Agenda		About Electric 1.1-	z	Losov
Report Channels Usage	Billing						•
2 COR Report List			$\backslash$				
	et Date: * 04 Aug 2008		1 Field: Dette	ation w	[Filter.]		Delete
E	nd Date: * 04 Aug 2008		Status: ALL	*			and designed
2						and the state of the state	
late	Source	Destination	Src. Channel	Código de Cuenta	Dst. Channel	Status	Duration
008-08-04 11:48:36	2001	3002	SDP/2001-00417520		IAX2/MalaccaFBX-1	ANSWERED	9
008-08-04 11:49:16	0247443957	47313261	S2P/2001-003896a0		S3P//Primus-063ab8a0	ANSWERED	2089
008-08-04 13:29:26	3002	2001	IAX2/MalaccaPBX-1		519/2001-08418460	ANSWERED	436
008-08-04 15:08:43	0247443957	1300301466	53P/2000-06418ab0		53P//Primus-083acd58	ANSWERED	288
	0426473700	12	2ep/2-1		\$1P/2001-003896+0	ANSWERED	83
008-08-04 17:51:95			S0P/2001-003f7b08		IAX2/MalaccaPBX-6	ANSWERED	
008-08-04 17:51:35 008-08-04 23:20:51	2001	3005					

It provides the record of all incoming and outgoing calls including the channels used and duration of calls. There is no special setup required for this facility.

## **17.2 CHANNELS USAGE**

Channels Usage report provides a graphical representation of all the channels used;

elastix System PRX fax [mail 14] Reports fax	nda	About Elastic 1.1-2 Logout
CDR Report Character Stage Billing		• •
in chamies usere report	Simultaneous Channels (Total)	
22	- Total Owerets	
14	Siaultaneous Zap Channels	
2.0 1.0 6.0	Simultaneous SIP Channels	
14 60 60	Simultaneous IAX Channels - 160 Oannels - 5 8 8 8 8 8 8	

Like the CDR Report, there is no special setup or configuration to be conducted for this facility as it is simply an information gathering tool.

# **18 CALL PARKING AND TRANSFER**

Elastix provides this facility through the Parking Lot facility. If you want to allocate other extensions rather than the one pre-configured, simply change the appropriate Parking Lot extension number in the configuration screen. For this purpose we will leave it at the default extension number 70.

**Note:** You may have to increase your digittimeout duration from the default to 2000 depending on the type of phones you use. If you have to do this, insert the following in your features\_general\_custom.conf file.

featuredigittimeout = 2000 ; (you will need to experiment with this value)

	System	PBX Fax Fmail	1	Extras		About Elastix 1.0-12	Logout
PBX Configuration Flash Opera	ator Panel   ¥	picemails Monitoring	Echo Canceller	Endpoint Confi	guration Conference	Extensions Batch Tools	3
Option Unembedded freePBX Basic Extensions Feature Codes General Settings Outbound Routes Trunks Inbound Call Control Inbound Call Control Inbound Call Control Inbound Call Control Follow Me	Parking Lot O Enable Parking Lot Parking Lot Number o Parking Ti Parking Lo	Lot Configu options 2 rking Lot Feature ot Extension: f Slots: meout: ot Context:	45 seconds parkedcal	70 8 V 5			
IVR Misc Destinations Queues Ring Groups Time Conditions Internal Options & Configuration Conferences	Actions for T Parking Al CallerID P Announce	imed-Out Orphans lert-Info: irepend: ment:		None V			
Misc Applications Music on Hold PIN Sets Paging and Intercom Parking Lot System R bordings Remote Access Callback DISA	Ouclears     Ouclears	or Orphaned Parked C General <600> V stinations: Get voicerr te Call: Hangup DRS: <2000> Reception ail: <2000> Reception (ICCS: General <8000> ead this link before do ook Directory: Phone cements: Silent V inoffice V	alls:	(Zork) ¥			
	Submit Cha	inges					

Parking lot is managed through PBX -  $\rightarrow$  PBX Configuration -  $\rightarrow$  Parking Lot.

## **18.1 CALL TRANSFER - MANAGED**

When enabled, a call that is parked will be parked at extension 70+1. Each call parked will be parked at that extension +1 e.g. if you have 3 calls that you need to park, the first call parked will be parked at ext. 71, the next call will be automatically allocated ext. 72 and the next ext. 73 and so on.

This is handy if you want to manage the calls to be transferred. After parking the call, you may call the person you are transferring it to and tell him/her that there is a call on extension 71. The person can then dial 71 and get the parked call.

## 18.1.1 How is it done?

When you answer a call and want to transfer that call, do the following:

- 1. Dial **##70** the call will be transferred to a park extension and the number will be announced to you. In this case it will be 71. After the announcement, you may hang up and dial the extension of the person you wish to transfer the call to and inform him to dial 71 to accept the call.
- 2. While all this is happening, the caller will be listening to the music on hold.
- 3. If for any reason the person refused to take the call, you then hang up and dial 71 to pick up the call back to you and speak to the caller making the usual excuses "*I am afraid Mrs. Roberts, Dr. Roberts is busy with his nurse and his office door is being locked from the inside*" <sup>(2)</sup> and the next time Dr. Roberts will always accept the calls from his wife.

## **18.2 CALL TRANSFER – BLIND**

To do a blind call transfer (where you simply transfer the call and bad luck if the extension does not answer), you simply dial *##* followed by the extension that the call is meant for.

E.g. to transfer to extension 2001, you simply dial **##2001** and hang up.

## 18.3 PUT A CALL ON HOLD

Just like the managed call transfer above, you dial **##70** and after the parked extension announcement, you hang up. This is simply a work around for those that do not have the hold function on the telephone.

When you are ready to take the call, dial 71 (or whatever the park extension is) and the call will be picked up again. While the call is being parked, the caller will be listening to your MOH music.

## 18.4 CALL PICKUP

**Call Pickup** is the ability to pick up a call on a phone that is ringing from other than the ringing phone. E.g. you are sitting at your desk and the phone in the next office is ringing but there is no one there to answer the phone. You then use the phone on your desk to pick up the call.

There are two type of call pickups:-

• Group call pickup

This allows you to pickup a call from any ringing phone that is in the same pickup group as you. However, if there were more than one phone ringing, you would have no control over which call will be transferred to you.

• Directed pickup

This allows you to pickup a call that was made to a specific extension, such as when you are in another office and a phone rings in the general direction of your office. You are wondering if it is your phone that is ringing. In this case you simply dial the pickup number and your extension number. If the ringing phone is your phone, the call will be transferred to you.

Group call pickup is typically invoked by dialling \*\* from another phone in the call pickup group.

You can also change to \*8 or to something else if you wish. see *http://www.voip-info.org/wiki/view/Asterisk+config+features.conf* 

## 18.5 FILTER YOUR INCOMING CALLS - ONLY ACCEPT KNOWN CALLS

(Good for filtering Telemarketers and ex-girlfriends)

For this purpose, the Privacy Manager option of the extension can be invoked, however I found that caller often get confused with the need to enter the 10 digits phone number as often they may have only 9 digits or more than 10 digits.

I have opted for the Nerd Vittles method.

There is a good article on this site <u>http://mundy.org/blog/index.php?p=66</u> relating to filtering incoming calls and also some discussions here <u>http://forums.whirlpool.net.au/forum-replies.cfm?t=418909</u>

This feature will force callers to record their names before the calls will be parked. The extension that you nominate will ring and on answering, you will be informed that there is a call waiting for you in a parked extension e.g. 71.

After receiving the call and if you wish to take the call, hang up and dial 71 and you will be connected to the caller. Otherwise, just hang up and the caller will just have to leave a message in the voicemail. Real nifty

This is how I did it with my Asterisk. Since I am pretty simple myself, I just take the low road make it really simple.

I add the following scripts in my extensions\_custom.conf

```
[from-internal-custom]
; I create this extension for silent Auto attendant 30-10-2005
exten => 111,1,Answer ;
exten => 111,2,Goto(who-r-u,s,1)
```

```
(Note: DO NOT create extension 111 in freePBX. This is only a dummy extension)
```

At the end of the *extensions\_custom.conf*, I added the following 2 blocks of extra codes.

; experimental as per above 30-10-2005

```
[who-r-u]
exten => s,1,Background(privacy-unident)
exten => s,2,Background(vm-rec-name)
exten => s,3,Wait(2)
exten => s,4,Record(/tmp/asterisk-stranger:gsm|5|15)
exten => s,5,Background(pls-hold-while-try)
exten => s,6,Goto(parkedcalls,70,1)
exten => s,7,VoiceMail(2001@default)
exten => s,8,Playback(Goodbye)
exten => s,9,Hangup
```

```
[parkedcalls]
exten => 70,1,Answer
exten => 70,2,SetMusicOnHold(default)
exten => 70,3,ParkAndAnnounce(asterisk-friend:/tmp/asterisk-
stranger:vm-isonphone:at-following-
number:PARKED|40|local/2001@from-internal|who-r-u,s,7)
exten => 70,4,Hangup
```

(\*The codes in red must be in one continuous line)

I forward all calls that were parked to my existing extension **2001** (in **bold type**) and direct voicemail for unanswered calls to the same extension as well.

Next I create a new extension using freePBX e.g. 222. Using a softphone, I connect to extension 222 and unconditionally forward all calls to extension 222 to extension 111 by dialling \*72111

I then direct all my incoming calls from my PSTN trunk to extension 222.

The result is, when there is an oncoming call from PSTN, the call will be sent to extension 222 and in turn, the call will be forwarded to extension 111.

When the call gets to extension 111, the caller will hear a message saying that, "*unrecognised calls are not accepted and asking the caller to record his/her name after the tone*" (or something to that effect). After the caller has complied and press the # key, the call will be parked.

Extension 2001 will be called by Asterisk. When you answer the call, Asterisk will announce that there is a call by *<name as recorded>* on extension 71 (or 72 depending on how many calls are being parked). If you recognize the name recorded, or if you want to take the call, simply hang up and dial 71. You will be connected to the caller. While this is all happening, the caller will be put on hold listening to your MOH.

If however you do not wish to accept the call, you simply hang up and do not dial 71. The caller will be asked to leave a message instead.

# **19 CONFERENCE**

Tele-conference can be configured through FreePBX Conference option.

Conference is pretty easy to set up, as there is very little to do. However, to set up Conference, a Zaptel card may be required (at least in my case as I don't seem to be able to get it working without a Zaptel card installed). In the latest Elastix 1.0, this seems to have been resolved.

The Conference function of Asterisk is similar to a Tele-conference call where multiple callers can call in and participate in a two-way conference like in a party room where everyone can talk and listen to one another or just to listen to a Tele-presentation.

In Elastix, there are 2 .conf files that relates to conferencing. They are:

```
meetme.conf
meetme_additional.conf
```

#### meetme.conf

Ensure that you have the following 2 lines (only 2 lines in the file):

```
[rooms]
#include meetme_additional.conf
```

All these lines do is point to the *meetme\_additional.conf* where the conference room extensions are being created when you create a conference room through freePBX conference setup option.

## meetme\_additional.conf

This is where the extensions for the meeting rooms are created. One line per room. You may have just one extension for your meeting room or you may have several meeting rooms. In my case, I have 1 meeting room and therefore my *meetme\_additional.conf* contains the following extension:

```
; Extensions for my meeting rooms conf => 8200 |
```

## **19.1 CREATE CONFERENCE**

Elastix allows you to create one or more conference room through the **PBX** -> **PBX Configuration** -> **Conference** screen. All you have to do is fill in the appropriate fields on the screen and you are on your way to a voice conference.

Naturally there are some options that you may wish to have for the conference room. They are entirely up to you. The main important things are for you to create the **conference room number** and the **conference name** for you to know what that conference room is for. The rest of the fields are optional.

Below is what my conference configuration page looks like:



- This conference room number is 8000
- I call this conference room General
- Users are not required to enter a PIN number to join the conference.
- Admin PIN number is **not required** for the conference to wait for the leader to conduct the conference.
- I don't have a joining message greeting when someone joins the conference.
- I have **No** in the leader wait option for the conference to wait for the conference chairman.
- I require a user count so I have **Yes** in the user count field.
- I need to be informed when a user joins or leaves the conference. I have Yes here.
- The music on hold is also flagged as **Yes** as I want music on while waiting.

Once these are done, from one of your phones dial, 8000 and you will be able to join the conference.

You are now in the conference room and every time someone joins or leaves the conference, it will also be announced.

To test this, go to another phone and also dial 8000. You should hear an announcement (if you are listening on the first phone), indicating that someone has joined the conference.

If there are 2 (or more) of you, you will be able to talk to each other. If one of you hangs up, you will know about it.

## **19.2A ELASTIX CONFERENCE MANAGER**

The Elastix developers has included a conference control facility that enables the conference moderator to control the conference where people can be invited into the conference and participants can be silenced or kicked off the conference. If you have ever moderated a teleconferencing you would have realised how handy this facility is.

To set up this facility, under the PBX tab, you will notice the sub-tab "Conference". Click on the conference sub-tab and you will be presented with the screen as illustrated.

elastix recome to communicate	BX Fax Email IM Reports E	xtras 1	About Elastix 1.0-14	Logout
PBX Configuration   Flash Operator Panel   Voic	cemails   Monitoring   Echo Canceller   E	Endpoint Configuration Conference	Extensions Batch   Tools	00
The	e conference installation is almo Elastix is licensed under <u>GPL</u>	ost done. To complete it plea , by <u>PaloSanto Solutions</u> . 2006, 2007.	se <u>click here</u>	

To activate this module, it is necessary to click on the "click here" link.

Once fully installed and activated, you will be presented with a new screen where you can start populating the conference you want to control.

Click on the new conference button to create a new conference.

	stem PBX Fax	Email IM Reports	Extras		About Elas	tix 1.0-14	Loqout
PBX Configuration   Flash Operator P	anel Voicemails N	tonitoring Echo Canceller	Endpoint Configuration	Conference	Extensions Batch	Tools	0 0
💰 Conference							
New Conference		State	Current Conferences		Filter:		Show
	<				I <sup>e</sup>	/ Start / Previous (0 -	0 of 0) Next  > End  >
Delete Confere	nce #	Conference Name		Starts	Ends	Participants	
						Start / Previous (0 -	0 of 0) Next  > End  >
		Elastix is licensed under <u>G</u>	PL by PaloSanto Solutions.	2006, 2007.			

## 19.2.1 To control Call-In voice conference

The following screen will appear. In this case we want to control the general conference 8000 that we created earlier on.

	About Elactiv 1.0-14 System PBX Fax Email IM Reports Extras	Logout
PBX Configuration   Flash Operato	ar Panel   Yoicemails   Monitoring   Echo Canceller   Endpoint Configuration   Conference   Extensions Batch   Tools	0 0
Save Cancel		<ul> <li>Required field</li> </ul>
Conference Name: *	General Conference	
Conference Owner:	Ben Sharif	
Conference Number: *	BODD Change this number to the call in	
Moderator PIN:	1234 conference that we created earlier on.	
Moderator Options	🗹 Announce 🗹 Record	
User PIN:	123	
User Options:	Announce 🔲 Listen Only 🗹 Wait for Leader	
Start Time (PST/PDT): *	2008-04-07 12:37	
Duration (HH:MM):	1:0	
Max Participants: *	10	

In the fields, we enter the following:

Conference Name:	General Conference
Conference Owner:	<your name=""></your>
Conference Number:	Replace the pre assigned number to 8000
Moderator PIN:	If you require one, enter it here e.g 123
Moderator Option:	You may put check marks on the option that you choose
User PIN:	If you require a valid PIN to join the conference
User Option:	Select the user option you want.
Start Time (PST/PDT):	From the Calender icon you can choose the date and time.
Duration (HH:MM):	You can enter the duration of this conference.
Max Participation:	The maximum participation of this conference.

Participants of this conference can call Extension 8000 and join the conference.

## 19.2.2 Invite only voice conference

Similar screen to the above example will appear. In this case we the conference will be an internal conference by invitation only.

You will need to populate the fields as per the example above except that you may not have a pre-created conference number. The conference will be nominated by the system and you do not change this number to any pre-created conference that was done using freePBX.

	System PBX Fax Email	IM Reports Extras		About Elast	tix 1.0-14	Logout
PBX Configuration   Flash Operato	or Panel   Voicemails   Monitoring	Echo Canceller Endpoint Con	figuration Conference	Extensions Batch	Tools	0 0
Conference Save Cancel Conference Name: * Conference Owner: Conference Number: * Moderator PIN: Moderator Options User Options: Stort Time (PST/PDT): * Duration (HI:NM): Max Participants: *	Marketing Promotion Ben Sharf 44449 Announce Record Announce Listen Only 2006-04-07 13:07 1 : 0 10	Do not change f     Wait for Leader	this number			* Required field

Since the conference is by invitation, we only need to fill in a few fields.

Conference Name:	Marketing
Conference Owner:	<your name=""></your>
Conference Number:	Leave it as it is.
Moderator Option:	You may put check marks on the option that you choose
User Option:	Select the user option you want.
Start Time (PST/PDT):	From the Calender icon you can choose the date and time.
Duration (HH:MM):	You can enter the duration of this conference.
Max Participation:	The maximum participation of this conference

When you click save, you will have created a controllable conference.

## **19.2B CONTROLLING AND MODERATING THE CONFERENCE**

Once the above tasks is completed, you may start moderating the conference.

	System	PBX Fax Email IM	4 Reports Extras	About Elastia	x 1.0-14 Logout
PBX Configuration	Flash Operator Panel	Voicemails Monitoring E	cho Canceller Endpoint Configuration	Conference Extensions Batch	Tools 🛛 😌 🧿
🔬 Conference	e				
New Confere	ince		State: Current Conferences 💌	Filter:	Show
				PK 3	Start < Previous (1 - 1 of 1) Next  > End  >
Delete	Conference #	Conference Name	Starts	Ends	Participants
	44449	Marketing Promotion	2008-04-07 13:07:00	2008-04-07 14:07:00	0/10
i					

The Elastix team has taken all the hardwork away from you so you can control and moderate the conference with relative ease

Click on the participant as shown in the illustration above to start administering the conference (see arrow).

	System	PBX Fax Email	IM Reports	Extras		About Elast	ix 1.0-14	4 Logout
PBX Configuration	Flash Operator Panel	Voicemails Monitoring	Echo Canceller	Endpoint Configuration	Conference	Extensions Batch	Tools	0 0
💰 Conferer	ice				_ 1			
Update Id Ca	2 Cancel	Invite Caller Duration Elastix i		> CR003> <2003> <2003> <2004> <2004> CR003> CR004 CR04 CR004	2006, 2007. 2	ha. Ita	4 - Start /	Kick All Previous (1 - 0 of 0) Next > End P Kick Profiles (1 - 0 of 0) Next > End P 3

Here you can do the following:

.

- 1. Inviting people to join the conference through the drop-down list.
- You may mute any participant by clicking the mute button
   Eject any participant using the kick button
- 4. Terminate the conference by using the Kick All button.

The conference will automatically reset anad terminate at the end of the period you have determined at the time of setting up.

# 20 VOICEMAIL



If you have enabled voice mail when setting up your extension, and you hear a stutter tone (not the normal dial tone) when you initially pick up your phone, it is indicating to you that you have a voice mail waiting.

To enable voicemail on an extension simply "enable" it when you create the extensions from the Elastix GUI. If you require email notification of your voicemail, you may enter your email address in the email address field.

Click submit when done followed by the red bar on top of the screen.

## **20.1 ENABLING EMAIL NOTIFICATION OF VOICEMAIL**

Asterisk can send email to you informing you that you have voicemail waiting in your mailbox. In fact it can even send you the voicemail itself in the form of a .wav file if you so desire.

I was having a lot of fun trying to get email notification going without having to resort to SASL. Finally I was able to get it to work after a few trial and errors. MY SMTP Relay host is expecting mail from a valid DNS before it will accept the mail, but Postfix is sending the email from Asterisk@localhost therefore it was rejected.

To overcome this, I have to edit the /etc/postfix/main.cf file to include the following;

After having done that, you may do one of the following methods:

## 20.1.1 Method 1

This is a desperate method (*not recommended – do only on desperation*) where I have to change my Elastix hostname from *elastix.local* default to a fully qualified DynDNS name using the GUI as per the illustration below.



Select System Tab Select Network Option Click on Edit Network Parameter

Change your host name to *Elastix.<Your\_Dyndns\_Address>*. In my case, my DynDNS address is *netvoice.dyndns.org* therefore in this field I entered *elastix.netvoice.dyndns.org*.

elast	ix	System	PBX Fex	Email	Reports	Extras	1		Abs	out Elastix, 1.0-14	Logout
System Infe Network	User	Management	Load Module	Shutdown	Hardware D	etection	Updates	Backup/Restore	Preferences		• •
Network Parameters DHCP Server		Save Car	ork Paramete	205		_					* Required field
	1	Host (Ex. host.exampl Default Gate	le.com): • Ine way: • [19]	tvoice.dyndni 2.168.1.1	org 🖌			Primary DNS: * Secondary DNS:		192.168.1.1	
					Elastix is	licensed u	nder OPL	y PaloSanto Solutio	ns. 2006, 2007	7.	

## 20.1.2 Method 2

This method does not require you to change your Elastix host name like I did above. Instead locate the "myorigin = \$myhostname" in /etc/postfix/main.cf and change it as I have done below then save it and restart postfix.

# For the sake of consistency between sender and recipient addresses, # myorigin also specifies the default domain name that is appended # to recipient addresses that have no @domain part. # #myorigin = \$myhostname myorigin = netvoice.dyndns.org <--- Change this to reflect your DynDNS address</pre>

# RECEIVING MAIL

# The inet\_interfaces parameter specifies the network interface

When Postfix send the mail to your Relay Host, it will show that the mail is from *Asterisk@netvoice.dyndns.org* which is a valid address and not *Asterisk@localhost* which is, as far as the Relay Host is concerned is an invalid address.

If things go wrong with your email notification, you can look at the *maillog* file in the */var/log* directory.

The above methods are just two quick and dirty ways of getting your email out. In the Elastix Mail System and Webmail, we will touch a little deeper.

## **20.2 CUSTOMISING EMAIL NOTIFICATION**

Next, if you wish, you will need to customize your email notification message in */etc/asterisk/vm\_email.inc* file and to include the actual IP address of your asterisk PC. I have found that this is best done through a text editor (e.g. Nano) as AMP tends loose the formatting character. In the later versions of Elastix, this can be done through **Tools**  $\rightarrow$  File Editor.

; Change the email body, variables: VM\_NAME, VM\_DUR, VM\_MSGNUM, VM\_MAILBOX, VM\_CALLERID, VM\_DATE

emailbody=\${VM\_NAME},\n\nThere is a new voicemail in mailbox
\${VM\_MAILBOX}:\n\n\tFrom:\t\${VM\_CALLERID}\n\tLength:\t\${VM\_DUR}
seconds\n\tDate:\t\${VM\_DATE}\n\nDial \*98 to access your voicemail by
phone.\nVisit http://pbx.myhome.com:8080/cgibin/vmail.cgi?action=login&mailbox=\${VM\_MAILBOX} to check your voicemail
with a web browser.\n

#### You can be aggressive and change /etc/asterisk/vm\_general.inc as well.

; 1st listed format gets emailed	
format=wav49 wav	
attach=yes	
pbxskip=yes	; Don't put [PBX]: in the subject line
serveremail= <b>vm@asterisk</b>	; Who the e-mail notification should appear to come from
fromstring=Voicemail System	; Real name of email sender
maxmessage=180	; max length of vm message
minmessage=3	; Minimum length of a voicemail message in seconds
maxsilence=5	; Wait for 5 silent seconds and end the voicemail
silencethreshold=128	; What do we consider to be silence
skipms=3000 ;	; How many millisecs to skip forward/back when rew/ff in message ; playback
review=yes	; Allow sender to review/rerecord their message before saving it
operator=yes	; Allow caller to press 0
nextaftercmd=yes	; Automagically play next message when current message is
#### ; deleted or saved

Change the entry in red to reflect your requirement e.g you may change *vm@asterisk* to **FHC PBX** and change *Voice mail System* to read **Family Health Centre Voicemail**.

Once these are done, you will receive email notification of voicemail left in your voicemail box.

You may start testing it and your email received will look something like this:

From: Family Health Centre Voicemail To: Ben Sharif
Subject: New Message 1 in mailbox 201
Ben Sharif,
There is a new voicemail in mailbox 201:
 From: "Mary" <321>
 Length: 0:20 seconds
 Date: Friday, July 29, 2005 at 11:16:38 PM
Dial \*98 to access your voicemail by phone.
Visit http://pbx.familyhealth.com:8080/cgibin/vmail.cgi?action=login&mailbox=201
to check your voicemail with a web browser.

### **20.3 How to Retrieve Voicemail From External Phone**

To call in and retrieve your voicemail from external phone is as easy as simply calling your number and retrieve your voice mail. This can be done easily if you have IVR enabled in your new ELASTIX system.

This is how I do it (you may have been told of other different methods).

#### 20.3.1 Method 1

- 1. Call my number and when the Digital Receptionist answers, I will dial my extension number prefixing it with an \*. E.g. My extension number is **2001**, so when the DR answers me, I will dial **\*2001**
- 2. I will then be ask to leave a voicemail, but instead, I will dial \* again.
- 3. DR will ask for a password to be followed by the # key.
- 4. I enter my password followed by #
- 5. DR will then give me my voice mail if there is any.

#### 20.3.2 Method 2

- 1. From freePBX Setup Screen, create a **Misc Destination** and give it a name of e.g. "Check VM"
- 2. Select Feature Code Dial Voicemail (\*98) from the dropdown menu.
- 3. Submit the changes.

Next modify your IVR

- 1. Increase option. If you already have 7 options, this will be the 8<sup>th</sup>.
- 2. Create an unannounced IVR selection e.g. selection no: 8

- 3. For this option, select **Misc Destination** and from the drop down, select "Check VM"
- 4. Save and don't forget to click on the red bar.
- 5. When you call your PBX from an external phone, DR will naturally ask you to make a selection. Selection 8 will not be announced, however you should select 8.
- 6. DR will ask for a mailbox number
- 7. Enter your extension number (which is also your mailbox number)
- 8. Dr will then ask for your password
- 9. And you will get your mailbox IVR prompt.

As I said, it is indeed simple.

## **20.4 How to Change Voice Mail Greetings**

The default Voicemail greeting is fine but it is rather bland and quite boring. No doubt some user would like to change the standard greeting to something more personal and meaningful.



Ensure that you have changed your Direct Dial to Voicemail message type, in General Setting, from Default to something else e.g. Unavailable. Otherwise when you use follow-me and the like, your custom recorded message will not be played back. The Asterisk default message will be played back instead.

To do so is pretty simple. Here are the steps:

- 1. Use your existing extension and dial \*97
- 2. You will be asked for your password (if you have entered one)
- 3. When the Voicemail IVR starts, press 0
- 4. You will then be given the choice what type of message you want to record.
- 5. Choose the appropriate message you want to record.
- 6. Record your message "Thank you for calling, I am either busy elsewhere or on the phone. Please leave me your message and I will call you back as soon as I can".
- 7. Review your message
- 8. Re-record if you want to or if you are happy with the message, choose the option to save.

That's all – you are done.

# 21 SETTING UP VIRTUAL FAX

Before setting up Fax over IP, I suggest that we should read this: <u>http://www.soft-switch.org/foip.html</u> and : <u>http://www.voip-info.org/wiki-Asterisk+fax</u>

While sending and/or receiving Faxes are the available functions in ELASTIX, there have been reports of various degree of success if you are sending or receiving Faxes via IP. However it has been proven that receiving and sending faxes via Elastix through ISDN or PSTN have been consistently reliable and of commercial grade. This was achieved by utilising Hylafax which is a mature product (instead of the more commonly used mechanism by some other iPBX distributions).

One of the difficulties experience by many users are the inability to have the Fax (or voice mail notification) sent out by email. One of the possible reason is that postfix need a valid relay host address which you will need to add to *etc/postfix/main.cf* 

relayhost = optusnet.com.au (or whatever your service provider is)

Once this is done, changes are it may work (but then again it may not).

## 21.1 FAX FOR ELASTIX

Developers of Elastix have included this capability, which allows incoming faxes to be received over SIP, IAX & ZAP channel (though it is recommended that ISDN or PSTN are more reliable).

Faxing through IP are achievable however it is variable due to several factors which include hardware and bandwidth considerations and VSP support of T38. For commercial usage, it is not highly recommended.

Elastix does not utilise the standard Asterisk Fax handler for Faxing, instead utilising a separate module to handle fax traffic, which we will be discussing here.

As already stipulated, Elastix is using Hylafax. As such all tools and applications that interface to Hylafax can be used (you may even have 10 virtual fax machines). It is recommended that dedicated Fax lines be used for incoming and sending faxes although the same lines can be used for outgoing calls (not incoming calls) – *caveat: this assumption is made on the result of our testing.* 

#### **Configuring Elastix Fax Module**

	System PBX Fa	Email	1 Reports Extras		About Elastix 1.0-14	Logout
Virtual Fax List New Virtual	Fax   Fax Master   Fax Cli	nts   Fax Visor	Template Email			0 0
📄 Virtual Fax List						
	-				∥ Start 🖉 Previous (0	- 0 of 0) Next  > End  >>
Virtual Fax Name	Fax Extension	Secret	Destination Email	Caller ID Name	Caller ID Number	Status
					∥ Start / Previous (0	- 0 of 0) Next 🗅 End 🕅
		Elastix is li	censed under <u>GPL</u> by <u>PaloSanto</u>	Solutions. 2006, 2007.	″ Start ∕ Previous (O	-uoru) Next ▷ En

Click on the FAX tab and the following screen will appear.

Next is to Click on the New Virtual Fax selection - see arrow.

dual East List Now Mid	System PBX Fax Em	sall IM Reports Extras Agenda		
Edit Virtual Fax				
Apply changes Canor	al.			* Required fiel
Virtual Fax Name: *	Sydney Office	Fax Extension (IAX): *	1100	
	bensharif@gmail.com	Secret (IAX): *	1234	
Destination Email: *				
Destination Email: * Caller ID Name:	Netvoice Solutions	Country Code: *		

In this example we have populated the virtual fax field as the illustrated below.

- Virtual Fax Name: Sydney Office (or whatever you wish)
- Destination Email: I have used my email address but you should be using yours or a group email address.
- Caller ID Name: This should be your organization name.
- Caller ID Number: Your Fax number
- Fax Destination (IAX): Your extension which we will setup for receiving fax.
- Secret(IAX): This is the password of your extension.
- Country Code: In my case it is 61 for Australia, 60 for Malaysia etc.
- Area Code: 2 for NSW, 3 for Victoria etc.

Click on Save.

You will then notice the following screen with "Running and idle on ttyIAX1".

elast	System PB	X Fax E	imail IM Reports E	xtras	<u>About</u>	Elastix 1.0-14 Looc
Virtual Fax List New Vi	irtual Fax   Fax Master	Fax Clients	Fax Visor   Template Ema	1		0
📄 Virtual Fax Lis	st					
						≪ Start < Previous (1 - 1 of 1) Next ト End
Virtual Fax Name	Fax Extension	Secret	Destination Email	Caller ID Name	Caller ID Number	Status
Sydney Office	1100	1234	bensharif@gmail.com	Netvoice Solutions	61247364165	Running and idle on ttyIAX1
						″ Start / Previous (1 - 1 of 1) Next ト End
			Elastix is licensed under <u>GPL</u>	by <u>PaloSanto Solutions</u> , 20	06, 2007.	

That's a virtual FAX machine setup and ready to go as soon as we create an extension.

Create a Fax extension

By now we should already know how to set up an extension. This extension that we will be setting up is an IAX extension, which differs slightly from the SIP extensions that you are used to.

Select "Generic IAX device" at the "Create new extension" screen.

erator Panel	¥oicemails	Monitoring	Echo Canceller	Endpoint Config
Add a	a <mark>n Exte</mark> r	nsion		
Please s Device	elect your De	evice below t	nen click Submit	
Device Submit	Generic IA	X2 Device 📑	•	

You will be presented with the usual create new extension screen as illustrated below:

We have populated the various field:

- User Extension: 1100 (same user extension when you setup your virtual fax)
- Display Name: Sydney Fax (or whatever you want to call it)
- Secret: 1234 (make sure this is the same secret you use when you setup your virtual fax)
- Fax Extension: Disabled
- Incoming/outgoing recording: Disabled

Press submit when done.

Add Extension	
User Extension Display Name CID Num Alias SIP Alias Extension Options	1100 Sydney FAX
Direct DID DID Alert Info Music on Hold Outbound CID Ring Time Call Waiting Emergency CID Device Options	acc_1 V Default V Disable V
This device uses iax2 te secret	chnology.
Fax Handling	
Fax Extension Fax Email Fax Detection Type Pause after answer Privacy	disabled W
Privacy Manager Dictation Services	No V
Dictation Service Dictation Format Email Address Language	Disabled V Ogg Vorbis V
Language Code Recording Options	
Record Incoming Record Outgoing Voicemail & Directory	Never M
Status	Disabled 💌

We have now created a complete working virtual fax machine.

#### Directing Fax to the correct extension (Dedicated Line)

If you have dedicated line for the Virtual Fax, under freePBX, all you do is set the incoming route of your Fax DID to go directly to extension 1100.

#### Directing Fax to Fax extension on Non-dedicated Fax line (Auto Detect)

www.hellodigi.ir

This is where you wish to share one PSTN line for Voice and Fax. Your Elastix system will discriminate between Fax call and a voice call and send it to the appropriate extension for fax or pass it through to your normal handler if it is a voice call.

The following is what need to be done to achieve this:

In the freePBX General setting, ensure that the Extension of fax machine field is disabled (see below).

Fax Machine	
Extension of fax machine for receiving faxes: disabled	×
Email address to have faxes emailed to: bensharif@gmail.com	
Email address that faxes appear to come from: freepbx@gmail.com	

In the incoming route set your fax extension number to 1100 (the one we have set up) and set Fax Detect to Zaptel leaving your final destination to the voice call if it is not a fax call.

Description: DID Number: Caller ID Number: Fax Handling	Voice and FAX 61247364000	-	
Fax Extension: Fax Email: Fax Detection Type: Pause After Answer: Privacy	Sydney FAX <1100>		
Privacy Manager: Options	No 💌	-	
Alert Info: CID name prefix:: Music On Hold? Signal RINGING: CID Lookup Source	default V		
Source: Set Destination	None 💌		
Set Destination         Queues: General <600> V         Misc Destinations: Ben's Mobile V         Terminate Call: Hangup         Extensions: 1100> Sydney FAX         Voicemail: <2000> Reception (busy)         Conferences: General <8000> V         Zork (Read this link before downloading): ZoIP (Zork) V         Phonebook Directory: Phonebook Directory V         Announcements: Silent V         Custom Contexts: Full Internal Access V         DISA: VoIP2PSTN V         Ring Groups: Ring ALL <11> V         VR: Mainoffice V			

The last thing that you need to do is make a little change in Zapata.conf

1. Use nano to edit zapata .conf

- 2. Remove the comment on the line faxdetect=incoming (see illustration below)
- 3. If the line is not there then add faxdetect=incoming" to your zapata.conf file



Even though we have only set 1 virtual fax, you can set several if needed using the same procedure as outlined above except you need to use separate fax extension numbers e.g. 1100, 1101, 1102 etc

Reboot, and enjoy your new virtual fax.

## 21.2 FAX VISOR

To find out how many fax has been received and to see the content of the Fax can be done from Fax Visor.

- 1. Click on the Fax Visor Sub-Tab. The Fax visor screen will be displayed listing the Faxes that you have received.
- 2. Click on the blue link will open the fax.

See Illustration below.

	System PBX	Fax Email IM Reports	5 Extras		About Elastix 1.0-14 Logo	ut
Virtual Fax List No	ew Virtual Fax   Fax Master   Fa	ax Clients   Fax Visor   Templat	e Email		•	0
📄 Fax Visor		<b>N</b>				
Company Name:		Company Fax: 1		Fax Date:	Search	
					″ Start < Previous (1 - 4 of 4) Next ⊵ End	NY
Delete	File	Company Name	Company Fax	Fax Destiny	Fax Date	
	fax00000005.pdf	61247364165	61247364165	Sydney Fax - 1100	2008-04-06 16:57:20	
	fax000000004.pdf	61247364165	61247364165	Sydney Fax - 1100	2008-04-06 16:53:23	
	fax00000003.pdf	61247364165	61247364165	Sydney Fax - 1100	2008-04-06 16:06:20	
	fax00000002.pdf	61247364165	61247364165	Sydney Fax - 1100	2008-04-06 16:03:39	
	<b>^</b> 2					
		Elastix is licensed unde	er GPL by PaloSanto Solutions	. 2006, 2007.		

# **22 WEATHER REPORTS**

ELASTIX can be configured to allow the automatic downloading of Australian weather forecast information, convert it to a .wav file and play the announcement. The announcement will be a simulated announcement (ala Dr. Who's Dalek). However it will provide you with the weather report of your region of choice.

The Australia weather report reference file can be obtained from the following site: <u>http://www.bom.gov.au/</u>

You may choose the reference file (text file) for the area that you are interested in. You do not need to manually download the file, instead only the file name is required to be noted and can be incorporated within the Elastix's **extensions\_custom.conf** file later. The script will download the file for you in the background.

The following are examples of available weather report files that you can reference in your script.

Forecast Area	File reference
NSW Sydney Metro	IDN10064
All NSW	IDN10031
ACT	IDN10035
Snowy Mountain	IDN10103
Brisbane Metro	IDQ10090
All Queensland	IDQ10700
Melbourne Metro	IDV10450
All Victoria	IDV10310
Adelaide Metro	IDS10034
All SA	IDS10033
Perth Metro	IDW12300
All WA	IDW12000
Darwin and Rural Area (All NT)	IDD10110
Tasmanian Cities	IDT13400
All Tasmania	IDT13100

For a complete list, refer to the link below:

http://www.bom.gov.au/inside/cosb/stcm/idcodes/tables/current/vic\_products.shtml

The following step-by-step instruction will guide you through.

### 22.1 CONFIGURE WEATHER REPORT – ON DEMAND.

Since flite is not installed in Elastix, you will have to download flite and install it. Flite can be downloaded at <u>http://www.speech.cs.cmu.edu/flite/doc/index.html</u>.

After downloading copy the downloaded file into your root directory (*the file I downloaded* was *flite-1.3-release.tar.gz*) and install it thus:

```
tar zxvf flite-1.3-release.tar.gz
cd flite-XXX
./configure
make
```

Some preparations need to be manually conducted through Linux Command Line, or if you prefer, you may use Webmin.

I used Putty to log on to the Asterisk box remotely and used Nano to create a text file called "*convert2wav*" in the */usr/bin* directory containing the following:

/flite-1.3-release/bin/flite /tmp/weather.txt /tmp/weather.wav

(This command is used to convert the weather text file to a .wav sound file).

Once this is done, the permission for the "*convert2wav*" text file need to be appropriately set to allow it to be executed. To do this you need to run the following command from Command Line.

chmod 755 /usr/bin/convert2wav

Now you have completed all the necessary preparations – for me anyway.

#### 22.1.1 Create Extension script

The next step is to include the following in the *extensions\_custom.conf* through the *File Editor* option of ELASTIX Tools Tab. Select an unused extension code e.g. \*61, \*63, \*64 etc.

Add the following example (\*Note: The third line in each section (bold) should read as a complete single line- change the file reference in red to suit.).

```
; Forecast for Sydney Metro
exten => *61,1,Answer
exten => *61,2,Playback(national-weather-service)
exten => *61,3,Playback(pls-hold-while-try)
exten => *61,4,System(/usr/bin/wget -0 weather.txt
        ftp://ftp2.bom.gov.au/anon/gen/fwo/IDN10064.txt)
exten => *61,5,Wait(1)
exten => *61,6,System(/usr/bin/convert2wav)
exten => *61,7,Playback(/tmp/weather)
exten => *61,8,System(rm /tmp/weather.* -f)
exten => *61,9,Hangup
; Forecast for the whole of NSW
exten => *63,1,Answer
exten => *63,2,Playback(national-weather-service)
exten => *63,3,Playback(pls-hold-while-try)
exten => *63,4,System(/usr/bin/wget -0 weather.txt
        ftp://ftp2.bom.gov.au/anon/gen/fwo/IDN10031.txt)
exten => *63,5,Wait(1)
exten => *63,6,System(/usr/bin/convert2wav)
exten => *63,7,Playback(/tmp/weather)
exten => *63,8,System(rm /tmp/weather.* -f)
exten => *63,9,Hangup
```

Restart asterisk and test. You will be presented with the weather report of the areas you have chosen.

(Source Reference: http://forums.whirlpool.net.au/forum-replies.cfm?t=364420&p=1)

This method is very slow and therefore is only suitable for a very fast PC. Depending on the speed of your computer during the .wav conversion, there may be about 1 minute or more delay between dialling the extension and the weather announcement. Therefore I have included step 3 telling the caller to hold while trying to connect.

Therefore, there has to be a better method – so read on.

**Elastix Without Tears** 

## 22.2 CONFIGURE WEATHER REPORT – BACKGROUND METHOD.

After giving it a bit of thought, I have decided to let Asterisk do the task of getting the weather text files and converting them to the necessary sound files in the background every 4 hours. This will cut down the waiting period when a caller chooses to listen to the weather report.

First we will use putty to make a batch text file called "**convert2wav**" in the **/usr/bin** directory containing the following:

/usr/bin/wget -O /tmp/weather1.txt ftp://ftp2.bom.gov.au/anon/gen/fwo/**IDN10064.txt** /usr/bin/flite /tmp/weather1.txt /tmp/weather1.wav /usr/bin/wget -O /tmp/weather2.txt ftp://ftp2.bom.gov.au/anon/gen/fwo/**IDN10031.txt** /usr/bin/flite /tmp/weather2.txt /tmp/weather2.wav

(Change IDN10064.txt and IDN10031.txt to suit the areas you require)

If you want to get rid of a couple of nasty characters, you can always add the following line or lines in the appropriate places and changing the characters that you want to discard or change,

```
# sed 's/\[/ /g' /tmp/weather1.txt > temp.txt
# sed 's/\]/ /g' /tmp/weather1.txt > temp.txt
# sed 's/\_/ /g' /tmp/weather1.txt > temp.txt
as an example
```

- but we are not going there. It may get too confusing. For the moment we will just stick with the batch file example above.

Once this is done, change the permission for "*convert2wav*" thus:

chmod 755 /usr/bin/convert2wav

Use Webmin to schedule a Cron job to run this batch file every 4 hours (or schedule it to suit your purpose). This will run the batch job at the scheduled time avoiding having to download the weather file and converting them to sound files when a caller chooses the weather report option.

The sound files will always be available when a caller requires them without having to wait for them to be downloaded and converted.

#### 22.2.1 Create Extension codes.

Next include the following in *extensions\_custom.conf*. Select an unused extension code e.g. \*61, \*63, \*64 etc.

Add the following example

```
exten => *63,1,Answer
exten => *63,2,Playback(national-weather-service)
exten => *63,3,Playback(pls-hold-while-try)
exten => *63,4,Playback(/tmp/weather2)
exten => *63,5,Hangup
```

Restart asterisk and test. You will be presented with the weather report of the areas you have chosen without any delay whatsoever.

#### 22.3 Now put it in your Digital Receptionist

If you want the 2 procedures to be incorporated in your Digital receptionist, simply duplicate them in your **extensions\_custom.conf** with some minor changes. You will want the digital receptionist to know which weather report you are referring to when you make the choice. Give it a label – see example below (*the changes made are in bold*).

```
; Forecast for Sydney Metro
[custom-weathermetro]
exten => s,1,Answer
exten => s,2,Playback(national-weather-service)
exten => s,3,Playback(pls-hold-while-try)
exten => s,4,Playback(/tmp/weather1)
exten => s, 5, Hangup
; Forecast for the whole of NSW
[custom-weathernsw]
exten => s,1,Answer
exten => s,2,Playback(national-weather-service)
exten => s,3,Playback(pls-hold-while-try)
exten => s,4,Playback(/tmp/weather2)
exten => s,5,Hangup
```

In the older versions of freePBXs, when you create your menu option in Digital receptionist, e.g. "Press 6 for Sydney weather report, press 7 for the whole of NSW weather report", select the custom app: and enter the appropriate custom procedures above. See example below:

Custom App	custom-weathermetro,s,1	
Custom App	custom-weathernsw,s,1	

This option (Custom App) is no longer available instead freePBX has a different option under tools, called Custom Destinations. This is where you define the 2 custom applications above. Once defined, these custom applications will appear as a destination choice radio button called **Custom Applications** when you create your IVR choices. Using the dropdown selection, you can select the custom application that you have defined above.

#### 22.3.1 Custom Destinations

To create this custom destination (assuming that you already added the [custom-weathermetro] and [custom-weathernsw] in the extensions\_custom.conf file as illustrated above), you should select unembedded freepbx >> Admin >> Tools.

Select Add Custom Destination and add the custom application as illustrated below in the form of:

Custom Destination: custom-weathermetro,s,1

FreePBX	Admin Reports Panel Recordings Help
FreePBX 2.4.1.2 on 192.168.1.100	Logged in: admitr(Ltybut)
Setup Tools Admin FreePBX System Status	Add Custom Destination
Module Admin Support Asterisk Logfiles Online Support	Custom Destinations allows you to register your custom destinations that point to custom dialplans and will also 'publish' these destinations as available destinations to other modules. This is an advanced feature and should only be used by knowledgable users. If you are getting warnings or errors in the noticication panel about CUSTOM destinations that are correct, you should include them here. The 'Unknown Destinations' chooser will allow you to choose and insert any such destinations that are objective is not aylare of into the
FreePBX Support	Custom Destination field.
System Administration	
Asterisk API Asterisk CLI Asterisk Info	Add Custom Destination Custom Destination Custom Weathermetro.s.1
Backup & Restore	
Custom Destinations	Description: Sydney Metro weather
Custom Extensions	Wilds. Whatever notes you want to put here
Java SSH	
PHP Info	
PHPAGI Config	
Third Party Addon	
Customer DB	
Gabcast	Submit Changes - 3
Inventory	

You can create another one for custom-weathernsw using similar format.

Once these are done. Your new custom application will appear in your IVR selection under Custom Applications.

## **23 REMOTE MANAGEMENT**

Some may want to manage their Elastix Box remotely. If you are thinking about doing this, I highly recommend you register with Dyndns to get yourself a dynamic DNS unless you have a fixed public IP address.

You should also forward port 443 on your router to point to the local IP address of your Elastix box.

To log on to your Elastix box remotely using your browser, use https://your\_Dyndns\_name Of https://your\_public\_ip\_address

To log on locally you simply log on to https://192.168.1.101 (or whatever your local IP is).

# 24 HOW TO SET UP A REMOTE EXTENSION

If you carry your laptop/notebook everywhere you go, you may want to consider setting up a remote extension on your Notebook. Basically its set up is similar to the softphone that you set up on your other computers except that you should have a fixed IP address for your Asterisk server or use DynDNS to register a Domain ID for your Asterisk box.

Since I am on Optusnet cable, I only have the privilege of a dynamic IP hence I registered for a Domain ID with DynDNS pointing to my Dynamic IP address. Every time my dynamic IP changes, DynDNS will point to the new IP address. This way I don't have to keep changing IP addresses. I simply use the same DNS ID eg: *myasterisk.freedns.com* 

## **24.1 CREATE A NEW EXTENSION**

Create a new IAX2 extension in Elastix – just like the other extension that you created before. This time instead of creating a SIP extension, create an IAX extension (you can use SIP as well but since you don't have any control of the external router that you may be connected to when you are away from your own environment, IAX will avoid any NAT issue that you may encounter). *In your router, you will need to forward port 4569 to your Asterisk Box.* 

elastix	About Elastix 1.0-12 System PBX Fax Email IM Reports Extras	Logout
PBX Configuration Flash Op	erator Panel Voicemails Monitoring Echo Canceller Endpoint Configuration Conference Extensions Batch Tools	• •
Option Unembedded freePBX	Add an Extension	Add Extension
Extensions Feature Codes	Please select your Device below then click Submit Device	Ben Sharif <2000> Rossli Sharif <2003>
General Settings Outbound Routes Trunks	Device Generic 31P Device  Submit: Generic 31P Device Generic 31P Device Generic 40/2 Device Generic 40/2 Device	
Inbound Call Control Inbound Routes Announcements	Other (Custoin) Device	
Follow Me IVR		

As mentioned above, you may create a SIP extension instead, if you so wish but you must ensure that you set "**nat**" field to "**yes**" (in the device option) instead of "never" as illustrated in the screen below - see red arrow.

Faging and intercom		
Parking Lot	This device was a	in the share of a sure
System Recordings	This device uses s	ip technology.
Remote Access	secret	2003
Callback	dtmfmode	rfc2833
DISA	canreinvite	no
51011	context	from-internal
	host	dynamic
	type	friend
	nat	yes 🖌
	port	5060
	qualify	yes
	callgroup	
	pickupgroup	
	disallow	
	allow	
	dial	SIP/2003
	accountcode	
	mailbox	2003@default

Once you have done this, you will need to ensure a couple of things in your .conf files are set correctly.

## 24.2 IAX.CONF

The following is the content of my *iax.conf* file.

```
iax.conf
[general]
bindport = 4569 ; Port to bind to (IAX is 4569. Forward this port to your Asterisk box)
externip = <your fixed external IP> or
externhost = <myasterisk.freedns.com>
localnet = 192.168.1.0/255.255.255.0
bindaddr = 0.0.0.0
                          ; Address to bind to (all addresses on machine)
delayreject = yes
disallow=all
allow=g729
allow=ilbc
allow=qsm
allow=g723
allow=alaw
allow=ulaw
jitterbuffer = yes
mailboxdetail = yes
#include iax_additional.conf
#include iax custom.conf
```

## 24.3 SIP\_NAT.CONF

The only other .conf file that requires attention is the **sip\_nat.conf** in case you want to use SIP. If you do, ensure that you have forward ports 5004 - 5082/udp as well (strictly speaking you only require port 5060 as most if not all incoming SIP traffic will be coming in on this port). My file is as per the example below.

This file may have to be created as it may not exist.

```
sip_nat.conf
nat=yes
externip=<your fixed external IP> or
externhost=< myasterisk.freedns.com>
localnet=192.168.1.0/255.255.255.0
externrefresh=10
```

Those are the only 2 .conf files that require changing if needed.

### 24.4 SETTING UP AN IAX SOFTPHONE

I use an IAX softphone called IDEFISK. You can get IDEFISK from here: <u>http://www.asteriskguru.com/tools/idefisk\_beta.php</u>

You can also use Virbiage softphone as it will handle IAX.

Setting up IDEFISK is very simple. All you have to do is download the file, extract it and save it in its own directory.

Click on the idefisk.exe and just configure it like the example below.

Once the softphone is active, to start configuring, simply place your curser anywhere in the middle if the image and right click.

The configuration option window will be displayed.

**24.4.1 First the general options** In the general options, checked the "Check for update" and "Automatic popup on incoming call" options.

	SIDEFISK General options
	Record path
	Record calls
1	C Always on top
-	Check for updates
	V Automatic popup on incoming call
	OK Cancel

SIDEFISK V1.2	24 8
Phone to dial	LINE 1 LINE 2 LINE 3 LINE 4 LINE 5 LINE 5
Account Ben Sharif IAX	General options Account options Appearance options Audio options Filter options Codec options
1	About Exit

The next option to take care of is the Account option. Apart from that, there is nothing else critical to be done for you to use IDEFISK.

#### 24.4.2 Accounts options

Next click on the Account Options and set up your account as per the image below.

Accounts	Host				
Ben Sharif IAX	myasterisk.freedns.com				
	-User name				
	205				
	Password				
	isndhrtt				
	Caller ID				
	Ben on IAX				
	Number				
	Remote				
	Register on startup				
	New Update Set Default Delete				

The rest of the options are there for you to set up your sound preference and customize your look and feel. They are pretty straightforward.

Now you are ready to go on the road with an Asterisk extension under your arm.

# **25 CALL MONITOR & RECORDINGS**

There have been numerous discussions on the legality of recording telephone conversations. Some say its OK while others say its illegal. If you are recording conversation without the consent of all parties, you may be breaking the law – so be warned.

## **25.1 CALL MONITOR THROUGH ELASTIX**

As of version 1.0, monitoring does not work as expected. This is due to bugs that were not resolved. However after installing v1.0 from the ISO, updating it to 1.0-16 will resolve the problem somewhat.

To upgrade Elastix to the latest version, you simply log on to the console (or ssh using putty) and issue the following command at the root ptompt:

yum update elastix.

When completed, restart Elastix.

### **25.2 RECORD ALL CALLS BY DEFAULT**

To record all calls, on a specific extensions or all extensions, this function has to be enabled on the extension or extensions involved. This is done through the Extension Options Record Incoming and Record Outgoing field by selecting "Always" from the dropdown box for both fields.

Recording Options			
Record Incoming	Always	~	
Record Outgoing	Always	*	

## **25.3** RECORD CALLS ON DEMAND.

To record on demand, the caller or the receiver have to dial \*1 when he/she need the recording to start (Asterisk v1.2). This function will only work if the feature is enabled.

To enabled this feature, the following Dial Command Option must be enabled in freePBX **General settings** *Asterisk Dial command option* and *Asterisk Outbound Dial command option* fields by adding the following flag in the fields.

w: Allow the called user to start recording after pressing \*1 (Asterisk v1.2)W: Allow the calling user to start recording after pressing \*1 (Asterisk v1.2)

Some users have noticed that the \*1 need to be pressed very quickly for this feature to work. This can be remedied by adding the following line in the *features,conf* file under the general section.

featuredigittimeout = 2000 ;(the default is 500 which is too fast for an old guy like me

### **25.4 TO LISTEN TO THE RECORDED CALLS**

After logging on to Elastix, select PBX Tab then the Monitoring Tab and you will get the following screen where your calls recording will be displayed. You have the option of listening to the recordings or download the recordings.

Caracter and the second	System	PBX Fax	Email	Reports Extras		About Elas	tix 1.0-16	Load
X Configuratio	n Flash Operator Panel	Voicer ails	Monitoring Echo	Canceller Endpoint	Configuration Confere	nce Extensions Batch	Tools	0
Monitor	ing List	1						
	Start Data: 1 28 Arr 20			En	d Data: * 28 Arr 200		3 Eliter	
	Start Date: • 1 20 Apr 20						PROFE	
			1	2			- Start - Previous (1 - 2 of 2) Net	it > Eni
Delete	Date	Time	-		Duration	Туре	Message	
1	2008-04-28	11:40:00			23 sec.	auto - outgoing	Listen Download	
		11:48:49	2014	2001	13 sec.	always	Listen Download	
	2008-04-28							

## **25.5 ARI INTERFACE**

Another way of monitoring the calls is by installing the ARI interface. After installation, the ARI interface can be access from the unembedded freePBX GUI. However it must first be installed manually as it is not included in the Elastix distribution (download <u>here</u>).

If you want to listen to all the calls, you will need to log in as *admin* with the default password of *ari\_password*, otherwise you can only listen to your recordings only.

ARI	Asterisk Recording Interface
	Login
	Login: admin
	Password:
	Submit
	Remember Password
	English 💌
	Use your <b>Voicemail Mailbox and Password</b> This is the same password used for the phone
	For password maintenance or assistance, contact your Phone System Administrator.

To monitor the calls for any individual extension, log in using the **extension number** and the **extension password**.

To play the recording, simply click on the <u>**Play**</u> link related to the extension that you want to listen to, in the screen that follows.

ARI	Asterisk Recording Interface									
••••										
Voicemail Call Monitor	Call Monitor									
neip		Search								
<u>Settings</u> Logout	d	delete duration ignore Results 1 - 15 of 1300							00	
	sel	ect: <u>all</u> nor	<u>ne</u>							
		Date▼	<u>Time</u>	<u>Caller ID</u>	Source	Destination	Context	Duration	Monitor	
		2007-03-17	23:30:38	"Dr Hazanah" <3002>	3002	2001	default	304 sec	play	
		2007-03-17	23:18:47	"Ben Sharif" <2001>	2001	3002	from-internal	33 sec	<u>play</u>	
		2007-03-17	20:45:05	"The Sharifs" <899060194748>	899060194748	98337178	from-internal	35 sec	<u>play</u>	
		2007-03-17	20:41:33	"Kuala Lumpur" <60320565331>	60320565331	0410485476	from-internal	176 sec	<u>play</u>	

The configuration file for ARI is /var/www/html/recordings/includes/main.conf.php

For those who do record conversations through the ARI interface of unembedded freePBX, the recording are kept in the */var/spool/asterisk/monitor* directory.

## 26 STUN SERVERS

STUN (Simple Traversal of UDP through NAT - Network Address Translation) is a protocol for assisting devices behind a NAT firewall or router with their packet routing operates on TCP and UDP port 3478.

I never have the need to use STUN. The list below is provided for those who have the need to use public STUN Servers. These are the servers that I am aware off - there may be others.

stun.xten.com stun.fwd.org stun01.sipphone.com stun1.noc.ams-ix.net stun.voipbuster.com stun.voxgratia.org

# 27 DIALING THROUGH MS OUTLOOK®

One way of keeping a little black book for all your telephone contacts is through Microsoft<sup>™</sup> Outlook® (for the windows users) address book. We can also use Outlook to dial the phone numbers in the address book as Outlook® has a click-to-dial facility.

I was alerted of this facility in Outlook® and Asterisk® and asked to prove the application and publish a simplified instruction in this guide.

## 27.1 DOWNLOAD ASTTAPI

Before you get started, you will need to download <u>AstTapi</u>, which is a <u>Microsoft TAPI</u> to Asterisk Bridge. This little gem (still under further development at the moment but functional) will make it possible to do click-to-dial from Microsoft<sup>™</sup> Outlook and for that matter from other TAPI compliant applications.

Download AstTapi from <a href="http://sourceforge.net/projects/asttapi/">http://sourceforge.net/projects/asttapi/</a>

To get started, you will need to edit *manager\_custom.conf* file. The easiest is to use Elastix's File Editor (the easiest way!)

#### Tools -> File Editor -> manager\_custom.conf

On opening the *manager\_custom.conf* file, you will notice the following, which you will need to do a tiny weeny edit:

```
[phpconfig]
secret = php[onfig
deny=0.0.0.0/0.0.0.0
permit=127.0.0.1/255.255.255.0
read = system,call,log,verbose,command,agent,user
write = system,call,log,verbose,command,agent,user
[phpagi]
secret = phpagi
deny=0.0.0.0/0.0.0.0
permit=127.0.0.1/255.255.255.0
read = system,call,log,verbose,command,agent,user
write = system,call,log,verbose,command,agent,user
```

```
[AstTapi]
```

```
secret = AstTapi
deny=0.0.0.0/0.0.0.0
#permit=192.168.1.0/255.255.255.0
read = system,call,log,verbose,command,agent,user
write = system,call,log,verbose,command,agent,user
```

Under the [AstTapi] User context, remove the # from the permit line and change the 192.168.1.0 to the network address of your network. In my case I left it as it is because my network is under 192.168.1.0 (see red highlight).

Once this is done, click Update and restart Asterisk.

## 27.2 INSTALL ASTTAPI

Install AstTapi on your PC – the one that you have Microsoft Outlook installed. Follow the prompt and once completed you will need to restart your PC for the new driver to take effect.

## 27.3 CONFIGURE OUTLOOK ADDRESS BOOK

Fire up Outlook®.

Click on Contacts from your folder list.

Select a contact that you want to call (provided that a telephone number has been recorded).

Click on the phone icon.



Click on the Telephone icon.

The New Call dialog box will be displayed:

Select Dialling Options.



The Dialling Option box will be displayed.

In the "connect using line" drop down box, select Asterisk.

Then click on Line properties.

		X - 1 - 2
		Mog
Name	Number -	Delete
Ben Sharif	+61 (2) #36	
- Maria Caraltana ang	have for example to so and whether a	
ettings for phone num	ber formatting and dialing	
Automatically add o	ountry code to local phone numbers	
Dialing Properties		
oppert using line		
LOT IF IOUC USING IF IO		
connect dang inte	Line Prov	a shi a a
Asterisk	▼ Line Prop	erties
Asterisk	Line Prop	erties
Asterisk	CK	erties Cancel
Asterisk	CK	erties Cancel
Asterisk	CK	erties Cancel

In the **Host** field, enter the address of your asterisk server: **e.g. 192.168.1.100** 

In the Port Field: 5038 User Field: AstTapi Password: AstTapi

Asterisk Server		Context
Host	192.168.1.100	Dial by 'Context'
Port	5038	Context outbound-allroutes
Jser Information	1	Caller ID
User	AstTapi	Attempt to set outgoing ID
Password	****	Dial
User Channel	sip/2001	C Dial out by using the 'Dial' application
Inbound Chan		Outgoing Chan
	Lise regex for pattern matching	

User Channel: (your extension – the extension you want Asterisk to ring before connecting you the number you are calling) e.g. sip/2001

Select dial by context radio button.

Enter outbound-allroutes in the context box

When done, click Apply and you are done.

You can now start making calls from Outlook using the click-to-dial facility.

Now... that was quite painless.

## 28 SPEED DIAL

Sometime you may just need to call someone quickly without having to look up his/her phone number. Well, you can by simply defining a shortcut number. There are 2 ways you can do this - my way and Elastix's way.

## **28.1** THE DUMB-ME METHOD

Since I am very bad at remembering phone numbers – I cant remember my own phone number (or my car registration number), I have defined some shortcut numbers for my family using the first 3 characters of their first names e.g. Adam is ADA which translate to 232 on the telephone keypad. Make sure the numbers have not been allocated to your local extension.

To do this, you will need to edit *extensions\_custom.conf* by adding the lines below containing the relevant exten expressions.

You can do this through Tools -> File Editor

```
; Speed Dial to some predefined numbers (added by Ben 2/12/2005)
exten => 236,1,playback(pls-wait-connect-call)
exten => 236,n,Dial(SIP/6140036xxxx@pennytelsip,60,m) ; Ben's Mobile
exten => 236,n,Hangup
exten => 764,1,playback(pls-wait-connect-call)
exten => 764,n,Dial(SIP/6141242xxxx@pennytelsip,60,m) ; Rohani's Mobile
exten => 764,n,Hangup
exten => 667,1,playback(pls-wait-connect-call)
exten => 667,n,Dial(SIP/6140470xxxx@pennytelsip,60,m) ; Nor's Mobile
exten => 667,n,Hangup
exten => 767,1,playback(pls-wait-connect-call)
exten => 767,n,Dial(SIP/6140512xxxx@pennytelsip,60,m) ; Ros's Mobile
exten => 767,n,Hangup
exten => 232,1,playback(pls-wait-connect-call)
exten => 232,n,Dial(SIP/6140936xxxx@pennytelsip,60,m) ; Adam's Mobile
exten => 232, n, Hangup
```

You will need to substitute the example phone numbers above such as 6140036xxxx with the real phone numbers.

When 236 (which is BEN) is dialled, you will hear a voice prompt saying "Please wait while I connect your call" followed by Asterisk calling Ben's Mobile number which is 61400336510 using the Pennytel SIP trunk.

Pennytel SIP trunk was called Pennytelsip when it was initially set up.

The call will be tried for 60 seconds (,60) and while the call is being attempted, the caller will be listening to a MOH tune (,m) instead of the ringing tone. You may however change it to a ringing tone by substituting the "m" with a "r" or remove ",m" altogether

Once you have done that, click on Update and re-read the config. That's all there is to it.

## **28.2 UNEMBEDDED FREEPBX'S METHOD**

Elastix v1.x, does not provide this facility. To set up the Speed Dial, you need to use the unembedded freePBX under the TOOLS Tab.

FreePBX	Admin Reports Panel Help			and and a
FreePBX 2.4.0.0 on 192.168.1.37	K		Logged in:	admin (Liggelui)
Setup Tools				English 💌
Admin	Add or replace entry			
FreePBX System Status				
Module Admin				
Support	Number 2	0400123456		
Astensk Lognies	Name:	Ben Sharif		
Control Support	Concert distants			
System Administration	Speed dial code:	89		
Asterisk API				
Asterisk CLL				
Asterisk Info	Submit Changes			
Asterisk Phonebook	<b>←</b> 3			
Backup & Restore				
Custom Destinations	Import from CSV			
Custom Extensions				
Java SSH				
PHP Info	File: Browse			
PHPAGI Config				
Third Party Addon				
Customer DB	Upload			
Gabcast	o produce			
Inventory				
Print Extensions				
	FreePBX* Freedom to Connect® FreePBX 240 is licensed under OPL			

Here you simply enter the Phone number of the party you wish to call, the name and the Speed Dial Code in the appropriate fields as per the example below. Submit change when done.

To call the Speed dial, e.g. you want to call Ben Sharif, simply dial \*069 (the \*0 tells Elastix that you want to use the Speed Dial and the 69 is the Speed Dial Code for Ben Sharif).

The Speed Dial Prefix \*0 can be found under Feature Codes.

Naturally if you have a big list of Names and Phone Numbers that you want to enter in your Elastix Phone Book, you have the option of uploading the list. You have to scroll down the page to get to the "Import from CSV" section of the Phone Book (see illustration below)

Custom Destinatio	ns	Import from CSV		
Custom Extension	s			
Java SSH				
PHP Info		File	Browse	
PHPAGI Config	Import a CSV P	file formatted as follows:		
Third Party Addon	Names should	be enclosed by "" and fields separated		
Customer DB	by '/			
Gabcast	Example:			
Inventory "John Doe";12		345678;123		
Print Extensions				
		FreePBX	reedom to Connect <sup>®</sup> reePBX is a registered trademark of Atengo, LLC. reePBX 2.4.0 is licensed under OPL	

The "Import from CSV" is a misnomer because, if you havae created a list using EXCEL and trying to import the list, it will not work since EXCEL CSV file is a comma delimited one where as freePBX is expecting a semi-colon delimited file. Mousing over the word file will give you the format it is expecting;

"Name";Number;Speeddial code

Ensure that "Name" must be enclosed in quotation marks.

Any speeddial code that already exist in the phone book cannot be reused. If you want to use an existing speeddial code in your so called "CSV" file, the existing entry in the phone book must first be removed.

## **29 HOW TO USE WINDOW MESSENGER® 5.X**

Windows Messenger is **NOT** MSN Messenger. You may need to download Windows Messenger from Microsoft.

Set up Windows Messenger 5.x as a SIP client on your PC:

1. Start Windows Messenger from your PC.

From the Windows Messenger dialog box, go to Tools -> Options -> Accounts:

- Check the [My contacts include users of a SIP Communications Service] checkbox.
- Enter "sip:<your extension>@<Asterisk Address>" (e.g.."sip:206@192.168.1.101") in the [Sign-in name] field

Options	×
Personal Phone Preferences Privacy Accounts Connection	
.NET Passport Account — Signed In ™ My contacts include users of .NET Messenger Service	
SIP Communications Service Account Signed In -	
Exchange Account Not Signed In – Control In Not Signed In – Not Signed In – Sign-in name: Advanced	

2. Click on the [Advanced] button next to the Sign-in name.

SIP Communications Service Connection Configuration 🛛 🛛 🔀		
Select which method should be used to configure your connection to a communications service:		
O Automatic configuration		
Onfigure settings		
Server name or IP address: 192.168.1.101		
Connect using:		
O TCP		
O TLS		
⊙ UDP		
OK Cancel Help		

- Check the [Configure settings] button
- Enter Asterisk IP address (e.g.192.168.1.101) in the [Server name of IP address] field
- Check the [UDP] button
- Click [OK] and [OK] again.

In order for Windows Messenger to work as a voice communication device, you must have a microphone and speaker(s) connected to your PC.



This setting means the Windows Messenger has a user/phone number of 206 known to the SIP server (your Asterisk box) of 192.168.1.101.

Verify that Windows Messenger has successfully registered with Asterisk by logging in to Asterisk.

#### UnembeddedfreePBX -> Tools -> Asterisk Info.

You should see under SIP Registry that the Windows Messenger is registered as a user/phone number of 206, with a contact URL of your local PC IP Address e.g. 192.168.0.11, among other registration information.

Note: Windows Messenger 5.0 or later has support for VoIP communications but it seems that it may create problems with chat or other features. For this purpose, please focus your testing on VoIP only.

# **30 HOW TO INTERCONNECT 2 BOXES**

There may be a time when you want to interconnect 2 Asterisks boxes (*def.com.au* and *xyz.com.au*) together and if you are like me, you will probably be spending a good part of 3 hours trying to get them to talk to one another.

I have 2 different locations, the Main Office (*def.com.au*) with about 11 extensions and another office in a different location (*xyz.com.au*) about 20 km away with 9 extensions. The main office is the only box that will have accounts with different VSPs and all external communications are through the main office Elastix box. I settled for the simplest solution and after some fiddling around I managed to get them to work the way I wanted it but not happy with it, I solicited some advise from a friend (thanks to Mark Brooker) who provided further assistance.

Instead of being verbose in my explanation, I will just create a few tables outlining what I did. I hope this will help those in the same position as I am, to set 2 very basic systems together (you can refer to <u>DUNDi</u> for a more complete solution). A little tutorial on DUNDi can be found <u>here</u>.

## **30.1 METHOD 1 - WITH THE PEER ASTERISK BOX AS EXTENSIONS**

For the purpose of registering the peers to each other, I created 1 extension on each box eg: 90000 on System 1 and 91000 on System 2– using extension numbers that I am not likely to use as local extensions (while some users have had success using common extension, but I prefer 2 separate extensions as I have them working). For simplicity, I gave a common password *xxxyyy* to both boxes. Avoid using extension starting with 8 as it may clash with conferencing.

·	System 1 def.com.au	System 2 <i>xyz.com.au</i>
IAX Trunk		
Outgoing Dial Rules:	XX.	XX.
Trunk Name	Parramatta	MainOffice
Peer Details	host=xyz.com.au (or System 2 IP) qualify=yes secret=xxxyyy type=peer username=91000	host=def.com.au (or System 1 IP) qualify=yes secret=xxxyyy type=peer username=90000
User Context	Leave blank	Leave blank
User Details	Leave blank	Leave blank
Register String	91000:xxxyyy@xyz.com.au	90000:xxxyyy@def.com.au

Note: Registration isn't really necessary. It will still work without it unless you use Dynamic IP.

	System 1	System 2
Extensions		
Phone Protocol	IAX	IAX
Extension Number	90000	91000
Extension Password	хххууу	хххууу
Fullname	Parramatta	Main Office
Voicemail & Directory	Disabled	Disabled

	System 1	System 2
Outbound Routing		
Route Name	Parramatta	MainOffice
Route Password	Leave Blank	Leave Blank
Dial Patterns	6XXX (6001 to 6009 are Parramatta Office extensions)	XX. (Apart from Local extensions, all others go via Main Office)
Trunk Sequence	IAX2/Parramatta	IAX2/MainOffice

The above Outbound Routing rule assumes that you do not wish to use a dialling prefix. If you want to use a prefix to dial the remote extensions and to use the remote routing rules, you may place a prefix e.g. 9|6XXX and 9|XX. for system 1 and system 2 respectively instead of just 6XXX and XX.



While this method will provide some rudimentary security (though pretty weak), as it requires an extension to be created for the peer Asterisk box, it will not pass the calling party extension number to the remote Asterisk box. Instead, it will pass the Trunk ID only and all calls will seem to come from the same trunk and not individual extension – I did say that this is a simple solution.

## 30.2 METHOD 2 - IN A PEER/USER ARRANGEMENT

Another method that I use is described below. This method treats both the Asterisk box as internal to each other as peer and user. I am using IAX2 for this purpose, however I believe, you may be able to do this with SIP as well if you are trying to connect the older Asterisk with the newer incarnations (I have not proved it yet). This method does not require registration either and does not require you to create extensions for the peers. In many ways, this is simpler to set up.

Unlike the first method, this second method will pass the Caller ID to the receiving party. The receiving party will actually get the callers' extension number/ID instead of the extension number of the peer Asterisk box.

Rather than being verbose, I will illustrate this method using tables as follows:

	System 1 def.com.au	System 2 <i>xyz.com.au</i>
IAX2 Trunk		
Outgoing Dial Rules:	XX.	XX.
Trunk Name	InterOffice	InterOffice
Peer Details	host=xyz.com.au (or System 2 IP) Qualify=yes	host=def.com.au (or System 1 IP) Qualify=yes
	type=peer	type=peer
User Context	InterOffice-In	InterOffice-In
User Details	context=from-internal host=xyz.com.au (or IP)	context=from-internal host=def.com.au (or IP)
	type=user	type=user

	System 1	System 2
<b>Outbound Routing</b>		
Route Name	InterOffice	InterOffice
Route Password	Leave Blank	Leave Blank
Dial Patterns	6XXX (6001 to 6009 are Parramatta Office extensions) 9 XX. (To call external number via Main Office)	XX. (Apart from Local extensions, all others go via Main Office)
Trunk Sequence	IAX2/InterOffice	IAX2/InterOffice

## **30.3 METHOD 3 – FRIEND/FRIEND ARRANGEMENT**

(By far the simplest method)

This method is by far the simplest method where the boxes are treated as friend to each other. Like Method 2, registration is not required and this method will also pass the Caller ID to the receiving party.

	System 1 def.com.au	System 2 xyz.com.au
IAX2 Trunk		
Outgoing Dial Rules:	XX.	XX.
Trunk Name	InterOffice	InterOffice
Peer Details	host=xyz.com.au (or System 2 IP) Qualify=yes type=friend	host=def.com.au (or System 1 IP) Qualify=yes type=friend
User Context		
User Details		

	System 1	System 2
Outbound Routing		
Route Name	InterOffice	InterOffice
Route Password	Leave Blank	Leave Blank
Dial Patterns	6XXX (6001 to 6009 are Parramatta Office extensions) 9 XX. (To call external number via Main Office)	XX. (Apart from Local extensions, all others go via Main Office)
Trunk Sequence	IAX2/InterOffice	IAX2/InterOffice

All the above examples assume that both Asterisk boxes have Public Fix IP address. If you have Dynamic IP addresses, you will need to register both the boxes with DynDns to obtain a valid DNS ID. If you are a part of a Corporate LAN, then you will have no need to worry about DynDns and what not.

#### Warning: You must provide for security, as this is pretty wide open.

Like all installation, you must provide for security. As different installation resorts to different types of security arrangement, I will leave that to the individual implementer to deal with the security issues.

## **30.4** THINKING OF MORE THAN 2 BOXES?

Just as a matter of interest, you can connect several boxes using this method. While I have connected 3 boxes successfully, I believe, the same principle can be applied to more boxes.

In my implementation I have box A, B and C (System 1, 2 and 3) in a star configuration. Box A is the master box. All the other boxes use box A as the main exchange.



A peers with B and C - B peers with A - And C peers with A.

*Except for local traffic, all external and inter-office (inter-branch) traffic goes via Box A. – with the appropriate dial plan of course.* 

Both the above methods, while useable for a basic configuration, will not provide you with a complete solution. To provide a complete solution is beyond the scope of this document.

The following link will provide further reference for connecting two Asterisk boxes together <u>http://www.voip-info.org/wiki/view/Asterisk+-+dual+servers</u>

If you require a complete solution tailored to your exact requirement, my advise to you is to hire a VOIP consultant.

## **30.5** REDIRECT INCOMING IN BOX A TO BOX B

There will be times when you will want to redirect certain calls made to one Elastix box to another Elastix box in another location. This can happen in a scenario where you are moving your mail office from your existing location to another location (or moving your sales office to another location away from your main office).

Personally I really wouldn't do this as it is simpler to get a redirection to the new number from the exchange.

In this example, lets say that all calls coming to your head office Elastix PBX made on phone number **12345678** need to be sent to your Elastix PBX in a different location.

Let us call your Head Office PBX Box-A and your remote PBX Box-B and calls coming in to Box-A to be sent to Box-B.

There are a number of ways that this can be achieved but most involved editing conf files and adding extra codes in the conf files. Unless you are quite comfortable with doing this, it can be quite daunting to the uninitiated.

The example I am illustrating here does not involve editing any conf file. All can be done through freePBX. It is a little long winded but you don't have to touch any of the conf files.

#### Step 1 – (Do this in Box-A)

#### i. Create an IAX Trunk

You will only need the following in the *Outgoing Setting* and leave everything else blank. Let us call this trunk *TransferTrunk*.

host=BoxB.com.au (your qualified DNS or the IP address) type=peer

username=6000 (this number will be used later)

#### ii. Create and Outbound Route

After you have created the trunk, you will need to create an outbound route. Let us call this outbound route *TransferRoute*.

In the dial pattern, you will only require one entry:

6000

The trunk sequence will be *TransferTrunk* only.

#### iii. Create a new extension

Next create a new extension e.g. 500. You will be using this extension to unconditionally forward all calls made to this extension, to extension 6000 in BoxB which we will create later.

To unconditionally forward calls made to this extension to extension 6000, you need to connect to extension 500 using a softphone and dial **\*726000**. Any calls to this extension will be unconditionally forwarded to extension 9000.

#### iv. Create and inbound route

DID number is 12345678. This is your phone number of calls come in to Box-A from, assuming that all calls that come on this number are to be transferred to Box-B.

Set Destination to **500**.

That's all need to be done in Box-A

#### Step 2 – (Do this in Box B)

#### i. Create a Trunk

Trunk name: TransferTrunk. Leave the peer details blank.

#### User context: Transfer-In

You will only need the following in the *Incoming Setting* and leave everything else blank.

context=from-trunk

#### ii. Create a new IAX extension

Create a new IAX extension *6000* (this is the username defined in the outgoing setting in Step 1 above). Display name: *CallTransfers*.

Edit this extension context to read *from-trunk* (instead of from-internal).

#### iii. Direct all calls to IVR

Ensure that incoming route direct calls to the appropriate extension. In my case, all calls go to IVR.

That is all we need to do and from now on, calls that come in to Box-A on your phone number 12345678 will be directed to extension 500 of Box-A. However, since all calls to extension 500 has been unconditionally redirected to extension 6000, asterisk will forward the call to extension 6000 which your outbound route will pick up and send to Box-B.

When it gets to Box-B, it will be answered by the IVR in Box-B.

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# 31 CUSTOMISED VOICE

The default pre-recorded voice prompts and announcements in ELASTIX are suitable for almost every situation, however there will be times when customised voice prompts or announcements will be required, e.g. to mix English and other language on a single announcement.

Although there are separate voice sets that can be used, sometimes we only require a few prompts and not the entire voice set.

## **31.1 CUSTOMISING VOICE PROMPTS**

To satisfy this requirement, custom voice prompts can be recorded individually by using the **System Recording** facility of freePBX as covered in the chapter referring to <u>System</u> <u>Recording</u> earlier in this document.

However, if you want to manually cut the script, you can do the following (personally I wouldn't bother – why re-invent the wheel?). Under the *[from-internal-custom]* context of *extensions\_custom.conf*, add the following codes.

```
[from-internal-custom]
; For custom recording
exten => 5678,1,Wait(2)
exten => 5678,2,Record(/tmp/my-recording:gsm)
exten => 5678,3,Hangup
; for playback of custom recording
exten => 5679,1,Playback(/tmp/my-recording)
exten => 5679,2,Hangup
```

To start recording, use one of the phone extension and dial 5678. At the beep, start recording the voice prompt.

The voice prompt will be saved as *my-recording.gsm* (gsm format) in the /*tmp* directory.

When completed, hang up and dial 5679. The voice prompt will be played back.

If you are not satisfied, repeat the above process.

Once you are satisfied, rename the file to something related and recognisable e.g. **my\_office\_business\_hours.gsm**. The file should then be moved to the **/var/lib/asterisk/sound** directory.

The file can be played through your custom applications or prompts using the *Playback* or *Background* function of Asterisk.

A good information on Asterisk sound files and how to create them can be found here <u>http://voip-info.org/wiki/view/Asterisk+sound+files</u>

## **31.2 INSTALLING OTHER LANGUAGES**

If you wish to use a different language other that the default American English that comes with Asterisk, there are a couple of ways that you can do but as this guide is for beginners, I will go the easiest method.

Create a subdirectory for the particular language (eg: 'au' for Australian English, 'fr' for French, 'it' for Italian etc.) in the **/var/lib/asterisk/sounds** directory, **e.g. /var/lib/asterisk/sounds/au** and send all your prompts sound files there. Also create the language directories in the digits, letters etc directories, and send the entire prompt files there.

Add the following line in the iax.conf and sip.conf configuration file under [general]. In zapata.conf configuration file the line should be added under [channels].

language=au ; (or fr, it or whichever language is installed)



You may use the Edit Configuration facility of ELASTIX, otherwise you can use nano at the command prompt of your Asterisk box (or SSH to your Asterisk box) to edit the files that resides in the etc/asterisk directory.

## **31.3 CONVERTING WAV FILES**

For those who wish to record their own sound prompts, the following may be of some assistance.

Elastix will be able to play virtually any sound that it has the codec for e.g. wav(pcm), wav49, gsm, g711, g729 etc. However, the gsm format seems to be the common format used for the default voice prompts.

#### 31.3.1 Converting WAV to GSM

Since the WAV format is the most common format that is being adopted when recording with a windows based PC, and most times, the windows based WAV format does not play well with asterisk, below is a method of converting the windows recorded WAV format to GSM using SOX.

Your result will be better if you record your sound files in mono, 16 bit, 8000 Hz.

After recording the WAV sound files, transfer the sound files to the TEMP directory of the Elastix PC. In this example, let,s call one of the sound files *hello.wav*.

- □ Login as root and change directory to the directory where you have transferred the sound files to e.g. cd /tmp
- □ At the prompt issue the following commands.
  - If your sound files were recorded in mono, 16 bit, 8000 Hz

sox hello.wav hello.gsm

• If your sound files were recorded in mono but NOT 16 bit, 8000 Hz

sox hello.wav -r 8000 hello.gsm resample -ql

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• If your sound files were recorded in stereo, you will need the -cl switch.

sox hello.wav -r 8000 -cl hello.gsm resample -ql

 if your sound files were recorded in ADPCM wav files, to convert to standard wav file;

sox hello.wav -r 8000 -c1 -s -w hello-out.wav resample -ql

Next, move the sound file to the sound directory where all your prompts are stored and you are done.

You may also convert a number of WAV files at once using the following command. In this example, lets assume that the files were all recorded in stereo;

```
for a in *.wav; do sox "$a" -r 8000 -cl "`echo $a|sed -
e s/wav//`gsm" resample -ql
```

#### 31.3.2 Converting WAV to SLN

Starting from Asterisk version 1.2.0, the .sln (SLINEAR) format seems to be the format that is being adopted (the GSM format is still commonly being used), so don't worry about converting your existing prompts.

To convert wav file to sln, use the following command:

sox hello.wav -t raw -r 8000 -s -w -c1 hello.sln

Further reference for converting way sound files can be found here at voip-info.org.

# 32 MOH (MUSIC ON HOLD)

Elastix comes with a number of preinstalled music on hold selections, however you may want to add your own music to the existing selection or remove the default selection altogether, and only use your own selection instead.

## 32.1 CUSTOMISING MOH

ELASTIX supports native MP3. To change or add to your music on hold collections, simply upload all your MP3 to ELASTIX through the **Onhold Music** screen of freePBX.

You may also need to add the following, under [channels] in your zapata.conf file:

musiconhold=default

Pretty much any mp3 will work with Elastix. Prior to that, it's better to convert it to a standard format. When converting my MP3 music, I use <u>FreeRip</u> and convert the music as per the following:

Bit Rate: 128 Mono Constant Bit Rate (CBR)

Make sure that there is no ID3 Tagging.

I found that this conversion gives me the most satisfactory result.

To customise your music to your favourite, you may need to create a new music category directory to hold your favourite music. Do this through the freePBX "On Hold Music" GUI interface. I call my category *favourites*.

Upload your favourite music to this directory. Again, do this through the freePBX GUI. Once finished uploading, click the "Enable Random Play" button and the red bar.

Make a little change to your musiconhold.conf file like the example below (this way you don't have to delete the music in your default directory).

You will now have to get used to the newer approach (it no longer uses the [classes] tag):

The *musiconhold.conf* files in 1.2 is now set out differently as per the example of my file below:

#### musiconhold.conf

```
wusic on hold class definitions
Wusic on hold class definitions
This is using the new 1.2 config file format, and will not work with
1.0
based Asterisk systems
[default]
mode=files
;
valid mode options:
; quietmp3 -- default
mp3 -- loud
mp3nb -- unbuffered
; quietmp3nb -- quiet unbuffered
; custom -- run a custom application
; files -- read files from a directory in any Asterisk supported format
;
directory=/var/lib/asterisk/mohmp3/favourite ;<-I pointed this to my favourite</pre>
```

random=yes #include musiconhold\_additional.conf

The corresponding entries can be found in musiconhold\_additional.conf – see below:

#### musiconhold\_additional.conf

```
[acc_1]
mode=files
directory=/var/lib/asterisk/mohmp3/acc_1/
random=yes
```

## **32.2 STREAMING MUSIC ON HOLD**

Streaming music on hold is not directly supported using the GUI of ELASTIX or freePBX, however if you are prepared to do a little editing of the musiconhold.conf file and downloading some external files, you will be able to get it working somehow.

There are 2 ways of doing this and I will outline them both below:

#### Method 1

(old method of doing things but pretty straight forward and it works)

Below is what I have to do to get Streaming On Hold for my system. I realised that it is not the best method, but for the sole purpose of getting Music on Hold, it got me working.

From command line, find out if you have mpg123 installed using the following command.

mpg123 -v

You will find out if you have mpg123 installed or not and its version number. The version that we want is *mpg123-0.59r*.

Since I do not have mpg123 installed, first I have to download and install mpg123 - it is not included in Elastix implementation of Asterisk. To get mpg123 I have to log on as root and at the command prompt I did the following:

```
cd /usr/bin
wget http://www.mpg123.de/mpg123/mpg123-0.59r.tar.gz
tar -zxvf mpg123-0.59r.tar.gz
cd mpg123-0.59r
make linux
make install
ln -s /usr/local/bin/mpg123 /usr/bin/mpg123
```

I have to make a directory for the streaming music. You can do this though the freePBX GUI or manually. In this example, I have shown the manual method.

```
su asterisk
cd /var/lib/asterisk/mohmp3/
mkdir stream
```

Next I need to create a zero length dummy MP3 file for this purpose in the /var/lib/asterisk/mohmp3/stream directory (this can be done manually).

cd stream

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touch stream.mp3 exit

Now I have to do some editing of my *musiconhold.conf* file.

Open the musiconhold.conf file using the Config edit option on your Elastix and add the following lines on top of the file;

[classes]

default => quietmp3:/var/lib/asterisk/mohmp3/stream,http://128.177.3.80:4064/

and leave everything else out (or commented out) like my example below.

The *http://128.177.3.80:4064/* (SKY.FM – Country) is where I get my streaming music from.

This is what my *musiconhold.conf* looks like.

[classes]

default => quietmp3:/var/lib/asterisk/mohmp3/stream,http://128.177.3.80:4064/
;

The lines in BOLD are the lines that was added to the musiconhold.conf file.

After having done that you may reboot your system.

The above is not necessarily being the best way to approach it but that is the simplest way that I know of.

#### Method 2

(as suggested by Bob Fryer)

Bob has since updated his method of getting Streaming Music On Hold to work.

For complete instruction, please refer to:

http://www.elastixconnection.com/index.php?option=com\_content&task=view&id=62&Itemid=83

The previous method that was published in this document was tested on system running Elastix 0.9.2, but we understand that there was some problem getting it to work consistently hence the update on elastixconnection.com.

After installing as described in the tutorial, you must reboot your system afterwards.

Then go into your trunks and extensions in freepbx and choose the MOH Stream you want for on hold music.

The following are addresses where you may find some streaming music for your MOH.

181.kickin' country http://64.236.34.106:80/stream/1075/ Classical 96.3 FM http://216.126.84.50:8000/ SMOOTHJAZZ.COM http://scfire-chi0l-2.stream.aol.com:80/stream/1005/ Sky FM – Modern Jazz http://205.188.215.227:8008/ Energy 98 – Dance Hit http://193.138.205.194:8014/ Life FM – Adelaide Australia http://203.16.214.174:8040/ Groove Mix http://63.249.3.194:8010/ The 80s channel http://64.236.34.106:80/stream/1040/

# **33 BACKUP AND RESTORE**

Backup and Restore are two of the mandatory functions of any application. Elastix is no exception. With this in mind, the developer of Elastix has incorporated this facility in under the System section.

There are 2 ways you can do this. One is through Elastix or if you wish you can do it through the Unembedded FreePBX.

## **33.1 BACKUP AND RESTORE THE ELASTIX WAY.**

First let us look at doing a backup and restore the Elastix way through the Elastix GUI.

#### 33.1.1 Make a Backup

To schedule a backup, from Elastix choose System Tab -> Backup & Restore

	System PBX Fax Email IM Reports	About Elastix 1.0-12 Logou xtras
System Info Network	User Management Load Module Shutdown Hardware Dete	tion Updates Backup/Restore Preferences 20 (
ò System Resou	rces	1
CPU Info:	GenuineIntel Pentium III (Katmai)	5imultaneous calls, memory and CPU
Uptime:	1 hour(s), 06 minute(s)	1.0 2 30 300 Sin. calls

You will then be presented with the following screen:

System (PBX) Fax (Email) IM (Reports) (Extras	About Elastix 1.0-12 Lonout
System Info Network User Management Load Module Shutdown Hardware Detection Updates Backup/Rastore Pre	ferences 🕘 💿
🔢 Backup List	
Backup	
K	1 Start ∠ Previous (0 - 0 of 0) Next ト End №
Delete Name Backup Date	Action
	≪ Start ≺ Previous (0 - 0 of 0) Next ト End \\
Elastix is licensed under GPL by PaloSanto Solutions. 2006, 2007.	

Next select **Backup** as per the illustration above.

The screen that follows is where you define your backup requirement.

System PBX Fax Email IM Reports	About Elastics 1.0-12 Logost
System Info   Network   User Management   Load Module   Shutdown   Hardware	tection Updates Backup/Rostore Preferences O
Backup	
<< Backup List Process	
Lall options	
Elaxtix Database	
Sounds 2	
Configuration Files	
Fax	
U Voicemails	
Email Acccounts	
Elastix is licensed under	PL by <u>PaloSanto Solutions</u> . 2006, 2007.

I have chosen to select ALL option as per the illustration above:

- 1. Place a check mark in the All Options (which will tick everything)
- 2. Click Process

Once the backup process is completed, you will see a screen with the backup filename displayed as per illustration below.

	ystem PB	X Fax	Email IM	Reports Extras			About Flastix 1.0-12	Logout
System Info   Network   User Mana	gement L	oad Module	Shutdown	Hardware Detection	Updates	Backup/Restore	Preferences	0 0
Backup Backup Complete! : backup/elastixbackup-20080331140010-34.tgz								
All options Elastix Database [OK] Configuration Files [OK] Fax [OK] Voicemails [OK] Monitors [OK] Etry [OK] Email Accounts [OK]								
Elastix is licensed under <u>GPL by PaloSanto Solutions</u> . 2006, 2007.								

Once done you may exit the screen.

Naturally you can do as many backups as you want. It is up to you to decide. If you are wondering where it is being stored, it is here */var/www/html/backup*. The backup files are identified by the backup names. Depending on how big your content is the backup can take quite some time.



If this is your first installation, it is advisable to do the backup right away by because should anything happens you can always restore it to its original condition. After this is done, you may start doing your regular backup by following the above procedure.

#### 33.1.2 To Restore a Backup

Backup itself is not useful if it cannot be restored. The freePBX developer has thought of this too.

This is a very simple procedure. All you need to do is click on the appropriate options and the rest are all a matter of choosing what it is that you want to restore (see illustration below).

elasti	System PBX Fax Email	IM Reports Extras		About Elastix 1.0-1	2 Logout
System Info Network	User Management   Load Module   Shutdor	n Hardware Detection	Updates Backup/Restore	Preferences	0 0
🔲 Backup List			1		
Backup			2		
				∣≪ Start ×	Previous (1 - 1 of 1) Next $  \uparrow $ End $  \uparrow  $
Delete	Name Backup		Date		Action
	elastixbackup-20080331140010-34.tgz		31/03/2008 14:0	0:10	Restore
				14 Start 1	Previou 1 - 1 of 1) Next   End
	Elastix	licensed under GPL by P	aloSanto Solutions, 2006, 2007.	3	

- 1. Select System Tab
- 2. Select Backup/Restore option
- 3. Select the backup file you wish to restore and click on the restore button
- 4. As per the illustration below:
  - a. Select whatever components that you want to restore
  - b. Click on the Process button

	System	PBX Fax I	Email IM	Reports Extra	5		About Elastix 1.0-12	Logout
System Info   Network   User	Management	Load Module	Shutdown	Mardware Detection	Updates	Backup/Restore	Preferences	• •
Restore: elastixbackup	-20080331	140010-34.tgz						
< Backup List Process	-	2	2					
All options								
Elastix Database								
Sounds								
Configuration Files								
Eax .								
Voicemails			1					
Monitors								
□ tFTP								
Email Acccounts								
Elastix is licensed under <u>GPL</u> by <u>PaloSanto Solutions</u> , 2006, 2007.								

Whatever components you have selected will now be restored.

Once done you may leave the screen and check the system to make sure that all those that need to be restored have been restored.

## 33.2 BACKUP AND RESTORE THE FREEPBX WAY.

To use this facility you must first insure that the directory **/var/lib/asterisk/backups** exist as is might not. You may have to create it by user **asterisk** if it did not exist.

#### 33.2.1 To Schedule a Backup

To schedule a backup, from freePBX choose *Tools -> Backup & Restore* 

FreePBX 2 40.0 on 192 168.1 37	Admin Reports Panel Help	Loggod In. admin	
Setup Tools Admin		Englis	ih 💌
Support Asterisk Logfiles Online Support FreePBX Support System Administration	No email address for online update checks     48 New modules are available     show all	Processor Load Average 0.27 CPU 0%	
Asterisk API Asterisk CLI Asterisk Info Asterisk Phonebook Backup & Pestore	FreePBX Statistics Total active calls Internal calls Compared Comp	30% Swap 0% Disks 11%	
Custom Destinations Custom Extensions Java SSH PHP Info	Total active channels 0 FreePBX Connections (iP Phone Conine 0 Provide Contine 0 Pro	t 10% Jadeidam 0% Networks ethoraceive 0.33 KBis	
PHPAGI Config Third Party Addon Customer DB Gabcast	Uptime System Uptime: 3 hours, 2 minutes	eth0 transmit 0.76 kB/s Server Status Asterisk DM	
Print Extensions	Asterisk Uptime: 3 hours, 1 minute Last Reload: 0 minutes	Op Panel         DN           MyBolL         DN           Veb Server         DN           SSH Server         DN	
	FreeDOX is a Hightened trademak of Asings, LLC. FreePOX 2.40 is Reared under OPL		

You will then be presented with the following screen:

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FreePBX	Admin Reports Panel Help	
Setup Tools Admin FreePBX System Status	System Backup	English
Module Admin Support		Add Backup Schedule
Online Support FreePBX Support		NOW
System Administration Asterisk API Asterisk CLL		
Asterisk Info Asterisk Phonebook		CARA B
Backup & Restore Custom Destinations		
Custom Extensions Java SSH	FreePBX* Freedom to Connect* FreePIDX is a registered trademark of Atengo, LLC. FreePBX2.4.0 is licensed under OPL	
PHP Info		

#### Next select Add Backup Schedule

The screen that follows is where you define your backup requirement.

FreePBX	Admin Reports Panel Help	
FreePDK 24.0 or 192160.1 27 Sotup Tools Amin PreePDK System Status Module Admin Support PreePDK Support PreePDK Support PreePDK Support System Advansariation Asterisk API Asterisk Info Asterisk Rol Asterisk Info Asterisk Rol Asterisk Rol Asterisk Pronebook Backup & Restore Custom Destinations Custom Extensions Java SSH PHP Info PHPACI Config PHACI Config	Admin       Reports       Panel       Help         System Backup       1       1         System Recordings:       9 yes       no         System Configuration:       9 yes       no         System Configuration:       9 yes       no         Operator Panel       9 yes       no         Operator Panel       9 yes       no         Muntes Hours Day       Montha       Selected         Muntes Hours Day       Montha       Selected         January       January       Mexical         January       January       Weekdays         January       May       Selected         January       January       Weekdays         January       January       Weekdays         January       January       Weekdays         January       January       Selected         January       Selected       Nonday         January       Selected       Nowember         January       Se	Logged in: admit(Gold) English  Children Constraints Add Backup Schedule Restore from Backup NOW
	Submit Changes 4 FreePBX*Freedom to Connect® TreePBX TreePBX to a support to Adversary LLC.	

I have chosen to schedule a *Weekly (on Sundays<sup>t</sup>)* backup as per the illustration above:

- In the Schedule Name, I have given it a name Weekly (to signify that it is a weekly job)
- I have also selected all the items to be backed up VoiceMail, System Recordings, System Configurations and CDR only.
- Next from the Run Backup drop down choice, I have chosen Weekly (on Sundays)
- Once I have done these, I click on **Submit Changes**.

Naturally you can schedule numerous backups. It is up to you to decide. If you are wondering where it is being stored, it is here /var/lib/asterisk/backups. The backup files are identified by the backup names.

#### 33.2.2 To Restore a Backup

Backup itself is not useful if it cannot be restored. The freePBX developer has thought of this too.

This is a very simple procedure. All you need to do is click on the Restore option.

FreePBX	Admin Reports Panel Help	
Cotup Tools		coggeo an annua coggeo
Admin	System Backup	English 💌
FreePBX System Status	System Backup	
Module Admin		Add Backup Schedule
Support		Postore from Packar
Asterisk Logfiles		Resolution Backup
Online Support		NOW
FreePBX Support		
System Administration		
Asterisk API		
Asterisk CLI		
Asterisk Info		
Asterisk Phonebook		
Backup & Restore		
Custom Destinations	And the second se	
Custom Extensions	Freedom to Connect®	
Java SSH	I I III III III III III III IIII III I	
PHP Info	9926e.tak eski birtu aryonalita konor	

Select the backup schedule you want to restore from.

FreePBX 2 4 0.0 on 192 100 1 3	X Admin Reports Panel Help	Logged in admin (1.6940
Setup Tools	System Restore	English
Module Admin	• NOW	Add Backup Schedule Restore from Backup
Asterisk Logflies Online Support FreePBX Support		Now
System Administration Asterisk API		
Asterisk CLI	-	
Asterisk Phonebook		Costo
Disckup & Restore		
Custom Destinations		
Custom Extensions		
Java SSH	FreePBX Freedom to Connect	
PHP Info	FreePBX 2.4.0 in Noemed under GPL	

Select the backup set.

FreePB	Admin, Reports Panel Help	Logged in admin 10 Sports
Sotup Tools President Styles President Styles Commence States President Styles Online Support President Styles Asternsk Proceeding Asternsk Proceeding Asternsk Proceeding Styles	System Restore • DELETE ALL THE DATA IN THIS SET • 200800331.20.39 05 ter gz	Engent *
PHP Info PHPAGI Config Configuration	FreePBX Freedom to Connect®	

Determine which of the various items in the list you want to restore and click on it and your restore will be executed.

<b>FreePB</b>	Admin Reports Panel Help	
FreePBX 2.4.0.0 on 192.168.1.37		Logged in: admin (LogeUt)
Setup Tools Admin FreePBX System Status	System Restore	English 💌
Module Admin	Delete File Set	Add Backup Schedule
Support		Destars from Postars
Asterisk Logfiles	Restore Entire Backup Set	Restore from backup
Online Support		NOW
FreePBX Support	Restore VoiceMail Files	
System Administration	- Desters System Desertings Files	
Asterisk API	Restore System Recordings Pries	
Asterisk CLI	Restore System Configuration	
Asterisk Info		V V
Asterisk Phonebook	Restore Operator Panel	
Backup & Restore		
Custom Destinations	Restore Call Detail Report	
Custom Extensions		
Java SSH		
PHP Info		
PHPAGI Config		

# 34 SUGARCRM

SugarCRM is a contact management software that comes bundled with Elastix it needs to be activated. To activate it, do the following:

- 1. Select Extra Tab
- 2. Select SugarCRM option
- 3. Complete the installation by clicking on the <u>click here</u> link.

(see illustration below)

	stem PBX Fax Email IM Reports F	xtras 1	About Flastix 1.0-12 Log	lout
vTigerCRM Calling Cards Down	ds SugarCRM		•	۲
	The SugarCRM installation is almo 2	st done. To complete it please <u>c</u>	lick here 3	

To set up SugarCRM, first, you need to open the SugarCRM application *http://<your.Elastixip.address>/crm* using the default username of *Admin* and the password of *password*.

For security reason you should change the Admin password. To do this, click on 'My Account' in the upper right-hand corner, then click on the 'Change Password' button underneath 'Users: Administrator (Admin) in the center-left of the screen.

Change it to a new password and confirm your new password and click 'Save.'

Now it's time to set up your contacts. I will start off setting up a couple of my internal extensions.

Click on 'My Account' again and then click the 'Edit' button. Change 'Asterisk Phone Extension' to your Asterisk extension. My extension is 2001.

While you are at it, change your time zone and date format as well.

Click 'Save' to save that information.

Let's add another one.

Click on the contacts tab and then select 'Create Contact' from the left hand Shortcuts menu.

Add another extension, in my case I chose my daughter's extension 2002:

Firstname: Norsurya Last name: Sharif Home: 2002

Click 'Save' to save that information.

Add another and another if you want to, using the method above.

At this point, you may find that you are unable to make a phone call through SugarCRM. This is due to a little bug in the popup\_picker.php (this bug may have been fixed by the time you read this, but at the time of writing, this bug exists).

To fix this bug, you need to edit popup\_picker.php by doing the following:

Elastix Without Tears

From your Linux CLI, log in as root.

cd /var/www/html/crm/modules/Contacts

nano popup\_picker.php

Browse down to line 121 and change it from:

 $snumber = preg_replace ( "/[^\d\*]/", "", $number );$ 

То

```
snumber = preg_replace ( "/[^\d\*]/", "", $display_number );
```

You should now be able to dial from SugarCRM to your other internal extensions and to the outside world.



You can add multiple users who will each have their own settings/contacts etc.

# **35 VTIGERCRM**

VtigerCRM is another contact management software that comes bundled with Elastix it needs to be activated. To activate it, do the following:

(PLACE HOLDER FOR VTIGERCRM)

# 36 IM - OPENFIRE

Under the IM Tab in Elastix, you will find a product called Openfire. In simple terms, Openfire is a very popular Messaging (Chat) and Presence Server, that uses the Jabber/XMPP protocol. You will find that the actual protocol is correctly referred to as XMPP (Extensible Messaging & Presence Protocol).

You might be thinking, what do I want with a Openfire server, what use do I have for it. You actually may be using products that already use the Jabber/XMPP protocol right now such as Google Talk or similar. Jabber/XMPP is not just one product. It is the base for a range of products that use XMPP and as it says, it is Extensible, which means that its capabilities can be increased, especially with Plugins. One of those Plugins is the Asterisk IM (Instant Messaging) which joins the Openfire server to the Asterisk Server and allows them to talk. Naturally with any server product, you need a client before some functionality is realized. There are literally heaps of clients that support the XMPP protocol, but in particular, the one developed by the same developers that designed Openfire, is called Spark. It doesn't mean that it is the best and will suit everyone, but it is able to showcase some of the capabilities of the Openfire server.

So what can we do with Openfire and the Asterisk IM Plug-in and a suitable desktop client such as SPARK?

- Instant Messaging between people on your Network, even Networks joined by a Wide Area Network.
- Instant messaging to external people, using the Openfire Gateway plug-in which allows connection to other messaging systems such as (AIM, ICQ, MSN, Yahoo, etc).
- Presence monitoring. The system can monitor whether you are away from the keyboard, offline, on the phone. This makes it easier for people to see whether to call you or chat to you (if you are on the phone and it is urgent that they reach you). Furthermore, with another additional Plug-in, you can publish this presence information to a Web Page.

On paper it does not seem like a lot, but there are many more applications and Plug-ins that work with Openfire, and it is up to you to decide whether they are suitable to you and make it the product that you want. Even with those items alone, once implemented and integrated into a Business, it makes a huge difference in the efficiency of a business. So let's get started.

## **36.1 OPENFIRE INSTALLATION**

Login in to your Elastix Web GUI and click on the IM Tab and you should be in the following screen.

🕑 Elastix - Mozilla Firefox 3 Beta 5		00	
Elle Edit Yew History Bookmarks Tools Help			Q.
e cot per regory toolmada tools tubb	P		
Cover      Cover	39		
🥥 Disable* 🤱 Cookies* 🗔 CS* 🔄 Forms* 🔳 Images* 🚯 Information* 🛞 Miscellaneous* 🥜 Outline* 🐩 Resize* 🥜 Tools* 🙆 Wew Source* 🔗 Options*		×	0
O elastix System PBX Fax Email IN Reports Extras	About Elastix 1.0-13	Logout	^
OpenFire		• •	
The Openfire service is not active at this moment. If you want to active	ate it please <u>click here</u>		

Click on the CLICK HERE hyperlink and the following screen will appear after a short wait (it is setting up the application for use).

lastix - Mozilla Firefox 3 Be	la 5		
Edit Yew Higtory Bookma	rka Ioola Halo		
- C X O htt	ps://172.22.22.15/?menu=im	🟠 • G.• Google	
iome 📄 Smart Bookmarks 🚞 Pia	ces 🔀 Google 🚳 BoM - New South Wal 😫 Whirlpool - Australian 💷 📄 Joomla Auto Installer 💥 TCP/IP Addressing &	Brasil Vox Telecom 📑 Rochen Host - Multiple.	. 🐻 n'guild : Asteriko, en
Disable• 🤱 Cookies• 🥅 CSS• 💼	] Forms* 🔳 Images* 🕕 Information* 🏐 Miscellaneous* 🥜 Cutline* 💱 Resize* 🥜 Tools* 😥 View Source* 🔑 Options*		×
-			
nelastix 🖉		About Elastix 1.0-13	Logout
6	System PBX Fax Email IM Reports Extras		
penFire			0 0
<i>•</i>			
openfire Set	up		
Setup Progress			
Language Selection	Welcome to Setup		
Server Settings	Welcome to Openfire Setup. This tool will lead you through the initial setup of the server. Before you contin	ue, choose your preferred language	
Database Settings Profile Settings	, , , , , , , , , , , , , , , , , , , ,		
Admin Account	Choose Language		
Setup Progress	Czech (cs_CZ)		
	O Deutsch (de)		
	<ul> <li>English (en)</li> </ul>		
	O Español (es)		
	Français (fr)		
	Nederlands (nl)		
	O Polski (pl_PL)		
	Português Brasileiro (pt_BR)		
	○ 中文 (简体) Simplified Chinese (zh_CN)		
		Continue	

Select the language that you want to use and click on Continue and the next screen will appear

elastix	System PBX Fax Email IM Reports Extras	About Elastix 1.0-13	Legout
openFire	p		0 0
Setup Progress v Language Selection Server Settings Database Settings Porfile Settings Admin Account Setup Progress	Server Settings Below are host settings for this server. Note: the suggested value for the domain is based on Domain: electric homedomain(local ① Admin Console Port: 9999 ① Secure Admin Console Port: 9999 ①	the network settings of this machine.	
	Elastix is licensed under GPL by PaloSanto Solutions. 3	2006, 2007.	

This screen lets you setup your domain name for your system and must match the domain name of your Elastix System. If you haven't setup a domain name correctly on your Elastix system, then drop out of this installation program and do it now. In this guide, you will find a Brief Guide to Networking with Elastix, which explains in reasonable detail how to setup Networking.

If you change the way your system is setup in terms of domain, at a later date, it will break your Openfire application, and while it is possible to recover from this issue, it could take you hours.

With the ports, leave them as they are. They are the default ports.

Click on Continue and the next screen will appear

elastix	System PBX Fax Email IN Reports Extras	About Elastix 1.0-12	Logout
penFire	up		• •
Setup Progress V Languaga Seletion V Server Settinga Database Settinga Defile Settinga Admin Account Setup Progress	Database Settings         Choose how you would like to connect to the Openfire database.         Standard Database Connection         Use an external database with the built-in connection pool.         Embedded Database         Use an embedded database, powered by HSQLDB. This option requires no external database         However, is does not didter the same livel of performance as an external database	mal database configuration and is an easy way to get up and runnin e.	g quickly.
			antinue

In this screen, you have two choices. The first one is a standard Database connection, which unless you have done a large amount of database set-ups, including Java nomenclature, then avoid this.

The second is the favourite, where you press a button and it does it all for you and allows you to bring the product online with a minimal effort.

So the recommendation is select the Embedded Database, and click on Continue.

2 elastix	System PBX Fax Email IM Reports Extras	About Elastix 1.0-13	Logout
openFire	up		0 0
Setup Progress Language fielettinn Seture Tettings Database Istituting Admin Account Setup Progress	Profile Settings Choose the user and group system to use with the server. Default Store users and groups in the server database. This is the best option for simple Directory Server (LOAP) Integrate with a directory server such as Active Directory or OpenLDAP using the read-only.	deployments. LDAP protocol. Users and groups are stored in the directory a	and treated as

This screen allows you to choose where to store your users and groups. For most you will select default, using the database to store your users and groups, which is fine for most set-ups.

The other option is to use the LDAP protocol using and LDAP compliant system to store the users and groups such as Microsoft Active Directory, or another Linux System, where the users, passwords and groups are already implemented. Unless you are very familiar with LDAP, leave it at default.

#### **Click on Continue**

elastix	System PBX Fax	Email IM Reports Extras	About Elastix 1.0-13	Logout
OpenFire	_			0 0
openfire Setu	η			
Setup Progress C Language Selection Server Settings Database Settings Profile Settings Admin Account	Administrator Accou Enter settings for the system example, at least six charac users).	xt administrator account (username of "admin") below. It is imported ters long and containing a mix of letters and numbers. You can	rtant to choose a password for the account that cannot be eavily gue skip this step if you have already setup your admin account (not for	essed for first time
Setup Progress	Admin Email Address: New Password: Confirm Password:	admin@yourentaildomain.com A vald email address for the adtein account.		
			Skip This Step	ntinue

Enter your normal accessible email address as the admin email address. Now the confusing part is that it asks for a password and password confirmation. It has nothing to do with the email address. It is the admin password that you will use for the Openfire system. - For a new user it would have made it easier to understand if they had separated it (still keeping it on the same page as it is to do with the Admin Account), and gave a separate explanation.

Make sure that you use a password that you know well, or at least write it down. You lose it, and it is not a simple measure to recover it.

System PBX Fax Email IM Reports Extras	About Elastix 1.0-13	Leaput
tup		
Setup Complete! This installation of Openfire is now complete. To continue:		
	System PBX Fax Email 14 Reports Extras	System       PBX       Fax       Email       M       Reports       Extras         tup       Setup Complete!       This installation of Openfire is now complete. To continue:         Login to the admin console       Setup

Click on continue

That's the first part. You now have functional Openfire Server, which you can login in to and setup further.

Click on LOGIN TO THE ADMIN CONSOLE and you will get the following screen, which will allow you to login.

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Login with the username admin and the password is the one that you selected back at the admin account. Remember, that was one which you were told to write it down and don't lose it.

The next screen will confirm that you indeed have a functional Openfire Server running.

Server Users	Groups Sessi	ons G	roup Chat Pl	igins Enterp	nse	Logout (admi		
r Nanager er Information m Properties lage and Time erring s Summary ase Settings Settings r Settings r Settings r to Servier and Components errion Managers	Server Info Below you will Server Prop Server Up Ver Server Direc Server N Environmen Java Ver Appse OS Hardo	rmation find serve erties time: 6 m sion: Op tory: /op ame: ela sion: 1.6 sion: 1.6 sver: jett vare: Lin	er information, p ninutes started entire 3.4.2 bogenfire stor. netintegrity.lo 0_03 Sun Micros e6.1.x wc/1386	orts being used a Apr 5, 2000 6:03. <del>1</del> cal ystems linc Jav	ind latest news about Openfire. 54 AM 19 HotSpot(TNI) Server VM	Ignite Realtime News		
Binding ge Updates tration & Login urce Policy e Messages age Audit Policy e Data Storage	Locale / Time: Java Mer Server Port	tone: en mory <b>m</b>	/ Eastern Standa	d Time (New Sou 0.14 MB of 63.31	m Wales) (10 GMT) MD (16.0%) used	Cpenfire 3.4.5 has been released, Peb 9, 2000		
ity Settings	Interface	Port	Туре	De	escription			
r Certificates ression Settings ransfer Settings	All addresses	5222	3 Client to S	erver Th	e standard port for clients to connect to the servi ttings for this port.	er. Connections may or may not be encrypted. You can update the $\underline{\operatorname{security}}$		
h Service rties	All addresses	5223	Client to S	erver Th me	e port used for clients to connect to the server use thod and will be deprecated in the future. You c	sing the old SSL method. The old SSL method is not an XMPP standard an update the <u>security settings</u> for this port.		
services	All addresses	9090	Admin Cor	isole Th	e port used for unsecured Admin Console acces	9.		
Proxy	All addresses	9091	Admin Cor	sole Th	e port used for secured Admin Console access.			
Settings	All addresses	7777	File Transf	er Proxy Th	e port used for the proxy service that allows file t	ransfers to occur between two entities on the XMPP network.		
	All addresses	8080	HTTP Bind	ing Th	e port used for unsecured HTTP client connectio	nş.		
	All addresses	8483	- HTTP Bind	ing Th	e port used for secured HTTP client connections			
		3478	OTHER COM	in Th	a wat used for the couries that answers connection	ity koturee estilies when babied a NAT		

Don't panic, there is a lot on this screen, but we will take you through the few basics that you need. Don't upgrade Openfire yourself; these updates will come from Elastix. Remember that they have integrated it into the Elastix system, and there may be minor tweaks to allow this to happen. These minor tweaks however, may be the difference between your Openfire Server working or not working.

We are now going to add some Plugins into the system to make it functional. So in the Openfire Console, click on the Plugins Tab, and the following screen will come up

Server Use	rs/Groups Sessions Grou	ap Chat Plugins Enterprise				offensi fau
gin Admin	Plugins					
ilable Plugins	Plugins add new functional	fity to the server. The list of plugins currently installed is below. To download new pl	lugins, please visit the <u>Availabl</u>	e Plugina page.		
	Plugins	Description	Version	Author	Restart	Delet
	Search	Provides support for Jabber Search (XEP-0055)	1.4.1	Ryan Graham	2	0

This screen shows that there is only one Plugin installed by default. We are now going to quickly add a few more.

So click on Available Plugins from the side menu and the following screen will appear.

lable Plugins	Plugins add new functionality to the se in the list until it is actually installed.	rver. The li	st of plugins available to install is below. Once a plugin is downloaded it may	take a mon	nent to be installed. Th	ie plugin will s	itill appear
	Open Source Plugins		Description	Version	Author	File Size	Install
	Asterisk-IM Openfire Plugin		Integration for Asterisk and Openfire.	1.4.0	Jive Software	426.0 K	0
	🐨 Broadcast		Broadcasts messages to users.	1.7.0	Jive Software	19.7 K	0
	💎 Content Filter		Scans message packets for defined patterns	1.5.0	Conor Hayes	17.0 K	0
	🄺 IM Gateway		Provides gateway connectivity to the other public instant messaging networks	1.2.2a	Daniel Henninger	1.0 MB	0
	MotD (Message of the Day)		Allows admins to have a message sent to users each time they log in.	1.0.3	Ryan Graham	11.9 K	0
	Presence Service		Exposes presence information through HTTP.	1.4.0	Jive Software	17.5 K	0
	🐍 Registration		Performs various actions whenever a new user account is created.	1.4.1	Ryan Graham	22.1 K	0
	SIP Phone Plugin		Provides support for SIP account management	1.0.2	Ignite Realtime	186.6 K	0
	<ul> <li>Subscription</li> </ul>		Automatically accepts or rejects subsription requests	1.2.0	Ryan Graham	14.2 K	0
	🔏 User Import Export		Enables import and export of user data	2.2.0	Ryan Graham	280,1 K	0
	User Service		Allows administration of users via HTTP requests.	1.3.0	Justin Hunt	14.2 K	0
	Commercial Plugins						
	👄 Openfire Enterprise		Enterprise Edition of Openfire.	3.4.5	Jive Software	5.4 MB	0
	5 Spark Fastpath Webchat	10	Web based chat client for Spark Fastpath.	3.4.5	Jive Software	2.1 MB	0

What you see in this list is a list of all Plug-ins that have been registered with Openfire. You may find this list increase, allowing you to add another new plug-in that might be suitable for your system. You can click on the UPDATE NOW hyperlink, if you want to confirm if there are any more available.

You can also manually upload Plugins, e.g. ones that are not part of the registered Plugins, but we don't require that at this stage.

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Now to install the Registered Plugins, all we need to do is click on he Green + Plus sign, under the heading install. So the first one we are going to install is the Asterisk-IM-Openfire-Plugin, so click the Green Plus Sign on that line and the following screen will appear.

Avall Plugin in the l	able Plugins s add new functionality to the ser ist until it is actually installed.	ver. The li	st of plugins available to install is below. Once a plugin is downloaded it may	take a mon	nent to be installed. Th	e plugin will s	itill appea
	Open Source Plugins		Description	Version	Author	File Size	Install
۲	Asterisk-IM Openfire Plugin		Integration for Asterisk and Openfire.	1.4.0	Jive Software	426.0 K	O
10	Broadcast		Broadcasts messages to users.	1.7.0	Jive Software	19.7 K	0
۴	Content Filter		Scans message packets for defined patterns	1.5.0	Conor Hayes	17.0 K	0
*	IM Gateway		Provides gateway connectivity to the other public instant messaging networks	1.2.2a	Daniel Henninger	1.0 MB	0
ø	MotD (Message of the Day)		Allows admins to have a message sent to users each time they log in.	1.0.3	Ryan Graham	11.9 K	0
μ	Presence Service		Exposes presence information through HTTP.	1.4.0	Jive Software	17.5 K	0
2	Registration		Performs various actions whenever a new user account is created.	1.4.1	Ryan Graham	22.1 K	0
ø	SIP Phone Plugin		Provides support for SIP account management	1.0.2	Ignite Realtime	186.6 K	0
++	Subscription		Automatically accepts or rejects subsription requests	1.2.0	Ryan Graham	14.2 K	0
-6	User Import Export		Enables import and export of user data	2.2.0	Ryan Graham	280.1 K	0
2	User Service		Allows administration of users via HTTP requests.	1.3.0	Justin Hunt	14.2 K	٥
	Commercial Plugins						
-	Openfire Enterprise		Enterprise Edition of Openfire.	3.4.5	Jive Software	5.4 MB	0
5	Spark Fastpath Webchat		Web based chat client for Spark Fastpath.	3.4.5	Jive Software	2.1 MB	٥

You will note that the Green Plus sign turns into a little "Loading Timer", and may take a short while to install. Once completed, it will automatically remove itself from the screen, as it has now ended up in the Plugins

Do the same for, IM gateway, Presence Service and SIP Phone Plugin if they exist in the list. If you now click on the Plugins on the left menu, you will see the Plugins that you have installed.

gin Admin ugins railable Plugins	Plugins Plugins add new functionality to the ser	ver. The li	st of plugins currently installed is below. To download new plugins, please visit	the <u>Availab</u>	le Plugins page.		
	Plugins		Description	Version	Author	Restart	Delete
	Asterisk-IM Openfire Plugin		Integration for Asterisk and Openfire.	1.4.0	Jive Software	æ	0
	🔺 IM Gabeway		Provides gateway connectivity to the other public instant messaging networks	1.2.2a	Daniel Henninger	æ	0
	Presence Service		Exposes presence information through HTTP.	1.4.0	Jive Software	2	0
	SIP Phone Plugin		Provides support for SIP account management	1.0.2	Ignite Realtime	2	0
	🧠 Search		Provides support for Jabber Search (XEP-0055)	1.4.1	Ryan Graham	æ	0
	Upload Plugin Plugin files (jar) can be uploaded direct Browse	tly by usin	ig the form below. Plugin				

You can also delete the Plugins from this screen or restart them if you need to. You will also notice that we now have a new tab at the top called ASTERISK-IM, which is the result of installing the ASTERISK-IM plugin.

Click on that ASTERISK-IM Tab and the following screen will appear.

1. S.	ups sessions aroup the	Asterisk-19	Enterprise				
sk-IM ral Settings G	eneral Settings						
e Mappings U	se the form below to edit Phone	integration settings. Changi	ng settings will cause the	plugin to be reloaded.			
7	Asterisk-IM: OEnabled ODis	abled					
	Name	Address	Port	Username	Options		
	Asterisk IM Not Enabled						
	C						
-	Asterisk Queue Presence	O Ves @ No					
	Drop-down device selection:	() Yes () No					
	Actanick Contant	\$ 105 \$ 115					
	Default Caller ID:						
	Dial Command Variables:						
	Firstlen Timenut						

Click on the Asterisk-IM Enabled Radio Button, click on the Asterisk Queue Presence Radio Button and add the word **default** to Asterisk Context input Box similar to the following screen

General Settings Use the form below to edit Ph	one integration settings. Changin	ig settings will cause the	plugin to be reloaded.	
Asterisk-IM: <ul> <li>Enabled</li> </ul>	Disabled			
Name	Address	Port	Username	Options
		Asterisk I	M Not Enabled	
Configure Phone Manage	1			
Asterisk Queue Presence:	Yes O No			
Drop-down device selection	e Yes O No			
Astensk Context:	detouit			
Default Caller ID:		_		
Dial Command Variables:				
Firstleg Imeout:				
Save				

Once you have done this, then click SAVE

Now before we can go any further, we have to correct a bug in the Asterisk-IM Plugin, otherwise we cannot connect the Openfire Server to the Asterisk system.

This means either using the Elastix System Console or using SSH to login to the Linux system.

Once you are logged in, we need to edit a file, only a small change, but it needs to be done. If you are well ahead and have already installed your favourite editor onto your Elastix system, then use your favourite editor, but if not you can download and install NANO, our favourite editor (see TOOLS section of this document on how to install NANO).

Next we need to edit the file. So issue the following commands

cd /opt cd openfire cd Plugins cd asterisk-im cd database nano asterisk-im\_hsqldb.sql *(if you are using nano)* 

Similar to the picture below

```
[root0elastix /]#
[root0elastix /]#
[root0elastix /]# cd /opt
[root0elastix opt]# cd openfire
[root0elastix openfire]# cd plugins
[root0elastix plugins]# cd asterisk-im
[root0elastix asterisk-im]# cd database
[root0elastix database]# nano asterisk-im_hsqldb.sql<mark>.</mark>
```

You will get a screen very similar to the following



Basically we have to remove the word **unique** from the third line as shown in the picture above, which is causing the bug we spoke about previously.

Using arrow keys, just move along and use the delete keys so that your screen now looks like the following

GNU nano 1.3.12 File: asterisk-im_hsqldb.sql Modifi	ied
<pre>create table phoneServer (     serverID bigint not null,     serverName varchar(255) not null,     hostname varchar(255) not null,     port integer not null,     username varchar(255) not null,     password varchar(255) not null,     constraint phoneServer_pk primary key(serverID) );</pre>	
create table phoneDevice ( deviceID bigint not null, device varchar(255) not null, extension varchar(255) not null, callerID varchar(255), isPrimary integer not null, userID integer, serverID bigint not null, constraint phoneDevice_pk primary key (deviceID)	
) ;	
create table phoneUser ( userID bigint not null, username varchar(255) not null, constraint phoneUser_pk primary key (userID) ); create unique index phoneUser_username_idx on phoneUser(username);	
INSERT INTO jiveVersion (name, version) VALUES ('asterisk-im', 2);	
◇G Get Help ◇O WriteOut ◇R Read File ◇Y Prev Page ◇K Cut Text ◇C Cur Pos ◇X Exit ◇J Justify ◇W Where Is ◇V Next Page ◇U UnCut Text◇T To Spell	

Now hit CTRL-X, which will allow us to exit Nano, and also prompt us to save the file.

Once you hit CTRL X, you will get the following screen



Enter Y for yes, which means that we are going to save the changes and the following screen will appear.

GNU nano 1.3.12	File: ast	erisk-im_hsqldb.sql	Modified
create table phoneSe serverID bigint serverName varch hostname varchan port integer not username varchan password varchan constraint phone );	rver ( not null, ar(255) not null; (255) not null, (255) not null, (255) not null, (255) not null, eserver_pk primar	y key(serverID)	noalliea
<pre>create table phoneDe deviceID bigint n device varchar(25 extension varchar callerID varchar isPrimary integer, userID integer, serverID bigint n constraint phoneI ); create table phoneUs userID bigint not username varchar constraint phoneI );</pre>	evice ( hot null, 55) not null, (255), r not null, hot null, pevice_pk primary ser ( r null, (255) not null, Jser_pk primary k	key (deviceID) ey (userID)	
create unique index INSERT INTO jiveVers	phoneUser_usernat	me_idx on phoneUser( on) VALUES ('asteris	username); k-im', 2);
File Name to Write:	asterisk-im hsql ^T To Files	db.sql M-M Mac Format	M-P Prepend

You will note that it has used the original name of the file, so all we need to do here is press the ENTER key and **Nano** will save the file and exit the program.

You have now completed the changes which will allows us to complete the Asterisk-IM implementation.

**Reboot your Elastix!!!** – It is important that you do this so that it picks up the changes.

Now go back to your Elastix Web GUI, click on the IM tab to bring up OPENFIRE and then click on the ASTERISK-IM Tab and then click on **Add Server** hyperlink which will take us to the following screen.

Ele Edit Yew History D	ookmarks Iools Help				0
S → C × C	Mtp://172.22.22.15:90	90/plugins/asterisl-im/create-server	(ISD)	G • G • Scoole	£
🔿 Home 🏠 Smart Bookmarks 🛙	- Places 🔀 Google 🍩 IS* 📰 Forms* 😹 Image	BoM - New South Wal 🔛 Whit s* 🕐 Information* 🗇 Miscellane	pool - Australian 🚺 Joomia Auto Installer 💥 TCP/IP Addressing b rous* 🥜 Outline* 📜 Resize* 🧈 Tools* 🛐 View Source* 🧽 Options*	. 🗋 Brasil Vox Telecom 🗋 Rochen Host - Multiple	🗰 n'guld : Asteriko,en 🔅 🗙 😋
Openfire Admin	istration Console:				
Server Divers	Groups Sessions	Group Chat Plugins	Asterisk-IM Enterprise		Logeut (edmin)
Asterisk-IN General Settings Phone Mappings	Create Phone : Add a connection to	Server o a new phone server.			
	Server Name:				
	Server Address:				
	Port	0			
	Username:				
	Password:				
		Create Server Cancel			
			Opentine 3.4.2. Italit by Jive Software and the landeficeatine or of	community	
			opening one at one of the polyage and the president ord	Contraction in p	

Add the following items into the input boxes

Server Name:	elastix.yourdomain.local
Server Address:	127.0.0.1
Port:	5038
Username:	admin
Password:	elastix456

(Same as your Elastix box domain) (Always this for this install) (This is the Asterisk AMI Port number) (this is a user in manager.conf) (this is the default password)

So you screen when filled in, should look this

Openfire Admini	istration Console 'Groups Sessions	Group Chat Plugins Asterisk-IM Enterprise
Asterisk-IM General Settings Phone Mappings	Create Phone	Server o a new phone server.
	Server Name:	elastix.netintegrity.local
	Server Address:	127.0.0.1
	Port:	5038
	Username:	admin
	Password:	•••••••
		Create Server Cancel
	Emmananananananananananana	

Click on Create Server and if everything is successful, you should see the following screen

Server Users	/Groups Sessions Group Cha	at Plugins <mark>Asterisk-I</mark> M	Enterprise				Logout (admin
Asterisk-IM General Settings Phone Mappings	General Settings Use the form below to edit Phon Server created successfu Asterisk-IM: ©Enabled ODi	e integration settings. Chang <b>Ily</b> sabled	ing settings will cause the plug	in to be reloaded.			
	Name		Address	Port	Username	Options	
	<ul> <li>elastix.netintegrity.local</li> </ul>		127.0.0.1	5038	admin	2 3	
	Add Server						
	Configure Phone Manager Asterisk Queue Presence: Drop-down device selection: Asterisk Context: Default Caller ID: Dial Command Variables:	Yes     No     Yes     No     default					

Take special note of the green dot next to the name of our Elastix box. If it is green then the Openfire server has a working connection to our Elastix Server. If the server address is missing altogether, then you either did not reboot the Elastix Server (which meant the bug fixes are not applied), or you have not edited the file correctly.

If the green dot is actually grey, then you have correctly edited the file, but it appears that for some reason you have not correctly connected to the Elastix Server. This may be the result of the user and password not being set correctly for the Asterisk Management Interface. The ones that we have provided in this chapter are ones that are setup by default by Elastix. If you have changed *manager.conf* under /etc/asterisk, you will need to correct the login and password to suit.

So if you have the green dot, you now have a working Openfire Server connected to your Elastix Server. All we need to do now is add users and install the client on the desktops. Click on users and groups tab at the top and the following screen will appear

Server Users	/Groups Sessions Group Chat F	Plugins Asterisk-IM Enterprise			Logout [adm
Users User Summary Create New User User Search	User Summary Total Users: 1 Sorted by Username	Users per page: 15 💌			
Advanced User Search	Online Username	Name	Created	Last Logout	Edit Delete
<b>Groups</b> Group Summary Create New Group	1 & admin	Administrator	Apr 6, 2008		2 3

Only the admin user will appear. We now need to create users for your system. Click on create new user on your system. And the following screen will appear.

<b>o</b> penfire Admir	nistration Console
Server Users	/Groups Sessions Group Chat Plugins Asterisk-IM Enterprise
Users User Summary Create New User User Search	Create User Use the form below to create a new user.
Advanced User Search	Create New User
Groups Group Summary Create New Group	Username: *  Name: Name: Email: Password: * Confirm Password: * Create User Create & Create Another Cancel * Required fields

Fill in the details for each user you want to connect to Openfire. Keep the usernames in lower case (makes it easier), fill in their proper name, their current email address, and provide them with a password. This password does not have to match anything, it will be used by the client that resides on their desktop to connect to the Openfire server.

You should now have a screen like the illustration below:

<b>o</b> penfire Admir Server Users	istration Console	Group Chat Plugins Asterisk-IM Enterprise
Users User Summary Create New User User Search Advanced User Search	Create User Use the form below Create New User	to create a new user.
<b>Groups</b> Group Summary Create New Group	Username: * Name: Email: Password: * Confirm Passwor Create User	bob         Bob Fryer         bob@netintegrity.com.au         ••••••••         d: *         ••••••••         Create & Create Another         Cancel

Click on CREATE USER (or Create and Create Another if you want to keep adding more).

After you have done this, you should see the screen (we have only done one user) like this

<b>o</b> penfire Admini	stration Console	
Server Users/	Groups Sessions Group Ch	at Plugins Asterisk-IM Enterprise
Users User Summary User Options • User Properties • Roster • Password • Delate User	User Properties Below is a summary of user prop New user created successfu	perties. JII y.
Create New User	User Properties	
User Search	Username:	bob
Advanced User Search	Status:	🚨 (Offline)
Groups	Name:	Bob Fryer
Group Summary Create New Group	Email:	bob@netintegrity.com.au
credit New Group	Registered:	Apr 6, 2008
	Groups:	None
	Edit Properties	

We have now setup one user on the system. For the system to recognise when we are on the phone, we need to map the user to an extension

We do this by clicking again on the Asterisk-IM Plugin Tab and then clicking on Phone Mappings on the left menu. You should see a screen like so:

<b>o</b> penfire Adr Server Use	ninistration Console rs/Groups Sessions Group Cl	hat Plugins <mark>Asterisk-IM</mark>	Enterprise	_	
Asterisk-IM General Settings Phone Mappings	Phone Mappings Total Users: 0 Sorted by Use	ername - Users per Page: 15	~		
	Username	Device	Extension	Caller ID	Options
	No User/Device Mappings				
	Add User/Asterisk Phone	mapping			
	* Device: Select v or				
	* Extension:				
	Caller ID:				
	Primary:				
	Add Cancel				
	* Required fields				

We are now going to setup a phone mapping for the user that we just setup under Users and Groups.

Username is the username you setup, in this case it was bob (remember I said to setup in lowercase, it just makes it easier, as the system will not recognise if you use an uppercase char, it sees it has a different login).

The device is the actual phone, and you should be able to drop down the box and it will show extensions that you have in your Elastix System. If it doesn't show, then you can enter it manually (e.g. for our one user we would add **SIP/301).** Then add the extension number which is the same, without the SIP/, so we would enter **301** in here, and then a caller ID. I normally enter 301 in here as well. You can click on the primary field as well if you like, but it is not crucial. This does have a purpose, but it is for more complex systems, which are beyond the scope of this document.

<b>o</b> penfire Adr Server Use	ninistration Console rs/Groups Sessions Group	Chat Plugins Asterisk-IM	Enterprise		-
Asterisk-IM General Settings Phone Mappings	Phone Mappings  Operation completed su Total Users: 1 Sorted by U	ccessfully. sername - Users per Page: 15	v		
	Username	Device	Extension	Caller ID	Options
	bob	SIP/301	301	301	2 3
	Add User/Asterisk Phon Username: Device: Select v Caller ID: Primary: Add Cancel	e mapping			

After you have pressed the add button, you should see the following user setup

You have now successfully mapped a user and phone together.

## **36.2 INSTALL SPARK CLIENT.**

As mentioned the SPARK Client from the same people that developed Openfire, is a good starting point. You can always change the client later, whenever you want, and by then, you will know what you are looking for in a client.

So for now we are going to use SPARK. You can download it from <a href="http://www.igniterealtime.org/projects/spark/index.jsp">http://www.igniterealtime.org/projects/spark/index.jsp</a>

Install it as per default install instructions. And you should end up with the following screen





Now we enter in the details that we setup as a user

Username: **bob** (remember the lower case user name) Password: **Whatever** (the password that you setup for that user) Server: The IP address of your server (not the 127.0.0.1) address - in our case **172.22.22.100** 

#### 36.2.1 Enabling Remote Client

Username: bob

Server:

To enable remote client, do the following:

Password: (the password you setup for that user)

You must open your Router port 5222 and direct

In my case, my DynDNS namae is: *pcnovation.homelinux.org* 

it to your Elastix IP address.

Your server DNS (or DynDNS)

Spark

Click on Save Password and click on Auto Login.

And then click on Login. If all is successful, you will end up with the following screen

www.hellodigi.ir

Spark - bob	_ 🗆 ×
bark Contacts Actions Help	
bob	
😝 Available 🔻	
i 🤏 🥥 😒	
▼ Offline Group	
💰 Contacts 🧠 Conferences	
Search for other people on the server.	

Our client has now successfully connected to the Openfire server. If we had more users installed, we would see the users listed, showing their presence status, whether they are offline, on the phone, away from keyboard etc.

# **Note.** In the mode that we have shown, you will have to accept users and they you, before they will show up on your list (very much like ICQ and similar).

If this is not what you want, and you want all the people that are on your local Network to be immediately available to each SPARK user, then you can set them into Groups (a subject we did not broach).

If you go back to Openfire Users and Groups Tab, create a Group Name and add the selected users to the Group and they will be immediately available to communicate with if they are members of that group.

There are many more features, and functions within Openfire. It deserves a book all by itself, which again is not the purpose of this document. We hopefully have provided enough to get you started, so that you can explore Elastix and the integrated Openfire server.

# **37 ELASTIX EMAIL SYSTEM AND WEBMAIL**

The following step will prepare your Elastix webmail.

## 37.1 ELASTIX MAIL SYSTEM – SETUP & USE

Elastix developers have included the following products within the Elastix System:

- PostFix Well known and tested Mail Server software
- **RoundCube Webmail** Generally regarded as one of the best Webmail Clients

Again like most items included in Elastix, they have designed an interface, to reduce the amount of "manual" changes to configuration files.

However, having said that, the topic of setup of email is never easy, as everyone's Network and Mail host/design, and even ISP are different, so it is impossible to describe a step by step setup that will suit all, and some manual changes may be required.

To this end, we are going to discuss the setup of four types of Email setup. These are:

- 1. **Email Directly Via Postfix** using your Elastix Postfix to send out mail direct using Postfix as a SMTP Server (including relaying via your ISP/Webhost)
- 2. **Email Directly Via Postfix with Authorisation** Setup of Elastix postfix to relay mail via your external mail server which requires SMTP-Auth
- 3. Email via your local Mail Server Using your local Network Mail server. This is typical of most business networks where they have a proper Network infrastructure, with Internal DNS Server, Mail Server (Microsoft Exchange or other completely configured mail server e.g. Postfix, Sendmail, QMail etc)
- 4. Email using your Elastix Postfix Server and Roundcube This works on the basis that you have a domain name and you choose to make the Postfix Server the actual mail server for that domain.

Before we start, I just want to cover a small bit of concept, which may make it easier as you read the following sections.

One of the biggest misconceptions, especially from people new to the area of mail servers is the use of the domain names. I have seen network systems setup to use the external domain name that their company owns, as they believe that is the one that they are supposed to use.

You will note that this guide talked about the **.local** domain. It is a "dummy" domain basically. It still meets all the needs of a domain name and DNS internally, but it is not routable, in other words it does not have a root name server in the Internet, unlike the **.com, .com.au, .net, .org** etc.

The **.local** is a domain name that we can use locally on our network, very similar to the private IP address ranges that we use, that are generally not routable on the internet. Just because we have used the **.local** domain name, does not mean it cannot handle parts of the .com domain name as well. As you will see from this guide, our .com Website, can reside on a Webserver somewhere on this Internet, while our mail server can reside on our Elastix Server, in a different location.



mail servers, the users on the network are performing POP3 receives on a regular basis, all through the External IP address of their Internet connection, so to the ISP/Webhost it is seeing someone authenticate using POP3 every few minutes, and providing your Postfix server is sending out mail with 30 secs of one of these POP3 receives, your mail will go out and be accepted, miss a window and your mail fails. At night it is worse, as the mail users have gone home, turned off the PC, and no one is doing any POP3 receives, so all your mail from Elastix system fails. You get back in the morning and it is working again.

You might wonder why all the hassle with trying to send email via your own server, but over the last few years it has become necessary as ISP/Webhosts have tried to stem the spread of SPAM email. In some cases, it is a legal requirement. If they do not try to limit the SPAM coming out of their system, or IP addresses that they have control over, they can be seen as promoting SPAM, which in turn can lead to possible legal ramifications.

If on your initial installation of Elastix, your emails work then great, you are probably one of the lucky ones that can send out email direct, but they are generally not the rule.

So that we don't go off on the wrong track, it is necessary to look at the mail log to see why your email failed. In most cases it will tell you what went wrong. You will need to SSH into your system and get to the Linux Operating System or use your Console screen after you have logged in. The file you want to look for is **/var/log/maillog**. So at the prompt type **less /var/log/maillog**, and scroll up and down until you find the error close to the time and date that the system tried to send an email. You will usually identify easily as it will also have the email address in it, where the email was supported to go.

An example of a log file from a system that cannot send an email is as follows:

Apr 24 12:15:12 elastix postfix/smtp[30137]: 3D2EBCC5908: to=<peter@myrealdomain.com.au>, relay=myrealdomain.com.au[74.12.87.146]:25, delay=1.9, delays=0.02/0.01/1.5/0.36, dsn=5.0.0, status=bounced (host mywebsite.com.au[74.12.87.146] said: 550-Verification failed for uucp@myrealdomain.local

What it tried to do here is send an email to **peter@myrealdomain.com.au**. As we have told it nothing of how to send the email, it has made a smart choice and tried to use the myrealdomain.com.au mail server. Of which it has promptly replied Verification failed (that's because we have not setup any authentication).

So now we know it is not going to work in a default mode, so we have to make some changes to the Postfix config file. We are now going to tell Postfix to route the mail via our ISP's mail server. Your ISP is going to be whomever you have an Internet connection through e.g. Verizon might be *mail.verizon.com*, Westnet might be mail.westnet.com.au (which is the one that we are using). So the changes in the next part include a relay host (in other words a place that we route mail through), and the correct setup of the domain names for the mail. The file is found in the /etc/postfix directory and is called main.cf. So issue the following commands

```
cd /etc/postfix nano main.cf
```

#### Change

```
#myhostname = host.domain.tld
#myhostname = virtual.domain.tld
```

to

```
myhostname = elastix1.myrealdomain.local
#myhostname = virtual.domain.tld
```

and a few lines further down change

#mydomain = domain.tld

to

mydomain = myrealdomain.com.au

and quite a bit further down

#relayhost = \$mydomain
#relayhost = [gateway.my.domain]
#relayhost = [mailserver.isp.tld]
#relayhost = uucphost
#relayhost = [an.ip.add.ress]

change to

```
#relayhost = $mydomain
#relayhost = [gateway.my.domain]
relayhost = mail.westnet.com.au (remember to take the # out and make it your ISP)
#relayhost = uucphost
#relayhost = [an.ip.add.ress]
```

After you have completed the config changes, then issue the following command.

#### postfix reload

After this, it was expected that we should be working, but upon looking at the logs we got the following:

Apr 25 14:56:59 elastix postfix/smtp[7959]: C3797CC5911: to=<peter@myrealdomain.com.au>, relay=mail.westnet.com.au[202.72.128.70]:25, delay=1.3, delays=0.12/0.05/0.68/0.48, dsn=4.0.0, status=deferred (host mail.westnet.com.au[202.72.128.70] said: 450 <uucp@myrealdomain.local>: Sender address rejected: Domain not found (in reply to RCPT TO command))

The issue was the sender address, our ISP rejected it on the grounds that it recognises that the sender address is not a valid domain.

We finally found the issue was down at the bottom of the main.cf file. The developers in their wisdom have written these entries down the bottom of the file, not where I normally expect them, so basically they were overwriting the ones we had set above. From what I can read, they have been included for Webadmin??

So you need to change the main.cf file again and also add them to the bottom.

Save it and do a **postfix reload** again, and lets look at the mail log again after leaving a voicemail to email or fax to email, and see what a proper one looks like.

```
Apr 25 16:45:31 elastix postfix/qmgr[8516]: EF5C9CC5921: from=<uucp@myrealdomain.com.au>,
size=6382, nrcpt=1 (queue active)
Apr 25 16:45:33 elastix postfix/smtp[8547]: EF5C9CC5921: to=<peter@myrealdomain.com.au>,
relay=mail.westnet.com.au[202.72.128.70]:25, delay=1.5, delays=0.02/0.01/0.28/1.2, dsn=2.0.0, status=sent
(250 Ok: queued as 8E7453B913D)
```

So you are probably asking why did I include the mistake in their as well, when I could just tell you to go to the bottom of the file and change the domain entries. I left it in there to show you basic fault finding with the Email system. Even someone who has done it many times before, can still forget the basics, and needs to fault find the issue.

That's it for the first method of implementing the mail system using the Direct From Postfix method. Test your fax to email and also your voice to email. It uses the same method and should function well.

## **37.3 EMAIL DIRECTLY VIA POSTFIX WITH AUTHORISATION**

As mentioned, sometimes there is a requirement to provide authorization to your External SMTP server. Some occasions where this is needed is that you have an ISP that under all circumstances requires authorization to send mail out through the External SMTP Server.

Another time this is required, is when you cannot use your ISP's SMTP server (due to ongoing issues with your ISP SMTP Server or similar (e.g. timeouts etc), and you wish to use the Webhosts mail server, where your POP accounts currently reside. Understandably, almost every Webhost requires you to use SMTP-Auth to move your mail through their SMTP Server (or they use the method previously spoken about which is POP before SMTP).

So to start with, you would have performed almost everything in the previous section on using email via PostFix directly, and you have use the following line in main.cf.

**relayhost = mail.mywebhost.com.au** (naturally you would be using your hosted web address).

instead of pointing it to your ISP Mail Server, and now you are up to the point where your mail log file is showing the following or similar, but the important line is **550 - Verification Failed.** 

Apr 24 12:15:12 elastix postfix/smtp[30137]: 3D2EBCC5908: to=<peter@myrealdomain.com.au>, relay=myrealdomain.com.au[74.12.87.146]:25, delay=1.9, delays=0.02/0.01/1.5/0.36, dsn=5.0.0, status=bounced (host mywebsite.com.au[74.12.87.146] said: 550-Verification failed for uucp@myrealdomain.local

So, we now have to setup Postfix for SMTP-Auth which we will describe in this section.

Right now we are going to create a password file in /etc/postfix directory by doing the following:

nano /etc/postfix/sasl\_passwd

and in this file add the following

mail.mywebhost.com.au loginname:password

save the file, that's the first part. Now we are going to convert this to a database for Postfix to read by the following command

postmap hash:/etc/postfix/sasl\_passwd

Now to add the final bit of information for the process to work, so edit the main.cf in the postfix directory, by the following command

nano /etc/postfix/main.cf

Add the following lines

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```
smtp_sasl_auth_enable = yes
smtp_sasl_password_maps = hash:/etc/postfix/sasl_passwd
smtp_sasl_security_options =
```

and save the file. Now to reload Postfix and try your SMTP-Auth mechanism, so reload by the command

postfix reload

and test it out, remembering to check the /var/log/maillog after each send to confirm what is actually happening.

If indeed you needed to authenticate this, should get you up and running, but just in case, there are plenty of resources on the Web about this process, so have a look.

## 37.4 EMAIL VIA YOUR LOCAL MAIL SERVER

This is probably the easiest section to write. If you have a network in place that follows industry standards, which includes an Internal DNS Server and Internal Mail server, then you have already realized that Elastix, via Postfix already is sending mail via your mailserver.

Postfix will try and determine the MX address, which if the DNS Server is configured correctly, it will respond with the Internal Mail Server in your organisation. As a guide, anyone running Microsoft Server Suite, especially Small Business Server, is probably setup in this way by default. So as soon as Postfix gets a reply, it sends mail out via that Server.

As it is an internal server, you will probably find it is a little more forgiving on Senders email addresses, but that it purely dependent on the version, and how strict the IT administrator is and whether he has really locked it down.

If this is the case, you may find that you may have to utilise a couple of methods in the previous chapters. As an example, Microsoft Exchange server can be configured to allow unauthenticated SMTP mail to pass through as long as it is from their local range of LAN IP addresses. Others say that you must authenticate, and have removed the ability to send mail without authentication. This means that you need to get a mail user (or an existing user) setup on the mail server, and use the SMTP –Auth method described in the previous section.

There is not much else, other than performing faultfinding with the logs, as described previously.

### **37.5 EMAIL USING ELASTIX POSTFIX SERVER AND ROUNDCUBE**

Now we come to the option that many want to use, but are unsure how to configure it. To be honest, for the user that has not had much to do with mail servers and DNS MX records, it might be a little bit daunting, but if you have setup the SMTP and that is all working, then you have done most of the job.

There are a few prerequisites for this implementation to be successful, these are

- 1. Ability to send email by SMTP (if you have managed to get one of the Email Direct or Email by SMTP Auth methods functioning, then you have it covered.
- 2. You need a real domain name and you need someone to host your DNS. Now many of the Domain registrars offer this service, and usually include enough

space for a Small Website in the package as well. There are other DNS hosting sites such as DynDNS and ZoneEdit, which also do the job.

3. With these Webhost/DNS providers, you need the ability to redirect the MX record to your Elastix box. Some have it in the form of a management page for your domain name, whereas some others can perform the change on request.

So lets start, go to the Elastix GUI and click on the Email tab and then the domain tab

elastix	System PBX Fax Email IM Reports Extras	About Elestix 1.0-15 Logout
Domains Accounts Relay	Webmail	0 0
Domain List	×	
Create Domain		
		🕾 Start 🖉 Previous (1 - 1 of 1) Next 🗅 End 🖄
Domain 2	Number of Ac	counts
penavation.homelins	2	
	·	If Start / Previous (1 - 1 of 1) Next / End M

You should see an empty domain list, which is correct. Click on the Create Domain button.

You will now see the next screen asking for the domain name.

System PBX Fax Email IM Reports Extras	About Elastix 1.0-16 Load
Domains Accounts Relay Webmail	0
New Domain Save Cancel	Required fi
Domain name:	

Now, you should have a domain name registered by now, and for this example, we are going to use elastixconnection.com as the domain name that we have just registered. Naturally you replace this with the one that you hopefully have registered.

So in the box, we are going to type **elastixconnection.com** and click on SAVE and the following screen will appear

	System PBX Fax Email IM Reports Extras	About Elastix 1.0-16 Logout
Domains Accounts Relay	Webmail	00
📃 Domain List		
Create Domain		
		✓ Start ✓ Previous (1 - 1 of 1) Next ト End 🕅
Domain	Number of Accounts	
elastixconnection.com	0	
		" Start / Previous (1 - 1 of 1) Next 🗅 End 🕅

This shows the domain name and the number of email accounts against this domain name. This mail server can handle multiple domains, but for this guide we will stick to one.

Click on the Accounts tab, and select the new domain you setup from the drop down box next to the word Domain as per the following screen.

System PBX Fax Email IM Reports Extras	About Elastix 1.0-16 Logou
omains Accounts Relay Webmail	0
Email Account List	
Demois Interference the second P Chang	
DUIIair: elastixconnection.com	
	Z // Start / Previous (0 - 0 of 0) Next > End
mail Address Account Name	Used Space

Now click on Create Account button, which will provide you with the following screen.

	System	PBX Fax Email IM Repo	rts Extras	About Elastix 1.0-16	Logou
omains Accounts F	telay Webmail				0
New Email Accou	int				
Save Cancel					* Required fie
Email Address: *	bob	@elastixconnection.com	Password: *	•••••	

Fill in the details, which are fairly straightforward. First is the email address that you want to setup, set a password, and then set a quota, which be aware, is in Kb, hence the 20000 that has been entered above will provide a quota of 20Mb.

Click on Save and the following screen will appear showing your email address setup.

System PBX	Fax Email IM Reports Extras	About Elastix 1.0-16 Loa	out
Domains Accounts Relay Webmail		•	•
Email Account List			
Domain: elastixconnection.com 💙 Show	Create Account		
		∥ Start / Previous (1 - 1 of 1) Next  > End	d M
Email Address	Account Name	Used Space	
bob@elastixconnection.com	bob.elastixconnection.com	0 Kb / 20000 Kb (0.00%)	
		🖓 Start 🧹 Previous (1 - 1 of 1) Next 🗅 End	d M

Naturally, if you have more users, then, keep setting them up, there is basically no limit except space and I suppose CPU cycles used. Take note of the account name. As an example the one in the screen above is **bob.elastixconnection.com**, which is a slightly different to what I have been used to over the years.

You will also need to determine the relay server for your network. In this case normally your local server and therefore we will accept the default that was provided. If you have other relay servers, this is where you defined them.

Click Apply changes to confirm as per the illustration below.

Declaration System PBX Fax Email IM Reports Extras	About Elastix 1.0-15	Logou
Networks which can RELAY		
These networks are allowed to send e-mail to external domains through the Elastin- using the SMTP protocol. You must insert one network per row and each network For example, to allow RELAW from the Elastic you must insert 127.0.0.1/32. We recommend leaping the 127.0.0.1/32 network in the configuration because some processes need it.	127.0.0.1/32	
	a na seconda de la constructione de la constructione de la construction de la constructio	

Now click on the Webmail Tab and the following screen will appear.

	System PBX Fax Email IM Reports Extras	About Elastix 1.0-16	Logout
Domains Accounts Relay	Webmail		0 0
	Username bob.elastixconnection.com Password ••••••• Login		

Enter in the login name that you previously setup in the email account list and the password and click on the Login button. The following screen will appear.

ains Accounts Rela	v Webmail			
BOUND		🐱 E-Mail 🛛 Address Boo	ok - 🎡 Personal Settings	🙆 Logout
WEBMAIL	🛃 🖉 😼 😼 🧭 🚍		(Q.	0
Folders Inbox Drafts Sant Sant Trash	Subject No messages found in this mailbox	Sender	Date 4	Size 🧬

You are now looking at your Webmail screen. Before you get too excited and start sending emails, we still have a little work to do.

Up in the right hand top corner of the Webmail screen, you will see the Personal Settings link, click on this and the following screen will appear.

Language	h (US)	
Time zone (GMT	+10:00) EAST/AEST: Guam, Vladivostok	/
Daylight savings		
Rows per page 40	· · ·	
Pretty dates	-	
Display HTML		
Compose HTML messages  😨		
Show preview pane		
Automatically save draft every!	5 minutes 💌	
Save		

Correct the Time Zone information for area and click save. Now click on the Identities Tab and the following screen will appear.

elastix	System PBX Fax Email IM Reports Extras	About Flastic 1.0-16	Logout
omains Accounts Rela	Webmail		0 0
	Preferences Folders Identities	😹 E-Mail 🗿 Address Book   旧 Personal Settings	Logout
Display name bob.elastiv.connection.com New identity	L-Mail Organization Reply-To bob-elastixconnection.com@lecalhost		

You will note that an email account is setup, <u>bob.elastixconnection.com@localhost</u> instead of <u>bob@elastixconnection.com</u>, - this is not what you expected. This is best referred to as the internal domain email address. You can use it to send an email internally to another user (as long as they are on the Elastix Postfix Mail server that you are registered on). What we need to do now, is associate another account, one with the correct details for external email use.

To do this click on the New Identity Button and the following screen will appear.

	System PBX Fax Email IM Reports Extras	About Elastix 1.0-16
	Preferences Folders Identities	🔯 E-Mail 🔄 Address Book 🌸 Personal Settings 🔇 Logov
Display name oob.elastixconnection.com New identity	E-Nail Organization Reply-To- bob-elasticconnection.com@localhost	
New identity Display name E-Mail Organization Reply-To BCC		
Signature		
Set default		

Now the details we will fill in here are:

Display Name:	Bob
Email:	bob@elastixconnection.com
Organisation:	ElastixConnection
Reply to:	bob@elastixconnection.com

Set Default: make sure this is ticked!!

So you should have similar to the following screen filled in.
	System PBX Fax Email IM Report	Extras	About Elastix	1.0-16	Lo
	Preferences Folders Identities	E-Mail	Address Book	旧 Personal Settings	😢 Logout
play name belastixconnection	E-Mail Organization /	орју-То			
ew identity					
w identity					
Display name	Bob				
E-Mall	bob@elastixconnectic				
E-Mail					
Organization	ElastixConnection				
Organization Reply-To	ElastixConnection bob@elastixconnectic				
Organization Reply-To Bcc	ElastixConnection				
Organization Reply-To BCC Signature	ElastixConnection				
Organization Reply-To Bcc Signature	ElastixConnection				

Now click save, and you will see the additional identity come up in your list, and as we have ticked it as the default, it is the one that the system will use.

elasti	System PBX Fax	Email IM Reports Extras		About Elastix	1.0-16	Logout
omains Accounts R	telay Webmail					0 0
	Proferences Folders	Identities	E-Mail	Address Book	🎡 Personal Settings	🗵 Lagout
Display name	E-Mail	Organization Reply-To				
Bob bob.elastixconnection.co	bob@elastixconnection.com om bob.elastixconnection.com@localhos	ElastixConnection bob@elastixconnection.com				
Bob bob.elastixconnection.co New identity	bob@elastixconnection.com om bob.elastixconnection.com@localhos	ElastixConnection bob@elastixconnection.com t				

In terms of setup of the Webmail system, that's it, unless you have more users to setup which will be setup identical to the one we have run through as an example.

At this point, you will not receive any mail, as all the Mail servers throughout the world, don't know anything about your mail server at the moment. This is where the DNS MX records come into play.

We now need to either set a MX record to point to your Elastix Server. As mentioned, a Control Panel of your Webhost facilities or DNS Registrar can sometimes do this.

As an example, some of the Webhosts running the latest CPanel software, now have MX record editing capabilities built in. Everyone's system is different, and definitely not within the scope of this document, but speaks to your Webhost or registrar and in almost all cases they will understand what you are after.

Correct DNS implementation dictates that DNS MX records must point to a Fully Qualified Domain Name (FQDN), not directly to an IP address. What this means is that you could point your MX record to a dynamic DNS name. Also I have not tested it, but it is understood that Dyndns and ZoneEdit amongst many others, have the ability to manage your MX records, for either a small fee or in the case of ZoneEdit, a few domain names for free. If you are using a dynamic IP address, I recommend doing a bit of research as I have not used these services for many years, but based on knowledge, it is all feasible.

One last thing, make sure that you have port 25 (SMTP) on your Internet router forwarded to your Elastix Mail server to allow the mail server to receive the mail.

# 38 CODEC (CODER DECODER)

When making a call over the net, the software (soft-phone) or hardware needs to use a codec so as to send/receive information in a certain format and convert it to what you hear. What this means to you and me is that codecs compress data, allow you to transmit the compressed data which is then uncompressed at the receiving end. If you compress something with G729 and you don't have that installed, then you wont be able to decompress it and vice versa.



g729 does not come preinstalled with Asterisk – you will need to purchase it.

Different codecs have different compression ratios resulting in different bandwidth requirements. Generally the higher the compression the more CPU power required to compress, and sometimes decompress.

In most cases this means:

- Higher compression = More CPU power = Lower Bandwidth
- Lower compression = Less CPU power = More Bandwidth

So the more bandwidth you have, the less need for a higher compression ratio and the less powerful the CPU that will be needed.

Most VoIP providers/hardware/licensed software will support G711 and G729 (However be sure to check this before purchasing hardware, or signing up with a VoIP provider!), with G711 requiring almost 3 times as much bandwidth, and providing excellent quality and the G729 providing quality similar to a good mobile call but using less bandwidth.

Below are the approximate bandwidth requirements for Several Common VoIP Compression Algorithms using SIP. They vary slightly depending on the protocol used.

Codec	Codec Bit Rate (Kbps)	Nominal Ethernet Bandwidth (Kbps)	Approx. MBytes usage per hour
G.711	64	87.2	39.24
G.729	8	31.2	14.04
G.723.1	6.4	21.9	9.86
GSM	13.2	28.7 approx	12.92 approx
iLBC	15.2	30.83 approx	13.87 approx
G.723.1	5.3	20.8	9.36
G.726	32	55.2	24.84
G.726	24	47.2	21.24
G.728	16	31.5	14.18

Codecs used by ELASTIX reside in the /usr/lib/asterisk/modules directory.

The following links will give you some indications of the bandwidth usage of the various codecs that you may be using or contemplating on using.

http://www.asteriskguru.com/tools/bandwidth\_calculator.php

http://www.cisco.com/en/US/tech/tk652/tk698/technologies\_tech\_note09186a0080094ae2.shtml http://www.voip-calculator.com/calculator/lipb/

http://www.connect802.com/voip\_bandwidth.php

http://www.newport-networks.com/pages/voip-bandwidth-calculator.html

There are others. All you have to do is Google for them ©

## 38.1 INSTALLING G729 AND G723 CODECS

If you have the licence for the g729 codec and found that the Digium version of g729 is a little problematic and would like to try the free version, following is the method of installing it as described by David Klaverstyn – thanks David.

## **DISCLAIMER:**

You might have to pay royalty fees to the G.729/723 patent holders for using their algorithm.

Before downloading the codec module you need to verify what your processor type is, as to get a compatible compiled codec.

From the Linux command type in:

# cat /proc/cpuinfo

Look specifically at the model name and the flags. The following is an example of the process that I am using.

Process:	0
vendor_id:	GenuineIntel
cpu family:	б
model:	8
model name:	Pentium III (Coppermine)
stepping:	3
cpu MHz:	648.110
cache size:	256 KB
fdiv_bug:	no
hlt_bug:	no
f00f_bug:	no
coma_bug:	no
fpu:	yes
fpu_exception:	yes
cpuid level:	2
wp:	yes
flags:	fpu vme de pse tsc msr pae mce cx8 mtrr pge mca
cmov pat pse36 mmx	fxsr <b>sse</b>
bogomips:	1297.41

From this information we can see that it is a Pentium III processor and it is using **sse**. This is the 2 bits of required information that we need. If you don't see **sse** in your flag then the file you need must contain the words **no-sse**.

Now go to <u>http://kvin.lv/pub/Linux/Asterisk/</u> and select wither Asterisk version 1.2 or 1.4 depending on the version you are using.

The files are named as codec\_g72x-???-pentiumx.so

So from the information above to obtain the G729 codec we need to get the file codec\_g729-gcc-pentium3.so and for the G723 codec we need to get the file g723-gcc-pentium3.so

The gcc represents the way the module was compiled. This is your most likely choice. To view the different type of compiled version look at this file <u>http://kvin.lv/pub/Linux/Asterisk/README</u>

Now that we know what file we need you can now download the appropriate file.

Change folders to /usr/lib/asterisk/modules

Elastix Without Tears

# cd /usr/lib/asterisk/modules

Download the required file(s)

## For the G729 codec

For Pentium III

```
# wget http://kvin.lv/pub/Linux/Asterisk/built-for-asterisk-
1.2/codec_g729-gcc-pentium3.so
```

## For Pentium IV

# wget http://kvin.lv/pub/Linux/Asterisk/built-for-asterisk-1.2/codec\_g729-gcc-pentium4.so

## For the G723 codec

### For Pentium III

# wget http://kvin.lv/pub/Linux/Asterisk/built-for-asterisk-1.2/codec\_g723-gcc-pentium3.so

### For Pentium IV

# http://kvin.lv/pub/Linux/Asterisk/built-for-asterisk-1.2/codec\_g723-gcc-pentium4.so

Rename the appropriate codecs to:

codec\_g729.so and codec\_g723.so

Restart asterisk and you are done.

If Asterisk does not start then you may need to try *icc* files instead of *gcc*.

# **39 PSTN TRUNK – SPECIAL HANDLING**

## **39.1 DIAL 0 FOR OUTSIDE LINE**

There will be times when a user would like to get an outside line something similar to a PABX by simply dialling a number e.g. 0 or 9.

I got around it by creating an entry in extensions\_custom.conf.

In my case I use 0 as the prefix to get an outside line.

I have also pre-recorded a voice prompt to instruct the caller what to do after the second dial tone.

### 39.1.1 Using Zaptel Card Method

The example above is based on systems that have Zaptel card (e.g. TDM400 or X100P etc.) installed. *Note:*\* *If you're using a PRI line, you need to enable 'overlapdial' in /etc/asterisk/zapata.conf for this to work.* 

## 39.1.2 Using SPA3K Method

Using SPA3K will be a little tricky. I am describing how I got around it (assuming that you have set up the SPA3K as described in this document), however there maybe other method.

1. In the SPA Line 1 Tab, I have included the following entry in the dialplan:

<0:#><:@gw0>)

2. The following is what my dialplan looks like in SPA3K

(\*\* | \*xx | xx. | <0: #><:@gw0>)

3. In the PSTN Line Tab, I have the following:

- VOIP-PSTN Gateway Enabled: Yes
- VOIP Caller Auth Method: None
- VOIP Caller Default DP: None

In made the following entry in my Asterisk entensions\_custom.conf

Elastix Without Tears

www.hellodigi.ir

```
exten => 0,1,Playback(start-dial-after-dialtone)
exten => 0,2,Dial(SIP/Telstra/{EXTEN}) ; Telstra being the name of the SPA3K trunk.
exten => 0,3,Hangup
```

Every time when I dial '0', the SPA3K will send it to Gateway 0, which is the PSTN line and present me with a second dial tone.

## 39.2 GIVING PRIORITY TO EMERGENCY NUMBERS (E.G. 000)

Emergencies, by definition should be given right of way. This applies to the right of using the telephone for emergency purposes above other uses.

Imagine what will happen if you want to use the PSTN line to call emergency 000 (in Australia) and someone else was hogging the line. The sensible way is for Asterisk to disconnect the line and allow you to make the emergency call instead.

The following method will provide you with the facility, however you need to do some editing to the extensions\_custom.conf file in /etc/asterisk.

In my case I have pre-recorded a voice prompt called 'calling-emergency' which says, "You are trying to call emergency assistance. If you have dial this in error, please hangup"

exten => 000,1,Background(calling-emergency); voice warning

exten => 000,2,Wait(1); give caller a chance to hang up

exten => 000,3,Dial(Zap/1/000,,j); If ZAP 1 is busy, jump to n+101

exten => 000,4,Hangup(); hang up after the call

exten => 000,104,Softhangup(Zap/1-1) ; hang up the line if being used

exten => 000,105,Goto(3); make call after disconnecting the other user

# 40 ADVANCED TRUNKS HANDLING

Some users would like to balance the trunks usage between VSPs and some prefer to cascade the usage amongst their VSPs.

Adrian Macarthur-King has been working on a more advanced trunk handling to allow the load balancing and cascading of trunk usage and came up with 2 methods to handle these requirements.

The 2 methods outlined below are similar in concept but not in usage. One of the methods may be suitable for you.

## 40.1 BALANCING TRUNK USAGE

This is a fairly good way of achieving trunk usage balancing, but it does take a little fiddling. This shouldn't take more than 10 minutes to set up, provided you know at least how to modify your extensions\_custom.conf and configure a call queue.

This method **doesn't** intelligently balance trunks based on number of calls per trunk (i.e > 300 = failover or similar), but it **does** balance outgoing calls evenly between as many trunks as you wish.

This has the same effect in many ways, but it won't stop you going over your FUP if you make more than you're supposed to for ALL of your trunks.

### Summary

You can use the "Automatic Call Distribution" aspect of asterisk queues to distribute calls to different outbound routes.

From the caller's perspective, this is seamless and appears exactly as a normal outbound call (except without correct progress tones - see Limitations)

I've used the "fewestcalls" ring strategy so that the trunks will be balanced based on the fewest completed calls. This should basically just alternate outbound calls back and forth between each trunk. Feel free to try other ring strategies such as roundrobin etc.

*Note\** This method balances calls between outbound routes, not trunks, which retains FreePBX flexibility with trunk order/failover.

#### Limitations

- Please note that this will disable outbound progress tones, if you don't know what that means, then it might not matter. Having said that, progress tones are very important to some, so this method might not be the best for them. See here for more info http://whirlpool.net.au/wiki/?tag=VOIP\_Progresstones
- 2) This will disable outbound route ordering from within freepbx. In fact, changing the route order in freepbx might break this, so set up your routes how you want them, and then don't change them after configuring this method. (If you do, it's not the end of the world, you just have to modify the outbound route names as you'll see later on).

#### Brief method description

We're going to create some scripts to hihack outbound calls from the standard outbound routes, save the number to dial into a variable, direct the call into a queue, which will then distribute outbound calls evenly between one of two or more "fake agents", which will then retrieve the number to be dialled from the variable, and pass the call on to the corresponding outbound route.

### **Detailed Steps.**

## These look long and complicated, but they're really not. Don't be dissuaded, give it a try!

I'll do as much of this as I can from FreePBX using the "Custom Destinations" and "Misc Applications" modules, but you **will** need to make conf file modifications. This guide assumes you want to balance two trunks, but it should work for as many as you want.

#### Step 1.

Ensure your FreePBX is relatively up to date. This should work with vanilla Elastix 1.0 just fine.

### Step 2.

- Create a new queue within freepbx. The queue will take our outbound calls, and distribute them to our fake agents (which will then redirect calls to our outbound routes)
- Assign an arbitrary queue number like '102' (You'll need this later)
- Assign some Static Agents with arbitrary numbers such as "1001" and "1002" (on seperate lines, you'll need these numbers later). Assign as many agents as you have outbound routes that you want to balance between. E.G Two static agents for two trunks.
- Check "Ringing Instead of MoH:" (This is the part that breaks progress tones, because the ringing is generated by the queue, but it's either that, or music on hold, so go figure)
- Set "Ring Strategy:" to fewest calls (or whatever you like if you know what you're doing)
- Leave everything else on defaults (unless you know what you're doing).

### Step 3.

Create the first half of the fake agents to represent our outbound routes.

You're going to need two different outbound-routes that you want to balance between. Each route can have the same dial prefixes, and this method will ignore the outbound route "order" and balance between two or more of the routes that you specify. If you know what you're doing, you can modify these scripts to allow for multiple route failover as well, but ignore that for the moment.

So firstly, create two outbound routes in FreePBX that we will use to alternate between for outbound calls.

Next, add the following to the end of your /etc/asterisk/extensions\_custom.co nf file.

[route1]
exten => \_.,1,Goto(OUTBOUNDROUTE\_1\_CONTEXT\_NAME\_HERE,\${DIALLEDNUMBER},1)

[route2]
exten => \_.,1,Goto(OUTBOUNDROUTE\_2\_CONTEXT\_NAME\_HERE,\${DIALLEDNUMBER},1)

Now before you add this, you're going to have to replace..

OUTBOUNDROUTE\_1\_CONTEXT\_NAME\_HERE and OUTBOUNDROUTE\_2\_CONTEXT\_NAME\_HERE

...with the contexts that represent your outbound routes. Now this is a little tricky, because there's no easy way to find out what these names are. The names seem to be a combination of the text "outrt-" with a three digit number representing their dial order + the name you gave them in FreePBX. The easiest way to find out the name, is browse to one of the outbound routes in FreePBX, and look at the URL in location bar of your browser, it should look something like the following.

https://10.0.0.100/admin/config.php?display=routing&extdisplay=002-gotalk1

If you take that 002-gotalk1 and add "outrt-" to the front of it, to become "outrt-002-gotalk1", then you have your route context name.

So my two are:

outrt-002-gotalk1 and outrt-002-gotalk2

So the end of my extensions\_custom.conf file would look like:

```
[route1]
exten => _.,1,Goto(outrt-002-gotalk1,${DIALLEDNUMBER},1)
[route2]
exten => _.,1,Goto(outrt-002-gotalk2,${DIALLEDNUMBER},1)
```

## Step 4.

Create the second half of the "fake agents" to represent our outbound routes.

- We have to create "Custom Destinations"s and "Misc Applications"s so that we can direct the calls to the custom script we've added.
- Create two custom destinations first.
- Have the route for the first one exactly "route1,\${EXTEN},1" where "route1" is the first name in the square brackets in Step 3.
- Create a description of say "Route1-Script"
- Have the route for the second one exactly "route2,\${EXTEN},1" where "route2" is the second name in the square brackets in Step 3
- Create a description of say "Route2-Script".

#### Step 5.

Create two Misc Applications.

#### First one:

Description: "WhateverYouLike" Feature code: One of the static agents you added in step 2 (e.g 1001) Destination: Custom Applications (Route1-Script)

#### Second one:

Description: "WhateverYouLike2" Feature code: The second of the static agents you added in step 2 (e.g 1002) Destination: Custom Applications (Route2-Script)

#### Step 6.

Add a custom outboud-allroutes context to your **/etc/asterisk/extensions\_custom.conf** to direct all outbound calls into our queue.

You can do this by copying and pasting the following at the end of your extensions\_custom.conf.

Make sure you CHANGE THE NUMBER in line 9 ("102") to be whatever queue number you assigned in Step 2.

```
[outbound-allroutes-custom]
exten => _.,1,Set(__DIALLEDNUMBER=${EXTEN})
exten => _.,n,Set(__BLKVM_OVERRIDE=BLKVM/${EXTEN}/${CHANNEL})
exten => _.,n,Set(__BLKVM_BASE=${EXTEN})
```

```
exten => _.,n,Set(DB(${BLKVM_OVERRIDE})=TRUE)
exten => _.,n,Set(_DIAL_OPTIONS=${DIAL_OPTIONS}M(auto-blkvm))
exten => _.,n,Set(__NODEST=${EXTEN})
```

## ;CHANGE THE NUMBER IN THE FOLLOWING LINE (102) to be whatever queue

## number you assigned earlier.

exten => \_.,n,Queue(102,tr,,)
exten => \_.,n,dbDel(\${BLKVM\_OVERRIDE})
exten => \_.,n,Set(\_\_NODEST=)
exten => \_.,n,Goto(app-blackhole,hangup,1)

# (There may be some weird spaces and newlines appearing so be careful copying and pasting this).

If you're interested, this last script assigns the number the person has dialled to a variable "Diallednumber" (so we can use it after the call comes out of the queue to pass the call on to where it was intended to go) and then places the call into a queue using the normal FreePBX method.

## 40.2 CASCADING TRUNK USAGE

This is another multiple trunk solution for "cascading" trunks instead of "balancing" them like in the previous solution.

This method allows you to specify a maximum number of calls, per type of call, per trunk, per billing month, and will automatically cascade through as many trunks as you like.

## Summary

This method will use a dial plan that keeps a count of outbound calls per type, per trunk, per month in the asterisk database.

In order to have this work with FreePBX, this method makes use of a custom trunk, and uses ordinary FreePBX trunks and routes.

Note that this method cascades FreePBX outbound routes, not trunks, so you can still have the flexibility of autofailover on particular routes.

This method is trickier to install than the load balancing because you have to mess around with the dial plans a little more to change the variables to suit your own set up.

## Limitations

1. Non retrospective. This method starts counting calls from 0 automatically from the moment it's installed, so if you start using this script halfway through the month, it won't know how many calls you've made so far.

2. If you lose the asterisk DB somehow, or delete the tree that the counts are stored in, then your call counts will start again and you'll lose where you're up to. This shouldn't happen though I don't think.

3. This method counts numbers of calls only, and NOT duration. This may be a problem for counting mobile minutes for example.

## Brief steps description

We're going to add several custom trunks in freepbx and point them at some contexts that we're going to add to the extensions\_custom.conf. Then we're going to add some

outbound routes to send calls to these custom trunks. Then we're going to modify these contexts to work how we want them to.

#### **Detailed Steps**

### Step 1

Work out how many different call types you want to count, and what prefix's (outbound route patterns) each calltype will need.

## E.G

mobiles 04XXXXXXXX

locals NXXXXXXX 02XXXXXXX

nationals 0[356789]XXXXXXXX

Step 2

In FreePBX, create as many custom trunks as you have call types.

In my case I created three custom trunks. I didn't change any settings when creating these three, except the "Custom Dial String", which I set to..

Local/\$OUTNUM\$@outbound-mobile-calls/n and Local/\$OUTNUM\$@outbound-local-calls/n and Local/\$OUTNUM\$@outbound-national-calls/n

## Step 3

This is now going to get a little complicated, so just bear with me. You're going to need a pile of outbound routes, and they're going to need to be in a particular order.

Firstly, we're going to need one outbound route for each "call type" we want to count. So if you have three calltypes, create three out-bound routes.

For each outbound route, give it a name, and set the dial patterns for each type to correspond with one of the call types we want to count, and set its trunk sequence as the corresponding custom trunk we added in step 1.

E.G Mobiles route Dial prefix: 04XXXXXXX Trunk Sequence: Local/\$OUTNUM\$@outbound-mobile-calls/n

Local Route Dial prefix: NXXXXXX 02XXXXXXX Trunk Sequence: Local/\$OUTNUM\$@outbound-local-calls/n

National route Dial prefix: 0[356789]XXXXXXX Trunk Sequence: Local/\$OUTNUM\$@outbound-national-calls/n

Make sure that these three outbound routes are at the **top** of the list in FreePBX. Reorder them using the arrows if you have to.

Secondly, we're going to need one outbound route for every actual trunk that we have. So if you have three actual trunks, create three more out-bound routes. Set each route with each trunk in its trunk sequence. Each route here will need a full list of dial patterns to **match ALL** call types (As if each route is the ONLY route you have. Setting them to "." will work to catch everything)

E.G

Route 1 -> Trunk sequence: Trunk 1 Route 2 -> Trunk Sequence: Trunk 2 Route 3 -> Trunk Sequence: Trunk 3

MAKE SURE that these routes are at the **bottom** of the list in FreePBX Reorder them using the arrows if you have to.

### Step 4

We're going to need to add the following to your /etc/asterisk/extensions\_custom.co nf Here is a link to the file <u>http://presstheinter.com/part1.conf</u>.

```
[outbound-count-calls]
; Don't change anything below here
exten => _.,1,Set(routeToUse=route1.000000)
exten => _.,2,Set(call-limit-ToUse=call-limit-route1.000000)
exten => _.,3,Set(yeardate=${STRFTIME(${MATH(${EPOCH}]_
          .,3,Set(yeardate=${STRFTIME(${MATH(${EPOCH}}-
${MATH(${billingoffset}*86400)})},GMT+10,%C%y%m)})
exten => _.,4,NoOp( Billing YearMonth = ${yeardate})
exten =>
_.,5,GotoIf(${DB_EXISTS(outboundcallcount/${outboundcallcounttype}-
${routeToUse}-${yeardate})}?${EXTEN},100:${EXTEN},110)
exten => _.,100,NoOp( Same Month / Trunk )
exten => _.,101,NoOp( ${outboundcallcounttype} is currently
${DB(outboundcallcount/${outboundcallcounttype}-${routeToUse}-
${yeardate})} )
exten =>
 .,102,Set(callcount=${DB(outboundcallcount/${outboundcallcounttype}}-
${routeToUse}-${yeardate})})
exten => _.,103,GotoIf($[${callcount} < ${${call-limit-</pre>
ToUse}}]?${EXTEN},107:${EXTEN},104)
exten => _.,104,Set(routeToUse=${routeToUse:0:5}${MATH(${routeToUse:-
8:8}+1)})
exten => _.,105,Set(call-limit-ToUse=${call-limit-
ToUse:0:16}${MATH(${call-limit-ToUse:-8:8}+1)})
exten => _.,106,Goto(5)
exten => _.,107,Set(DB(outboundcallcount/${outboundcallcounttype}-
${routeToUse}-${yeardate})=${MATH(${callcount}+1)})
exten => _.,108,NoOp( ${outboundcallcounttype} count is now
${MATH(${callcount}+1)} )
exten => _.,109,Goto(${${routeToUse}},${EXTEN},1)
exten => _.,110,NoOp( New Month / Trunk )
exten => _.,111,Set(DB(outboundcallcount/${outboundcallcounttype}-
${routeToUse}-${yeardate})=1)
exten => _.,112,Goto(${${routeToUse}},${EXTEN},1)
```

## Step 5

Now we have to add more to the /etc/asterisk/extensions\_custom.co nf file, but this time, we will have to change the text.

Basically we need to add one context for each of our call types that we want to count. Each context will need to look like the following but will need several parts changed. I've bolded the sections you need to change. One particular part is the "billingoffset=1" part. You need to replace this number 1 with the number of the day of the month that your billing cycle starts.

The hard part is that you're going to have to replace...

OUTBOUNDROUTE\_1\_CONTEXT\_NAME\_HERE **and** OUTBOUNDROUTE\_2\_CONTEXT\_NAME\_HERE

.. (and more if you want) with the contexts that represent the second set of outbound routes you created in Step 3. Now this is a little tricky, because there's no easy way to find out what these names are. The names seem to be a combination of the text "outrt-" with a three digit number representing their dial order + the name you gave them in FreePBX. The easiest way to find out the name, is browse to one of the outbound routes in FreePBX, and look at the URL in location bar of your browser. it should look something like the following:

#### https://10.0.0.100/admin/config.php?display=routing&extdisplay=002-gotalk1

If you take that 002-gotalk1 and add "outrt-" to the front of it, to become "outrt-002-gotalk1", then you have your route context name.

So my two are:

outrt-002-gotalk1 and outrt-002-gotalk2

So, make three copies of the following context. Modify each one to have correct settings and add them to the end of your **extensions\_custom.conf**.

Here is a link to the file http://presstheinter.com/part2.conf

```
[outbound-mobile-calls]
; Set the name of the calltype here. One word, no spaces.
exten => _.,1,Set(__outboundcallcounttype=mobile)
exten => _.,n,NoOp( Calltype = ${outboundcallcounttype} )
; Set billing offset here
exten => _.,n,Set(__billingoffset=0)
; Set each route name and the call limit for this route for this call
type. They are in the order in which they will cascade.
exten => _.,n,Set(__route1.000000=OUTBOUNDROUTE_1_CONTEXT_NAME_HERE)
exten => _.,n,Set(__call-limit-route1.000000=100)
exten => _.,n,Set(__route2.00000=OUTBOUNDROUTE_2_CONTEXT_NAME_HERE)
exten => _.,n,Set(__call-limit-route2.000000=100)
exten => _.,n,Set(__route3.00000=OUTBOUNDROUTE_3_CONTEXT_NAME_HERE)
exten => _.,n,Set(__call-limit-route3.000000=100)
exten => _.,n,Set(__route4.000000=UNUSED)
exten => _.,n,Set(__call-limit-route4.000000=100)
exten => _.,n,Set(__route5.000000=UNUSED)
exten => _.,n,Set(__call-limit-route5.000000=100)
;You can create as many more trunks as you like by following the above
formula
exten => _.,n,Goto(outbound-count-calls,${EXTEN},1)
```

So my three entries looked like the following (<u>http://presstheinter.com/part3.conf</u>)

```
[outbound-mobile-calls]
; Set the name of the calltype here. One word, no spaces.
exten => _.,1,Set(__outboundcallcounttype=mobile)
exten => _.,n,NoOp( Calltype = ${outboundcallcounttype} )
; Set billing offset here
exten => _.,n,Set(__billingoffset=0)
; Set each route name and the call limit for this route for this call
type. They are in the order in which they will cascade.
exten => _.,n,Set(__route1.000000=outrt-004-gotalk1)
exten => _.,n,Set(__call-limit-route1.000000=100)
exten => _.,n,Set(__route2.000000=outrt-005-gotalk2)
exten => _.,n,Set(__call-limit-route2.000000=100)
exten => _.,n,Set(__route3.000000=outrt-006-gotalk3)
exten => _.,n,Set(__call-limit-route3.00000=100)
exten => _.,n,Set(__route4.000000=UNUSED)
exten => _.,n,Set(__call-limit-route4.000000=100)
exten => _.,n,Set(__route5.000000=UNUSED)
exten => _.,n,Set(__call-limit-route5.000000=100)
;You can create as many more trunks as you like by following the above
formula
exten => _.,n,Goto(outbound-count-calls,${EXTEN},1)
[outbound-local-calls]
; Set the name of the calltype here. One word, no spaces.
exten => _.,1,Set(__outboundcallcounttype=local)
exten => _.,n,NoOp( Calltype = ${outboundcallcounttype} )
; Set billing offset here
exten => _.,n,Set(__billingoffset=0)
; Set each route name and the call limit for this route for this call
type. They are in the order in which they will cascade.
exten => _.,n,Set(__route1.000000=outrt-004-gotalk1)
exten => _.,n,Set(__call-limit-route1.000000=300)
exten => _.,n,Set(__route2.000000=outrt-005-gotalk2)
exten => _.,n,Set(__call-limit-route2.00000=300)
exten => _.,n,Set(__route3.000000=outrt-006-gotalk3)
exten => _.,n,Set(__call-limit-route3.00000=300)
exten => _.,n,Set(__route4.000000=UNUSED)
exten => _.,n,Set(__call-limit-route4.000000=300)
exten => _.,n,Set(__route5.000000=UNUSED)
exten => _.,n,Set(__call-limit-route5.000000=300)
;You can create as many more trunks as you like by following the above
formula
exten => _.,n,Goto(outbound-count-calls,${EXTEN},1)
[outbound-national-calls]
; Set the name of the calltype here. One word, no spaces.
exten => _.,1,Set(__outboundcallcounttype=national)
```

www.hellodigi.ir

```
exten => _.,n,NoOp( Calltype = ${outboundcallcounttype} )
; Set billing offset here
exten => _.,n,Set(__billingoffset=0)
; Set each route name and the call limit for this route for this call
type. They are in the order in which they will cascade.
exten => _.,n,Set(__route1.000000=outrt-004-gotalk1)
exten => _.,n,Set(__call-limit-route1.000000=300)
exten => _.,n,Set(__route2.000000=outrt-005-gotalk2)
exten => _.,n,Set(__call-limit-route2.00000=300)
exten => _.,n,Set(__route3.000000=outrt-006-gotalk3)
exten => _.,n,Set(__call-limit-route3.00000=300)
exten => _.,n,Set(__route4.000000=UNUSED)
exten => _.,n,Set(__call-limit-route4.000000=300)
exten => _.,n,Set(__route5.000000=UNUSED)
exten => _.,n,Set(__call-limit-route5.000000=300)
;You can create as many more trunks as you like by following the above
formula
```

exten => \_.,n,Goto(outbound-count-calls,\${EXTEN},1)

#### Step 6.

After you make each call, you can watch the count go up. I set my call-limits to 5 just to test

After 1 call.

elastix\*CLI> database show /outboundcallcount/mobile-route1.0 00000-200805 : 1

After 6 calls. elastix\*CLI> database show /outboundcallcount/mobile-route1.0 00000-200805 : 5.000000 /outboundcallcount/mobile-route2.0 00000-200805 : 1

I guess this will fill your database up with junk after awhile, so if you really want, on the first day of your billing cycle, you could run

elastix\*CLI> database deltree outboundcallcount, to clear this out.

I would include this in the code, but some people might be interested in their statistics over time, and it really won't use up that much space, so I've left it in.

# 41 PORT FORWARDING AND QOS – ROUTERS

## 41.1 PORT FORWARDING

While most domestic routers using **UPnP** do not require port forwarding, some routers do. Even if your router does not require port forwarding for ELASTIX to work, do yourself a favour and forward a few of the relevant ports to your Asterisk IP address anyway – it doesn't hurt.

## **IAX Port:** 4569 (UDP)

**Sip ports:** 5004 - 5082 (UDP and TCP), more if you want to, although you only require 5060.

**RTP ports:** 10000 – 10500 (UDP) – this should be enough although in the conf file it was defined as 10000-20000.

I am using the LinkSys WRT54G wireless router. In the example below, I have forwarded all the ports mentioned above to my asterisk box IP, which is 192.168.1.101

Applications										
& Gaming	Setup	Wireless	Securi	A Ay Res	ccess A trictions 8	Applications & Gaming	Administration Status			
	Port Range F	orward	Port Trigg	gering	DMZ   (	QoS				
Port Range Forward							Port Range Forwarding :			
	Applications & Gaming       Setup       Wireless       Security       Access Restrictions       Applications         Port Range Forward       Port Triggering       DMZ       Gos         Port Range Forward       Port Range       Port Range       Enable         Image Forward       Image Forwar									
	Application	Start	End	Enable	for it to function correctly. Examples of these applications					
	IAX	4569 to	4569	Both 🔽	192.168.1. 101		include servers and certain online games. When a request			
	AMP*	8080 to	8080	Both 💌	192.168.1.101		for a certain port comes in from the Internet, the router will route			
	Azareus	50001 to	50010	Both 💌	192.168.1. 3	<b>~</b>	specify. Due to security			
	SIP	5060 to	5065	Both 💌	192.168.1.101		concerns, you may want to limit port forwarding to only those ports you are using, and			
	RTP	10000 to	11000	UDP 🗸	192.168.1. 101	. 🗸	uncheck the Enable checkbox			

These should be enough for starters.

## 41.2 QOS – ROUTERS

Some routers have QOS facility while some routers haven't. It is highly recommended that you enable this function if your router supports it.

There have been a number of discussions on the effectiveness of QOS on broadband utilising the public Internet circuit. It is beyond this guide to document the effectiveness or otherwise of enabling QOS on your router. For a start, there are just too many routers there for me to give any meaningful examples.

I use the Linksys<sup>™</sup> WRT54G<sup>®</sup>. Taking the easy way out, I simply restrict the PCs (see below) that more often than not will be used as a P2P client or chatting on the MSN chat channels.

I restrict the PCs according to their MAC addresses.

Applications									
& Gaming	Setup V	Vireless	Security	Access Restrictions	Applications & Gaming	Administration	Status		
	Port Range Forw	vard	Port Triggering	I DMZ I	QoS				
Wired QoS	Upstream Bandv	⊙e vidth Aut	Two types of Qu features are ava QOS which contr	ality of Service ilable, Wired rols devices					
Device Priority	Device name Asteris	Prior Highes	<b>ity</b> st 💙 00 : 0.	MAC Address	30	Ethernet cable, a QoS, which cont are wirelessly co router.	nd Wireless rol devices that onnected to the		
	Device name Family	Low	✓ 00 : 0:	7:3D:57:9C:	B4	Wired QoS			
	Device name	Low	✓ 00 : 0	D: 00: 00: 00:	00	Device Priority specify priority for from a device on	: You may or all traffic your network		

In the application priority section (see below), I have defined the various ports that I want priority to be accorded the highest priority. In this case I have defined the port ranges related to my VOIP requirement.

Application Priority		🗹 Optimize Gaming	Applications	Wireless QoS is also referred to
		Priority	Specific Port #	as Wi-Fi MultiMedia <sup>™</sup>
	Application Name SIP	Highest 💌	5060-5065	(WMM) by the WI-Fi Alliance <sup>TM</sup> . Select Enable to utilize VMM if you are using other wireless
	Application Name RTP	Highest 💌	10000-11000	devices that are also WMM certified.
	Application Name RTP	Highest 💌	4569	No Acknowledgement : Enable this option if you would
	Application Name	Low 🚩	0	like to disable acknowledgement. If this option

My recommendation is, if your router has the function, use it. At the very least, it will prioritise traffic within your own LAN especially when you have sons and daughters (for that matter you too can be the culprit) who are always on the P2P, leeching files and music.



*This may not be the best solution. If anyone has a better solution for the Linksys WRT54G, please let me know.* 

# 42 UPGRADING YOUR SYSTEM

Mostly everyone would like to have the latest and the greatest and if you are like most, chances are that you would certainly want to update your ELASTIX and/or freePBX as soon as there is a new update.

You repository is repo.elastix.org

## 42.1 UPGRADING ELASTIX

To upgrade Elastix, at the command prompt, type the following;

yum -y update **or** yum update elastix

In most cases, it may be prudent to update the whole system including Centos, where other dependencies will also be addressed. This will also upgrade freePBX.

The approach I took is to first update yum itself.

yum update yum

This will only take a few minutes and it gives the advantage of the process looking for the fastest mirror which will reduce your online update time by a factor of up to 80%.

After having updated yum, you may then issue the full update command.

yum update

This method cuts down my online update process from about 5 hours to about 1 hour.

When you execute the command, you will see a screen similar to the one illustrated below.

🛃 root@elastix1:~						×
php-gd i386	5.1.6-15.el5 base			111	k	~
php-imap i386	5.1.6-15.el5 base			51	k	
php-mbstring i386	5.1.6-15.el5 base			968	k	
php-mysql i386	5.1.6-15.el5 base			83	k	
php-pdo i386	5.1.6-15.el5 base			61	k	
Transaction Summary						
					===	
Install 1 Package(s)						
Update 9 Package(s)						
Remove O Package(s)						
Total download size: 9.5 M						
Is this ok [y/N]: y						
Downloading Packages:						
(1/10): php-5.1.6-15.el5. 100%	========================	1.2	MB O	0:04		
(2/10): php-pdo-5.1.6-15. 100%	========================	61	kB O	0:01		
(3/10): php-imap-5.1.6-15 100%	==============================	51	kB O	0:01		
(4/10): php-mysql-5.1.6-1 100%	==============================	83	kB O	0:01		
(5/10): php-cli-5.1.6-15. 100%	========================	2.3	MB O	0:08		
(6/10): php-common-5.1.6- 100%	=============================	140	kB O	0:01		_
(7/10): php-xml-5.1.6-15. 100%	=============================	93	kB O	0:01		-
(8/10): php-gd-5.1.6-15.e 100%	==============================	111	kB O	0:01		
(9/10): elastix-1.0-17.no 27%	=====	1.2	MB O	1:51	ETA	~

At the end of it, it will tell you that your update is completed.

amportal restart

## 42.2 UPGRADING FREEPBX

It is suggested not to upgrade freePBX to a new Stable until the new version has been fully integrated into Elastix as experience dictates that it may break something. That does not mean that you shouldn't upgrade but do it with the caution in mind.

There are 2 ways to upgrade freePBX as explained at the aussievoip website here http://www.aussievoip.com/wiki/freePBX

Either one will be fine but I would like to suggest that you take the all-in-one version with modules and be done with it. Afterall, this is the method recommended if one is on a limited-bandwidth connection, or the FreePBX machine doesn't have easy Internet access.

To conduct this upgrade, you will need to log on to the Elastix box as root.

At the command prompt, do the following;

```
cd /usr/src/
wget http://mirror.freepbx.org/freepbx-2.4.0.tar.gz(or whatever the new
version is)
tar zxvf freepbx-2.4.0.tar.gz
cd freepbx-2.4.0
./install_amp
```

Once this is done, freePBX is now upgraded to version 2.4.0 and you can start managing and/or upgrading the freePBX modules.

# 43 TOOLS

The following are tools that I use regularly to maintain my Asterisk systems.

## 43.1 WEBMIN

Webmin is an invaluable web based tool for managing a Linux box. Webmin make it easy to configure application like SMTP mail, editing files etc.



Webmin is not installed as default in Elastix but user can download and install it manually.

Log in to the Elastix box as root and do the following at the command prompt:

wget http://internap.dl.sourceforge.net/sourceforge/webadmin/webmin-1.410-1.noarch.rpm

rpm --install webmin-1.410-1.noarch.rpm

I have found the above method is straightforward and simple. However there are some users who found that following an alternative method is simpler. If that is the case, the alternative installation method can be found here:

http://www.terrasoftsolutions.com/support/solutions/ydl\_general/webmin.shtml

Once installed, you may connect to Webmin remotely through your browser using the following address *https://<YourAsterisk\_IPAddress>:10000*. E.g.

192.168.0.101:10000

If you would like to avoid clashing the port with your RTP port which is 10000 – 20000, you may edit the file */etc/webmin/miniserv.conf* and change the port value to something else, such as 9000.

Locate the entry where it says:

Port=10000

And change this to;

Port=9000

You can use https://192.168.0.101:9000 to connect to Webmin instead.

The alternative is to edit the */etc/asterisk/rtp.conf* file like the example below and you can keep *https://192.168.0.101:10000* 

This will break PA1688 based phones. It's better to change your webmin.conf file

[general]

; RTP start and RTP end configure start and end addresses

rtpstart=10000	< change this to	10002(totally unnecessary ©)	)
rtpend=20000			

# 43.2 PUTTY

PuTTY is a free implementation of Telnet and SSH for Win32 and Unix platforms, along with an xterm terminal emulator.

🞇 PuTTY Configura	tion	$\mathbf{X}$
Category:		
🖃 Session	^	Basic options for your PuTTY session
Logging		Specify your connection by host name or IP address
Erminal		Host Name (or IP address) Port
Bell		22
Features		Protocol:
😑 Window		🔘 Raw 🔵 Telnet 🔘 Rlogin 💿 SSH
Appearance		CLoad, save or delete a stored session
- Benaviour Translation		Saved Sessions
Selection	=	
Colours		Default Settings
Connection		
Data		Save
- Proxy Telnet		Delete
SSH		
Kex		Close window on exit:
Auth		🔿 Always 🛛 Never 💿 Only on clean exit
Tunnels	~	
About		Open Cancel

It is written and maintained primarily by <u>Simon Tatham</u> and can be downloaded from the following link.<u>http://www.putty.nl/download.html</u>

# 43.3 WINSCP

WinSCP is an open source freeware SFTP client for Windows using SSH. Legacy SCP protocol is also supported. Its main function is safe copying of files between a local and a remote computer.

g root - root@192.168.1.10	0 - Wi	nSCP										
Local Mark Files Commands S	ession (	Options Remote H	elp									
🔹 🕅 • 🖙 📽 🎨	HOIT 🛃		<b>V</b>	Defau	ult		-   🐼 -					
ss Istra	<b>*</b> 🗿											
👄 C: Local Disk 🛛 🖌 🔶 🚽		📾 📾 🚮 🚱 🛛	🔄 දිසු				🚞 root	✓ [ 4 + 4 + ]	🛍 🙆 🚮 🛃	🗀 te		
21							/root					
Name 🔶 Ext	Size	Туре	Changed	Attr		^	Name - Ext	Size	Changed	Rights	Owner	
97136d82fb20b78da31		File Folder	18/11/200				<b>b</b>		12/03/2007 5:1	rwxr-xr-x	root	
BJPrinter		File Folder	18/11/200	h			asterisk_history	49	12/03/2007 5:1	rw	root	
Documents and Settings		File Folder	13/12/200				bash_history	2,550	12/03/2007 5:1	rw	root	
drivers		File Folder	18/11/200				🛅 .bash_logout	24	22/02/2005 3:2	rw-rr	root	
ed37be1d9687d5ee082		File Folder	18/11/200				bash_profile	191	22/02/2005 3:2	rw-rr	root	
GSpot252b01		File Folder	18/11/200				.bashrc	176	22/02/2005 3:2	rw-rr	root	
LAME		File Folder	18/11/200				.cshrc	100	22/02/2005 3:2	rw-rr	root	
MSOCache		File Folder	18/11/200	hr			tcshrc	102	22/02/2005 3:2	rw-rr	root	
My PSP Files		File Folder	11/03/200				🖬 anaconda-ks.cfg	807	4/12/2006 6:37	rw-rr	root	
Program Files		File Folder	13/03/200	r			dead.letter	3	6/12/2006 11:0	rw	root	
Putty		File Folder	18/11/200				🗐 install.log	14,745	4/12/2006 6:37	rw-rr	root	
QUICKENW		File Folder	12/12/200				🖬 install.log.syslog	3,760	4/12/2006 6:37	rw-rr	root	
RECYCLER		File Folder	18/11/200	sh			🗐 install asterisk.log	714,124	4/12/2006 6:43	rw-rr	root	
System Volume Informa		File Folder	27/11/200	sh			mbox	68.083	6/12/2006 11:0	rw	root	
The		File Folder	8/12/2006									
WINXP		File Folder	13/03/200			_						
AUTOEXEC.BAT	0	MS-DOS Batch File	18/11/200	a								
Soot.ini	209	Configuration S	18/11/200	ash								
CONFIG.SYS	0	System file	18/11/200	a								
l csh.log	126	Text Document	18/11/200	a								
Eilteri og.log	80	Text Document	18/11/200	a		~						
P of 2 559 MP in 0 of 29	00	, en e e danora					0 P of 705 KP in 0 -613					
D UI 2,003 MD III 0 01 28		-										
🖗 F2 Rename 🔡 F4 Edit. 📑 F1	Copy i	🚡 F6 Move 💣 F7	Create Directory	X F8 D	elete 🗎	F91	Properties  👖 F10 Qu	it				

It can be downloaded from the following link.

http://winscp.net/eng/index.php

## 43.4 NANO

A simple editor called NANO, which for the novice Linux user is far simpler to understand and control than the standard VI that comes with the Elastix Distribution.

Since Nano is not included in the Elastix distribution, it need to be downloaded and install manually. To do that you must log in to Elastic as user root.

So once you are logged into your Elastix Server either at the console or via SSH, you need to issue a command *yum install nano* similar to the following screen.



After a few lines flick by (this may take up to a minute or so), it will then ask you if it is ok to install **nano** which you will reply with a **Y**. Once completed you will then have Nano on your system.

## 43.5 PHPMYADMIN

PhpMyAdmin is a web interface for managing your MySQL server. Unfortunately it is not included in the Elastix distribution. If you want to use this tool, you will need to download it from <u>The phpMyAdmin</u> Project for their latest version.

After downloading the file, do the following step to install it.

- 1. Log in to your Elastix server, either at the console or using putty, as root.
- 2. Using nano, create a dbadmin directory in /var/www/html/modules

```
cd /var/www/html/modules mkdir dbadmin
```

- 3. After unpacking the phpMyAdmin file that you downloaded, copy the entire content to /var/www/html/modules/dbadmin
- 4. Create a text file called config.inc.php in /var/www/html/modules/dbadmin

nano config.inc.php

5. Populate the text file with the following:

```
<?php
/* Servers configuration */
$i = 0;
/* Server localhost (config:root) [1] */
$i++;
$cfg['Servers'][$i]['host'] = 'localhost';
$cfg['Servers'][$i]['extension'] = 'mysql';
$cfg['Servers'][$i]['connect_type'] = 'tcp';
$cfg['Servers'][$i]['compress'] = false;
$cfg['Servers'][$i]['auth_type'] = 'http';
$cfg['Servers'][$i]['user'] = '';
$cfg['Servers'][$i]['password'] = '';
/* End of servers configuration */
?>
```

Once this is done, you may log in to phpMyAdmin with your browser using the following address: *http://<your.elastix.address>/modules/dbadmin* (eg: *http://192.168.1.101/modules/dbadmin*).

You will be asked to enter the username and password. Use *root* as the username and *eLaStlx.2007* as the password and you can now start using the phpMyAdmin.

phpMuAdmin () () () () a steinskoddu () a normikoddu () information, Schema (16) information, Schema (16) Please select a database	elastix.pcnovation.local Server version: 50.22 Protocol version: 10 MySOL conscription (local via TCP/IP) Vers: root@localizati MySOL conscription collation: urt_unicode_ci version Collation version: collation: urt_unicode_ci version Show MySOL constraine information Show MySOL system versible: Collations Storage Engines Character Sets and Collations Character Sets and Collations Character Sets and Collations Character Sets and Collations Protoges Protoges Protoges Character Sets and Collations Character Sets and Collations Sets and Collations Character Sets an	<pre>byByByByByByByByByByByByByByByByByByByB</pre>
--	---	--

# 44 GET UNDER THE BONNET

1	Ð
	=

Config files manual tweaks in this guide are meant for advanced users. Unless you have a compelling reason to do so, users are advised against any manual config file tweaks.

Just as you think that all is OK, you realized something else requires attention. This is true with Elastix as well.

To do this we may need to edit some configuration files (.conf) that reside both in the /etc/asterisk directory and /etc directories. Configuration files in the /etc/asterisk are generally editable through File editor.

To start editing the **.conf** files we need to select the PBX tab and select the Tool option followed by File Editor selection from the menu as illustrated below.

	m PBX Fax Email IM Reports	Extras	About Elastix 1.0-12	Logout
PBX Configuration   Flash Operator Pane	Voicemails   Monitoring   Echo Cancell	er Endpoint Configuration Conferenc	e Extensions Batch Tools	0 0
Asterisk-Cli (a) As File Editor Comman	terisk-Cli Id Execute		2	

You may need to navigate through the pages to find the file you wish to edit from the resulting screen.

		🙀 Start 🔌 Previous (101 - 115 of 115) Next 🗅 End 🕅
File List		
telcordia-1.adsi		
udptl.conf		
unicall.conf		
users.conf		1 I
<u>vm_email.inc</u>		
vm_general.inc		
voicemail.conf		
voicemail.conf.template		
vpb.conf		
zapata-channels.conf		
zapata-channels.conf.bak		
zapata.conf		
zapata.conf.template		
zapata_additional.conf		
zoip.conf		
		₩ Start ┥ Previous (101 - 115 of 115) Next 🗅 End 🕅
	Elastix is licensed under GPL by PaloSanto Solutions. 2006, 2007.	

Eg. The zapata.conf can be found on the last page.

Select the file you wish to edit e.g. zapata.conf (marked with red arrow) and we will see a new screen below which can be edited manually.

	ystem PBX Fax Email IM Reports Extras	About Elastix 0.9.2-4	Logou
Configuration Asterisk-Cli Flas	h Operator Panel Voicemails Monitoring <u>File Editor</u> Echo Canceller		0
	zapata.conf		
	J Used for distinctive ring support for x100p. J You can see the dring% patterns is to set any or dring%context fields J and they will be printed on the console when an call comes in. J dring1=95,0,0 dring1=235,95,0 dring2=235,95,0 jdring2=235,95,0 jdring2=235,95,0 infing2context=internal2 J If no pattern is matched here is where we go. protext=drefault product => 1	ne of the Annound	
	<pre>#include zapata_channels.conf #include zapata_additional.conf </pre>	M N	
	2	- 1	

Once done with your editing, click Save and then click Back as illustrated above.

## 44.1 EDITING THE .CONF FILES

A number of **.conf** files may require editing to get asterisk to work, depending on the individual requirements. In this guide, as a base, we will be looking at **sip.conf**, **sip\_custom.conf**, **iax\_custom.conf** (just in case you want IAX, but not really necessary if you are just running SIP), **indications.conf**, **enum.conf**, **extensions\_custom.conf** and **modprobe.conf** (which lives in the **/etc** directory and may need to be edited if you have a FXO card e.g. X100P installed. We will look at them later on under Interfacing Asterisk to PSTN).



Every time you update freePBX, there is a good chance that changes or additions you have made in **sip.conf** will be overwritten. To avoid this, all additions I wish to make in **sip.conf** are made in **sip\_custom.conf** 

## 44.1.1 sip.conf

Lets make a slight change in sip.conf. We believe, the position of the **#include** *sip\_nat.conf* should be placed more towards the beginning of the file. Instead of deleting the line from the original position at the bottom of *sip.conf*, we simply commented it out and added it towards the beginning of *sip.conf* but *underneath the bindport and the bindaddr* as shown below.

```
; 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]
#include sip_general_additional.conf
bindport = 5060 ; Port to bind to (SIP is 5060)
bindaddr = 0.0.0.0 ; Address to bind to (all addresses on machine)
#include sip_nat.conf <--- insert this line here
disallow=all
allow=ulaw
```

```
allow=alaw
; If you need to answer unauthenticated calls, you should change this
; next line to 'from-trunk', rather than 'from-sip-external'
; You'll know this is happening if when you call in you get a message
; saying "The number you have dialed is not in service. Please check the
; number and try again."
context = from-sip-external ; Send unknown SIP callers to this context
callerid = Unknown
tos=0x68
; Reported as required for Asterisk 1.4
notifyringing=yes
notifyhold=yes
limitonpeers=yes
; enable and force the sip jitterbuffer. If these settings are desired
; they should be set in the sip_general_custom.conf file as this file
; will get overwritten during reloads and upgrades.
; jbenable=yes
; jbforce=yes
; #, in this configuration file, is NOT A COMMENT. This is exactly
; how it should be.
#include sip_general_custom.conf
#include sip_registrations_custom.conf
#include sip_registrations.conf
; #include sip_nat.conf <--- comment this line out or delete it altogether
#include sip_custom.conf
#include sip_additional.conf
```

## 44.1.2 sip\_custom.conf

This file may have to be created, as it may not exist.

```
language=au
videosupport=yes
allow=g729
allow=g723
allow=h261
allow=h263
allow=h263p
defaultexpirey=600 ; **Engin users: include this if necessary
maxexpirey=3600 ; **Engin & BBP Global users: include this if necessary
rtptimeout=60
rtpholdtimeout=120
```



Every time you update freePBX, there is a good chance that changes or additions you have made in **iax.conf** will be overwritten. To avoid this, all additions I wish to make in **iax.conf** are made in **iax\_custom.conf** 

## 44.1.3 iax\_custom.conf

This file may have to be created, as it may not exist.

```
; additions to iax.conf that would not get deleted if you upgrade freepbx
```

```
;externip = <your fixed external IP> or (one or the other NOT both)
externhost = <your DynDNS name>
localnet = 192.168.1.0/255.255.0
```

```
language=au
delayreject = yes
jitterbuffer = yes
mailboxdetail = yes
allow=g729
allow=ilbc
allow=g723
```

## 44.1.4 indications.conf

In the general section ensure the following exist:

```
[general]
country = au ; (The default is country = us, therefore replace "us" with "au")
```

## 44.1.5 enum.conf

```
[general]
;
; The search list for domains may be customized.
; Domains are searched in the order they are listed here.
;
search => el64.arpa
search => el64.org
```

### 44.1.6 extensions\_custom.conf

If you have a ZAP trunk and you want to get an external line and be presented with a dial tone first before you dial the external line, you may add the following codes in the extensions\_custom.conf file.

The example below assumes that your ZAP is on channel 1.

```
exten => 0,1,Dial(ZAP/1/{EXTEN})
exten => 0,2,Hangup
```

This will allow you to dial 0 and you will get a second dial tone from PSTN. Following which you can dial a PSTN number that you wish to call.



If you're using a PRI line, you need to enable 'overlapdial' in /etc/asterisk/zapata.conf for this to work.

## 44.2 CREATE MISSING DIRECTORIES

You may need to create a few missing directories but before you do that please make doubly sure that the directories are actually missing. They are:

/var/www/html/recordings /var/www/html/\_asterisk

You can either use WinSCP to create these directories or use Nano.

If you use Nano, do the following:

cd /var/www/html mkdir recordings cd /var/www/html mkdir \_asterisk

When you start Elastix or when you do amportal restart and if you see the following errors means that you will need to create those directories.

chown: cannot access `/var/www/html/recordings': No such file or directory chmod: cannot access `/var/www/html/recordings': No such file or directory chown: cannot access `/var/www/html/\_asterisk': No such file or directory.

# **45 TROUBLE SHOOTING**

With any installation, you will find some unexpected problems. Some are inherent issues due to bugs or undocumented features, while others are unintentional mistakes.

We will endeavour to list troubleshooting tips as we experience them (or those suggested by other users). These will be added progressively as we encounter them.

## **45.1 DEBUG MESSAGES AND LOG FILES ARE YOUR FRIENDS**

Whenever problem strikes, there is always a reason for it. To assist you in locating the problem is to monitor the activities of your Elastix using the following command at the command line (which will give you the Asterisk CLI).

Asterisk -rvvvv (the "v" depends on the level of verbosity you need)

```
🚰 root@asterisk1:~
                                                                                                [root@asterisk1 ~]# asterisk -rvvvv
  == Parsing '/etc/asterisk/asterisk.conf': Found
== Parsing '/etc/asterisk/extconfig.conf': Found
 sterisk 1.2.13 svn rev 47264, Copyright (C) 1999 - 2006 Digium, Inc. and others
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'show license' for details.
                             _______
Connected to Asterisk 1.2.13 svn rev 47264 currently running on asterisk1 (pid =
 3154)
 Verbosity was 1 and is now 4
     -- Executing Macro("SIP/2001-08e5ae00", "user-callerid|") in new stack
-- Executing NoOp("SIP/2001-08e5ae00", "user-callerid: device 2001") in new
stack
     -- Executing GotoIf("SIP/2001-08e5ae00", "0?report") in new stack
-- Executing GotoIf("SIP/2001-08e5ae00", "0?start") in new stack
        Executing Set("SIP/2001-08e5ae00", "REALCALLERIDNUM=2001") in new stack
Executing NoOp("SIP/2001-08e5ae00", "REALCALLERIDNUM is 2001") in new sta
ck
     -- Executing Set("SIP/2001-08e5ae00", "AMPUSER=2001") in new stack
-- Executing Set("SIP/2001-08e5ae00", "AMPUSERCIDNAME=Ben Sharif")
                                                                                            in new
```

Studying the activities will give you an indication of what has gone wrong.

## 45.2 ASTERISK FEATURE CODES NOT WORKING

If you are using ATAs such as the Sipura SPA3K, Linksys PAP2 etc., you will need to delete all the Vertical Service Feature codes in the ATA, otherwise the Asterisk feature codes such as \*77, \*99 etc may not work. This is caused by the ATA intercepting the codes thinking that it is its own feature codes.

This also applies to all other ATAs, FXS devices such as the various routers with built-in FXS ports and IP Phones that has their own feature codes built-in. Unless these feature codes are disabled, they are likely to clash with Asterisk's feature codes and as such Asterisk feature codes will not give you the desired results.

## 45.3 ASTERISK DROPS CALLS AFTER A FEW SECONDS

If Asterisk rings an extension but drops the calls too soon before it was answered, chances are, you may have set the '*Number of seconds to ring phones before sending callers to voicemail:*' too short. This setting can be found in the **General Setting** under **Voicemail**.

## 45.4 PENNYTEL ASTERISK PROBLEM

Pennytel, for some reason rejects user agent if it is Asterisk. To work around this, you are required to introduce the following line in your **sip.conf** under **[general]** 

useragent = My PABX ; required if you are with Pennytel

The user agent can be any string as long as it is not Asterisk.

## **45.5 UNABLE TO RECEIVE INCOMING CALLS**

If you are unable to receive incoming calls, chances are that you may have missed setting up the incoming call handling as suggested in the previous sections of this guide.

A quick fix for ELASTIX is to create an Inbound Route with the DID and CID field left blank. At least this will let incoming calls through while you try to solve the problem.

Please refer to the following section on <u>Calls Handling</u> in this guide:

## 45.6 UNABLE TO SEND OUT VOICEMAIL MAIL NOTIFICATION

THE Mailling is your source of information as to why your mail wont go out. Usually looking through the maillog will give you an indication of what went wrong as illustrated below:



The mail log resides in /var/log/maillog

## 45.7 SIPURA SPA3000 - NOT HANGING UP AFTER PSTN CALLS

For months I have to wrestle with the problem of SPA3K not hanging up after a PSTN call. Although the caller has hung up, SPA3K will hold the line off hook until I manually power cycle SPA3K.

The following settings may overcome the problem:

## PSTN Line Tab (Advanced Mode)

## **PSTN Disconnect Detection**

Detect CPC: yes Detect Polarity Reversal: yes Detect PSTN Long Silence: yes Detect VoIP Long Silence: yes PSTN Long Silence Duration: 180 VoIP Long Silence Duration: 180 PSTN Silence Threshold: medium Min CPC Duration: 0.2 Detect Disconnect Tone: no Disconnect Tone: Leave blank

PSTN Disconnect Detection						
Detect CPC:	yes	<b>v</b>	Detect Polarity Reversal:	yes	<b>v</b>	
Detect PSTN Long Silence:	yes	<b>v</b>	Detect VoIP Long Silence:	yes	<b>v</b>	
PSTN Long Silence Duration:	180		VoIP Long Silence Duration:	180		
PSTN Silence Threshold:	high	<b>V</b>	Min CPC Duration:	0.2		
Detect Disconnect Tone:	yes	<b>v</b>				
Disconnect Tone:	425@	-30,425@-30	(1(.390/.390/1+2)			



*I* deleted my original Disconnect Tone setting of: **425@-30,425@-30;1(.375/.375/1+2)** which seems to work well for Telstra, and replaced it with **425@-30,425@-30;1(.390/.390/1+2)** which works better with Optus.

Only one catch with this: if you put somebody on hold for longer than the above settings, if they don't make any noise, it will hang up.

## 45.8 VIRTUAL FAX LIST DOES NOT DISPLAY EXISTING VIRTUAL FAX.

An error in one of the lines of */var/www/html/libs/paloSantoFax.class.php* (in line 146) caused this problem.

Change the line from

```
if(is_array($result))
```

to

```
if(isset($result))
```

Once this is done, the display will return.

# **46 MY ELASTIX® PBX NETWORK**

The following is a representation of my network schematic on my Local Area Network with my Asterisk box as I have done according to my narration above.



# **APPENDIX A**

## A.1 ASTERISK FEATURE CODES

Asterisk feature codes are defined in the feature codes selection of Elastix PBX Configuration screen, illustrated below:

	System PBX Fax Email IM Reports	Extra	5			About Ela	istix 1.0-12	Logout
PBX Configuration Flash Oper	rator Panel   Voicemails   Monitoring   Echo Cancelle	r Endpo	int Con	figuration	Conference	Extensions Batch	Tools	0 0
Option Unembedded freePBX	Feature Code Admin							
Extensions Feature Codes	Placklist	De	Use fault?	Feature Status				
General Settings								
Outbound Routes	Blacklist a number	*30		Enabled 💌	1			
Trunks	Blacklist the last caller	*32		Enabled 💌	, ]			
Inbound Call Control	Remove a number from the blacklist	*31	2	Enabled 💌	1			
Appound Koules	Call Forward							
Follow Me								
TVP	Call Forward All Activate	*72	<b>V</b>	Enabled 💌	]			
Misc Destinations	Call Forward All Deactivate	*73	<b>V</b>	Enabled 💌	]			
Oueues	Call Forward All Prompting Deactivate	*74	-	Enabled 💌				
Ring Groups	Call Forward Busy Activate	*90	2	Enabled 💌				
Time Conditions	Call Forward Busy Deactivate	*91		Enabled 💌	1			
Internal Options & Configuration	Call Forward Busy Prompting Deactivate	*92		Enabled 💌	1			
Conferences	Call Forward No Answer/Unavailable Activate	*52	<b>V</b>	Enabled 💌	1			
Misc Applications	Call Forward No Answer/Unavailable Deactivate	*53		Enabled 💌				
Music on Hold	Call Waiting							
PIN Sets								
Paging and Intercom	Call Waiting - Activate	*70		Enabled 💌	1			
Parking Lot	Call Waiting - Deactivate	*71	•	Enabled 💌				
System Recordings	Core							
Remote Access								
Callback	Call Pickup (Can be used with GXP-2000)	••	¥	Enabled 💌				
DISA	ChanSpy	555	2	Enabled 🔛	1			
	Dial System FAX	666		Enabled 💌				
	In-Call Astensk Attended Transfer	*2		Enabled 💌				
	In-Call Asterisk Blind Transfer	##		Inabled M				
	In-Call Asterisk Toggle Call Recording	*1	1	Enabled 💌				
	Simulate Incoming Call	7777	1	Enabled 🛩				

- # Directory
- ## In-Call Asterisk Blind Transfer
- \*\* Call Pickup
- \*0 Speed Dial Prefix
- \*1 In-Call Asterisk Toggle Call Recording
- \*2 In-Call Asterisk Attended Transfer
- \*11 User Log on
- \*12 User Log off
- \*28 Day/Night toggle
- \*30 Blacklist a number
- \*31 Remove a number from the blacklist
- \*32 Blacklist the last caller
- \*34 Perform dictation
- \*35 Email completed dictation
- \*43 Echo Test
- \*52 Call Forward No Answer/Unavailable Activate
- \*53 Call Forward No Answer/Unavailable Deactivate
- \*54 User Intercom Allow
- \*55 User Intercom Disallow
- \*60 Speaking Clock
- \*65 Speak your Extension number (your extension is XXX)
- \*69 Last Caller number/ID (Call trace)
- \*70 Activate Call Waiting (deactivated by default)
- \*71 Deactivate Call Waiting
- \*72 Call Forwarding System
- \*73 Disable Call Forwarding
- \*74 Call Forward all prompting deactivate

- \*75 Set user speed dial
- \*77 IVR Recording
- \*78 Enable Do-Not-Disturb
- \*79 Disable Do-Not-Disturb
- \*80 Intercom prefix
- \*90 Call Forward on Busy
- \*91 Disable Call Forward on Busy
- \*92 Call Forward Busy Prompting deactivate
- \*97 Message Center (My Voicemail)
- \*97 + 0 Change email greeting message.
- \*98 Enter Message Center
- \*99 Playback IVR Recording
- \*422 Connect to Gabcast
- 70 Call Parking (or 700 see Call Parking and Transfer)
- 411 Phonebook dial-by-name directory
- 555 ChanSpy
- 666 Test Fax
- 888 Zap Barge (there are variations)
- 7777 Simulate incoming call

## A.1.1 Wakeup Call Feature \*62 missing

Those who are familiar with the Wakeup-Call feature will find that the feature is missing. This is relatively easy to remedy as the necessary agi is included within this distribution although not included in the feature.

To recover this functionality, all you need to do is simply include the following (in bold) in your **extensions\_custom.conf** file as per my example below. Place it directly under the [from-internal-custom] context.

[from-internal-custom]

```
exten => *62,1,Answer
exten => *62,2,AGI(wakeup.php)
exten => *62,3,Hangup
```

Save it and restart asterisk.

## A.2 ASTERISK CLI COMMANDS

These are some of the available CLI commands that can be executed from the console when you run: **asterisk** –**r** (or –**rvvv** depending on the level of verbosity you may want).

To stop ELASTIX – **amportal stop** To start ELASTIX – **amportal start** 

## A.2.1 General commands

!<command>: Executes a given shell command abort halt: Cancel a running halt add extension: Add new extension into context add ignorepat: Add new ignore pattern add indication: Add the given indication to the country amportal start: Stop ELASTIX and amportal stop: Restart ELASTIX. debug channel: Enable debugging on a channel dont include: Remove a specified include from context help: Display help list, or specific help on a command

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include context: Include context in other context

load: Load a dynamic module by name

logger reload: Reopen log files. Use after rotating the log files.

no debug channel: Disable debugging on a channel

pri debug span: Enables PRI debugging on a span

pri intense debug span: Enables REALLY INTENSE PRI debugging

pri no debug span: Disables PRI debugging on a span

remove extension: Remove a specified extension

**remove ignorepat**: Remove ignore pattern from context

remove indication: Remove the given indication from the country

save dialplan: Overwrites your current extensions.conf file with an exported version based on the current state of the dialplan. A backup copy of your old extensions.conf is not saved. The initial values of global variables defined in the [globals] category retain their previous initial values; the current values of global variables are not written into the new extensions.conf. Using "save dialplan" will result in losing any comments in your current extensions.conf.

set verbose: Set level of verboseness

**show** agents: Show status of agents

show applications: Shows registered applications

**show** application: Describe a specific application

show channel: Display information on a specific channel

show channels: Display information on channels

**show codecs:** Display information on codecs

show conferences: Show status of conferences

**show dialplan:** Show dialplan

**show image formats**: Displays image formats

**show indications**: Show a list of all country/indications

**show locals**: Show status of local channels

show manager command: Show manager commands

show manager connect: Show connected manager users

**show parkedcalls:** Lists parked calls

**show queues:** Show status of queues

**show switches**: Show alternative switches

show translation: Display translation matrix

show voicemail users: List defined voicemail boxes

**show voicemail zones**: List zone message formats

soft hangup: Request a hangup on a given channel

## A.2.2 AGI Commands

show agi: Show AGI commands or specific help
dump agihtml: Dumps a list of agi command in html format

#### A.2.3 Database Handling

database del: Removes database key/value database deltree: Removes database keytree/values database get: Gets database value database put: Adds/updates database value database show: Shows database contents

### A.2.4 IAX Channel Commands

iax2 debug: Enable IAX debugging

- iax2 no debug: Disable IAX debugging
- iax2 set jitter: Sets IAX jitter buffer
- iax2 show cache: Display IAX cached dialplan

iax2 show channels: Show active IAX channels

iax2 show peers: Show defined IAX peers iax2 show registry: Show IAX registration status iax2 show stats: Display IAX statistics iax2 show users: Show defined IAX users iax2 trunk debug: Request IAX trunk debug iax debug: Enable IAX debugging iax no debug: Disable IAX debugging iax set jitter: Sets IAX jitter buffer iax show cache: Display IAX cached dialplan iax show channels: Show active IAX channels iax show peers: Show defined IAX peers iax show registry: Show IAX registration status iax show stats: Display IAX statistics iax show users: Show defined IAX users init keys: Initialize RSA key passcodes show keys: Displays RSA key information

#### A.2.5 SIP Channel commands

sip debug: Enable SIP debugging
sip no debug: Disable SIP debugging
sip reload: Reload sip.conf (added after 0.7.1 on 2004-01-23)
sip show channels: Show active SIP channels
sip show channel: Show detailed SIP channel info
sip show inuse: List all inuse/limit
sip show peers: Show defined SIP peers (register clients)
sip show registry: Show SIP registration status (when Asterisk registers as a
client to a SIP Proxy)
sip show users: Show defined SIP users

## A.2.6 Server management

restart gracefully: Restart Asterisk gracefully
restart now: Restart Asterisk immediately
restart when convenient: Restart Asterisk at empty call volume
reload: Reload configuration
stop gracefully: Gracefully shut down Asterisk
stop now: Shut down Asterisk immediately
stop when convenient: Shut down Asterisk at empty call volume
extensions reload?: Reload extensions ONLY
unload: Unload a dynamic module by name
show modules: List modules and info about them
show uptime: Show uptime information
show version: Display Asterisk version info

## A.3 ASTERISK SPECIAL EXTENSIONS

The following are some Asterisk Special Extensions that are useful to know when creating your Dial Plan or Special Custom Context.

 $\mathbf{i}$  = invalid, jump here when an invalid number is dialled.

**t** = timeout, jump here when we timeout.

**h** = hang-up, jump here when a call in hung up.

**s** = standard, jump here if the dialled extension is unknown (used for generic processing).

**o** = operator, a special extension to escape voice mail (configurable)
# A.4 ASTERISK COMMON VARIABLES

The following are some of the special variables in use with the Asterisk scripts as defined in the Asterisk's document. These variables are automatically defined when a new call starts.

\$•	{ACCOUNTCODE }	Account code (if specified)
\$•	{CALLERID}	Caller ID
\$•	{CALLERIDNAME}	Caller ID Name only
\$•	{CALLERIDNUM}	Caller ID Number only
\$•	{CALLINGPRES}	PRI Caller ID presentation for incoming calls
\$•	(CHANNEL )	Current channel name
\$•	{CONTEXT}	Current context
\$•	{DATETIME}	Current date time in the format: YYYY-MM-DD_HH:MM:SS
\$•	(DNID)	Dialed Number Identifier
\$•	{ENUM}	Result of application EnumLookup
\$•	{EPOCH}	Current unix style epoch
\$•	{EXTEN}	Current extension
\$•	ENV(VAR)	Environmental variable VAR
\$•	{HANGUPCAUSE}	Asterisk hangup cause
\$•	{INVALID_EXTEN}	The invalid called extension (used in the "i" extension)
\$•	{LANGUAGE}	Current language
\$•	{LEN(VAR)}	String length of VAR (integer)
\$•	{MEETMESECS}	Number of seconds a user participated in a MeetMe
\$•	{PRIORITY}	Current priority
\$•	{RDNIS}	Redirected Dial Number ID Service
\$•	{TIMESTAMP}	Current date time in the format: YYYYMMDD-HHMMSS
\$•	{TXTCIDNAME}	Result of application TXTCIDName
\$•	{UNIQUEID}	Current call unique identifier
\$•	{SIPCALLID}	SIP Call-ID: header verbatim (for logging or CDR matching)
\$	{SIPDOMAIN}	SIP destination domain of an inbound call (if appropriate)
\$•	<pre>{SIPUSERAGENT}</pre>	SIP user agent

The following are set by the dial() application:

\${DIALEDPEERNAME}	Dialed peer name
\${DIALEDPEERNUMBER}	Dialed peer number
\${DIALEDTIME}	Total time for the call in seconds (Network time).
\${ANSWEREDTIME}	Time from answer to end of call in seconds.
\${DIALSTATUS}	Status of the call, one of:   CHANUNAVAIL   CONGESTION   BUSY
	NOANSWER   ANSWER   CANCEL

## A.5 INDICATIONS FOR VARIOUS COUNTRIES

The following are some of the indications for the various countries that I am aware off. Please submit others that you know of to me.

```
[at]
ringcadence = 1000,5000
dial = 420
busy = 420/400,0/400
ring = 420/1000,0/5000
congestion = 420/200,0/200
callwaiting = 420/40,0/1960
dialrecall = 420
record = 1400/80,0/14920
info = 950/330,1450/330,1850/330,0/1000
stutter = 380+420
```

```
[au]
ringcadence = 400,200,400,2000
dial = 413+438
busy = 425/375,0/375
ring = 413+438/400,0/200,413+438/400,0/2000
congestion = 425/375,0/375,420/375,0/375
callwaiting = 425/200,0/200,425/200,0/4400
dialrecall = 413+438
record = 1425/1000,10/15000,425/360,0/15000
info = 425/2500,0/500
std = 1525/100,10/100,1525/100,10/100,1525/100,10/100,1525/100
facility = 425
stutter = 413+438/100,0/40
ringmobile = 400+450/400,0/200,400+450/400,0/2000
```

## [br]

ringcadence = 1000,4000dial = 425busy = 425/250,0/250ring = 425/1000,0/4000congestion = 425/250,0/250,425/750,0/250callwaiting = 425/50,0/1000dialrecall = 350+440record = 425/250,0/250info = 950/330,1400/330,1800/330stutter = 350+440

### [be]

ringcadence = 1000,3000dial = 425busy = 425/500,0/500ring = 425/1000,0/3000congestion = 425/167,0/167callwaiting = 1400/175,0/175,1400/175,0/3500dialrecall = 1350+440/100,10/100,1350+440/100,10/100,350+440record = 1400/500,0/15000info = 900/330,1400/330,1800/330,0/1000stutter = 425/1000,0/250

## [ch]

ringcadence = 1000,4000dial = 425busy = 425/500,0/500ring = 425/1000,0/4000congestion = 425/200,0/200callwaiting = 425/200,0/200,425/200,0/4000dialrecall = !425/100,!0/100,!425/100,!0/100,425record = 1400/80,0/15000info = 950/330,1400/330,1800/330,0/1000stutter = 425+340/1100,0/1100

#### [cl]

ringcadence = 1000,3000dial = 400busy = 400/500,0/500ring = 400/1000,0/3000congestion = 400/200,0/200callwaiting = 400/250,0/8750dialrecall = !400/100,!0/100,!0/100,!0/100,!0/100,400record = 1400/500,0/15000 info = 950/333,1400/333,1800/333,0/1000 stutter = !400/100,!0/100,!400/100,!0/100,!400/100,!0/100,!400/100,!0/100,!400/100 0,!0/100,400

#### [cn]

ringcadence = 1000,4000dial = 450busy = 450/350,0/350ring = 450/1000,0/4000congestion = 450/700,0/700callwaiting = 450/400,0/4000dialrecall = 450record = 950/400,0/10000info = 450/100,0/100,450/100,0/100,450/400,0/400stutter = 450+425

#### [cz]

ringcadence = 1000,4000dial = 425/330,0/330,425/660,0/660busy = 425/330,0/330ring = 425/1000,0/4000congestion = 425/165,0/165callwaiting = 425/330,0/9000dialrecall = !425/100,!0/100,!425/100,!0/100,425/330,0/330,425/660,0/660record = 1400/500,0/14000info = 950/330,0/30,1400/330,0/30,1800/330,0/1000stutter = 425/450,0/50

## [de]

```
ringcadence = 1000,4000
dial = 425
busy = 425/480,0/480
ring = 425/1000,0/4000
congestion = 425/240,0/240
callwaiting =
!425/200,!0/200,!425/200,!0/5000,!425/200,!0/200,!425/200,!0/5000,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/200,!0/200,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0/200,!0
```

#### [dk]

```
ringcadence = 1000,4000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = !425/200,!0/600,!425/200,!0/3000,!425/200,!0/200,!425/200,0
dialrecall = !425/100,!0/100,!425/100,!0/100,!425/100,!0/100,425
record = 1400/80,0/15000
info = 950/330,1400/330,1800/330,0/1000
stutter = 425/450,0/50
```

```
[ee]
ringcadence = 1000,4000
dial = 425
busy = 425/300,0/300
```

ring = 425/1000,0/4000congestion = 425/200,0/200callwaiting = 950/650,0/325,950/325,0/30,1400/1300,0/2600dialrecall = 425/650,0/25record = 1400/500,0/15000info = 950/650,0/325,950/325,0/30,1400/1300,0/2600stutter = !425/100,!0/100,!0/100,!0/100

### [es]

ringcadence = 1500,3000dial = 425busy = 425/200,0/200ring = 425/1500,0/3000congestion = 425/200,0/200,425/200,0/200,425/200,0/600callwaiting = 425/175,0/175,425/175,0/3500dialrecall = !425/200,!0/200,!425/200,!0/200,!425/200,!0/200,425record = 1400/500,0/15000info = 950/330,0/1000dialout = 500

### [fi]

ringcadence = 1000,4000dial = 425busy = 425/300,0/300ring = 425/1000,0/4000congestion = 425/200,0/200callwaiting = 425/150,0/150,425/150,0/8000dialrecall = 425/650,0/25record = 1400/500,0/15000info = 950/650,0/325,950/325,0/30,1400/1300,0/2600stutter = 425/650,0/25

## [fr]

```
ringcadence = 1500,3500
dial = 440
busy = 440/500,0/500
ring = 440/1500,0/3500
congestion = 440/250,0/250
callwait = 440/300,0/10000
dialrecall = !350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330
stutter =
!440/100,!0/100,!440/100,!0/100,!440/100,!0/100,!440/100,!0/100,!440/100
```

## [gr]

```
ringcadence = 1000,4000
dial = 425/200,0/300,425/700,0/800
busy = 425/300,0/300
ring = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = 425/150,0/150,425/150,0/8000
dialrecall = 425/650,0/25
record = 1400/400,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000,0
```

#### stutter = 425/650,0/25

#### [hk]

ringcadence = 400,200,400,2000dial = 350+440busy = 480+620/500,0/500ring = 440+480/400,0/200,440+480/400,0/2000congestion = 480+620/250,0/250callwaiting = 440/100,0/4000dialrecall = 300/1500,0/500record = 1400/500,0/15000info = !950/330,!1400/330,!1800/330

#### [hu]

```
ringcadence = 1250,3750
dial = 425
busy = 425/300,0/300
ring = 425/1250,0/3750
congestion = 425/300,0/300
callwaiting = 425/40,0/1960
dialrecall = 425+450
record = 1400/400,0/15000
info =
1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,11800/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,11400/330,10/1000,1950/330,10/1000,1950/330,10/1000,0
```

## [il]

```
; Source: libtonezone.
description = lsrael
ringcadence = 1000,3000
dial = 414
busy = 414/500,0/500
ring = 414/1000,0/3000
congestion = 414/250,0/250
callwaiting = 414/100,0/100,414/100,0/100,414/600,0/3000
dialrecall = !414/100,!0/100,!414/100,!0/100,!414/100,!0/100,414
record = 1400/500,0/15000
info = 1000/330,1400/330,1800/330,0/1000
stutter =
!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!400,!0/160,!400,!000,!000,!000,!000,!000,!00
```

#### [in]

```
; Source: libtonezone.
description = India
ringcadence = 400,200,400,2000
dial = 400*25
busy = 400/750,0/750
ring = 400*25/400,0/200,400*25/400,0/2000
congestion = 400/250,0/250
callwaiting = 400/200,0/100,400/200,0/7500
dialrecall = 1350+440/100,10/100,1350+440/100,10/100,1350+440/100,10/100,350+440
record = 1400/500,0/15000
info = 1950/330,11400/330,11800/330,0/1000
stutter =
1350+440/100,10/100,1350+440/100,10/100,1350+440/100,10/100,1350+440/100,10/100,1350
```

```
0+440/100,!0/100,!350+440/100,!0/100,350+440
```

```
ringcadence = 1000,4000
dial = 425/200,0/200,425/600,0/1000
busy = 425/500,0/500
ring = 425/1000.0/4000
congestion = 425/200,0/200
callwaiting = 425/400,0/100,425/250,0/100,425/150,0/14000
dialrecall = 470/400,425/400
record = 1400/400,0/15000
info =
1950/330.11400/330.11800/330.10/1000.1950/330.11400/330.11800/330.10/1000.1950/330.11
400/330.!1800/330.!0/1000.0
stutter = 470/400,425/400
[ip]
: Source: libtonezone.
description = Japan
ringcadence = 1000,2000
dial = 400
busy = 400/500,0/500
ring = 400+15/1000,0/2000
congestion = 400/500,0/500
callwaiting = 400 + 16/500, 0/8000
dialrecall = !400/200,!0/200,!400/200,!0/200,!400/200,!0/200,400
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330,0
stutter =
0,10/100,400
[lt]
ringcadence = 1000,4000
dial = 425
busy = 425/350,0/350
ring = 425/1000,0/4000
congestion = 425/200, 0/200
callwaiting = 425/150,0/150,425/150,0/4000
dialrecall = 425/500,0/50
record = 1400/500,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000,0
stutter =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,
0,10/100,425
[mx]
ringcadence = 2000,4000
dial = 425
busy = 425/250,0/250
ring = 425/1000,0/4000
congestion = 425/250, 0/250
callwaiting = 425/200,0/600,425/200,0/10000
dialrecall = !350+440/100.!0/100.!350+440/100.!0/100.!350+440/100.!0/100.350+440
record = 1400/500, 0/15000
info = 950/330,0/30,1400/330,0/30,1800/330,0/1000
stutter =
0+440/100,!0/100,!350+440/100,!0/100,350+440
```

```
[nl]
ringcadence = 1000,4000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/4000
congestion = 425/250, 0/250
callwaiting = 425/500,0/9500
dialrecall = 425/500,0/50
record = 1400/500,0/15000
info = 950/330,1400/330,1800/330,0/1000
stutter = 425/500,0/50
```

## [no]

```
ringcadence = 1000,4000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = 425/200,0/600,425/200,0/10000
dialrecall = 470/400,425/400
record = 1400/400,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000,0
stutter = 470/400,425/400
```

```
[nz]
```

```
ringcadence = 400,200,400,2000
dial = 400
busy = 400/250,0/250
ring = 400+450/400,0/200,400+450/400,0/2000
congestion = 400/375.0/375
callwaiting = !400/200.!0/3000.!400/200.!0/3000.!400/200
dialrecall = !400/100!0/100,!400/100,!0/100,!400/100,!0/100,400
record = 1400/425,0/15000
info = 400/750,0/100,400/750,0/100,400/750,0/100,400/750,0/400
stutter =
!400/100!0/100,!400/100,!0/100,!400/100,!0/100,!400/100!0/100,!400/100,!0/100,!400/100,
10/100,400
```

## [pl]

```
ringcadence = 1000,4000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/4000
congestion = 425/500,0/500
callwaiting = 425/150,0/150,425/150,0/4000
dialrecall = 425/500,0/50
record = 1400/500,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000
stutter =
0,10/100,425
[pt]
ringcadence = 1000,5000
```

```
dial = 425
```

busy = 425/500,0/500 ring = 425/1000,0/5000 congestion = 425/200,0/200 callwaiting = 440/300,0/10000 dialrecall = 425/100,0/200 record = 1400/500,0/15000 info = 950/330,1400/330,1800/330,0/1000 stutter = !425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100 0,!0/100,425

#### [ru]

ringcadence = 800,3200dial = 425busy = 425/350,0/350ring = 425/800,0/3200congestion = 425/350,0/350callwaiting = 425/200,0/5000dialrecall = !350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440record = 1400/500,0/15000info = !950/330,!1400/330,!1800/330,0

### [se]

ringcadence = 1000,5000 dial = 425 busy = 425/250,0/250 ring = 425/1000,0/5000 congestion = 425/250,0/750 callwaiting = 425/200,0/500,425/200,0/9100 dialrecall = !425/100,!0/100,!425/100,!0/100,!425/100,!0/100,425 record = 1400/500,0/15000 info = !950/332,!0/24,!1400/332,!0/24,!1800/332,!0/2024,!950/332,!0/24,!1400/332,!0/24,!1800/3 32,!0/2024,!950/332,!0/24,!1400/332,!0/24,!1800/332,!0/2024,!950/332,!0/24,!1400/332,!0/24,!1800/332,!0/24,!1800/332,!0/24,!1400/332,!0/24,!1800/32,!0/20,!0/100,!425/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/10

## [sg]

```
ringcadence = 400,200,400,2000
dial = 425
ring = 425*24/400,0/200,425*24/400,0/2000 ; modulation should be 100%, not 90%
busy = 425/750,0/750
congestion = 425/250,0/250
callwaiting = 425*24/300,0/200,425*24/300,0/3200
stutter =
!425/200,!0/200,!425/600,!0/200,!425/200,!0/200,!425/600,!0/200,!425/200,!0/200,!425/60
0,!0/200,!425/200,!0/200,!425/600,!0/200,425
info = 950/330,1400/330,1800/330,0/1000; not currently in use acc. to reference
dialrecall = 425*24/500,0/500,425/500,0/2500; unspecified in IDA reference, use repeating
Holding Tone A,B
record = 1400/500,0/15000; unspecified in IDA reference, use 0.5s tone every 15s
nutone = 425/2500,0/500
intrusion = 425/250.0/2000
warning = 425/624,0/4376 ; end of period tone, warning
acceptance = 425/125,0/125
holdinga = !425*24/500,!0/500 ; followed by holdingb
holdingb = !425/500,!0/2500
```

#### [uk]

```
ringcadence = 400,200,400,2000
dial = 350 + 440
specialdial = 350+440/750,440/750
busy = 400/375,0/375
congestion = 400/400,0/350,400/225,0/525
specialcongestion = 400/200,1004/300
unobtainable = 400
ring = 400+450/400,0/200,400+450/400,0/2000
callwaiting = 400/100.0/4000
specialcallwaiting = 400/250,0/250,400/250,0/250,400/250,0/5000
creditexpired = 400/125, 0/125
confirm = 1400
switching = 400/200,0/400,400/2000,0/400
info = 950/330,0/15,1400/330,0/15,1800/330,0/1000
record = 1400/500,0/60000
stutter = 350+440/750,440/750
```

### [us]

## [us-old]

```
\begin{aligned} & \text{ringcadence} = 2000,4000 \\ & \text{dial} = 600^*120 \\ & \text{busy} = 500^*100/500,0/500 \\ & \text{ring} = 420^*40/2000,0/4000 \\ & \text{congestion} = 500^*100/250,0/250 \\ & \text{callwaiting} = 440/300,0/10000 \\ & \text{dialrecall} = !600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,600^*120 \\ & \text{record} = 1400/500,0/15000 \\ & \text{info} = !950/330,!1400/330,!1800/330,0 \\ & \text{stutter} = \\ !600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!600^*120/100,!0/100,!0/100,!600^*120/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100,!0/100
```

## [tw]

# **APPENDIX B**

# **B.1 – AUSTRALIAN VSPS (TRUNKS)**

The following is a collection of TRUNK settings recommended for other VSPs that are not covered above. Details of trunks configuration are correct at the time they were compiled, however changes may have been made. Users are advised to consult their respective VSPs for confirmation.

**Note:** In Peer Details, insert **qualify=yes** if you want fail-over to PSTN should you loose your internet connection.

## B.1.1 BBPglobal

**Peer Details** 

Trunk Name: bbpglobal

```
allow=g723&gsm
authuser=<user number>
disallow=all
fromdomain=sip2.bbpglobal.com
fromuser=<user number>
host= sip2.bbpglobal.com
insecure=very
secret=<password>
type=peer
username=<user number>
```

User Details User Context: <user number>

context=from-trunk
fromuser=<user number>
insecure=very
secret=<password>
type=user
username=<user number>
Note:You may need to include the
following in your sip.conf
Maxexpirey=3600
Qualify=yes

## **Register String:**

<user number>:<password>sip2.bbpglobal.com/<user number>

## B.1.2 Engin BYO

## **Peer Details**

Trunk Name: engin

```
allow=ulaw&alaw
auth=md5
canreinvite=yes
disallow=all
dtmfmode=rfc2833
fromdomain=voice.mibroadband.com.au
fromuser=02321XXXX
host=byo.engin.com.au
insecure=very
musiconhold=framed
nat=yes
port=5060
qualify=no
realm=mobileinnovations.com.au
reinvite=yes
secret=<password>
type=friend
username=02321XXXX
```

User Context: 02321XXXX

context=from-pstn fromdomain=voice.mibroadband.com.au host=byo.engin.com.au secret=<password> type=user username=02321XXXX

Note: You may need to include the following in your **sip.conf** 

Defaultexpirey=600
Maxexpirey=3600
externip=(your external ip address)
localnet = 192.168.1.0/255.255.255.0

Changing the fromdomain to byo.engin.com.au will also influnce the outcome.

## **Register String:**

02321XXXX:<password>@byo.engin.com.au/02321XXXX

## B.1.3 Faktortel (IAX)

## **Peer Details**

Trunk Name: faktortel

```
allow=g729&gsm&ulaw&alaw&ilbc
disallow=all
host=iax.faktortel.com.au
qualify=3000
secret=<password>
type=friend
username=xxxxxx
```

# User Details

User Context: xxxxxx

```
auth=md5
context=from-trunk
host=iax.faktortel.com.au
qualify=3000
secret=<password>
type=friend
username=xxxxxx
```

## **Register String:**

xxxxxx:<password>@iax.faktortel.com.au

Faktortel supports the following codecs: ulaw, alaw, ilbc, gsm, g729. The following is the link on the discussions for setting up <u>Faktortel</u> on ELASTIX. <u>http://www.faktortel.com.au/support-Elastix-settings.shtml</u>

## B.1.4 GoTalk

Peer Details Trunk Name: GoTalk

```
allow=alaw
disallow=all
dtmfmode=inband
username=094XXXXX
fromuser=094XXXXX
insecure=very
nat=yes
host=sip.gotalk.com
qualify=yes
secret=<password>
type=peer
```

User Details User Context: 094XXXXX

context=from-trunk
fromuser=094XXXXX
host=dynamic
insecure=very
nat=yes
secret=<password>
type=user
username=094XXXXX

The following may also neet to be set in sip\_custom.conf

useragent = whatever (as long as it is not asterisk)

## **Register String:**

094XXXXX:<password>@sip.gotalk.com/094XXXXX

#### B.1.5 iinet

**Peer Details** Trunk Name: iinet

```
allow=alaw&ulaw
canreinvite=no
disallow=all
username=<phone no>
fromdomain=iinetphone.iinet.net.au
fromuser=<phone no>
insecure=very
nat=yes
host=sip.nsw.iinet.net.au
pedantic=no
qualify=yes
secret=<password>
type=peer
```

User Details User Context: <phone no)

context=from-trunk
fromuser=<phone no>
host= sip.nsw.iinet.net.au
insecure=very
nat=yes
secret=<password>
type=user
username=<phone no>

The following may also neet to be set in sip\_custom.conf

useragent = whatever (as long as it is not asterisk)

#### **Register String:**

<phone number>@iinetphone.iinet.net.au:< password>:<phone number>@<trunk name>/<phone number>

## B.1.6 <u>iPrimus</u>

Primus is a strange animal. To get it to work with Elastix requires me to do a bit of detective work. With my partner in crime, Colin, we set out to play detective because we need to obtain the correct credential to get Elastix to register and to be able to make and receive calls through Elastix. Primus Telecom has refused to assist.

To get the necessary credentials, we cheated a little as the method was published in the whirlpool forum <a href="http://forums.whirlpool.net.au/forum-replies.cfm?t=604015">http://forums.whirlpool.net.au/forum-replies.cfm?t=604015</a>

First of all, using a browser, we need to enter the following address:

https://spconfig.voip.iprimus.net.au/SoftphoneConfigurationServer/Default.aspx?Action=LOG IN&username=**username**&password=**password**&computerid=db7b351e57af&MAC=**your** \_**NIC\_MAC\_address** 

Replace the bold type with your Talk Broadband username and password given to you by iPrimus. You can get your NIC MAC address by running *ipconfig /all*.

When you submit that address, the following data will be displayed in your browser.

[Data] Success=1 [Settings] proxies:proxy0:display\_name="02\*\*\*\*\*\*"
proxies:proxy0:username="AU\_\*\*\*\*\*\*\*" proxies:proxy0:password="\*\*\*\*\*"
system:auto\_update:config\_server\_url="https://spconfig.voip.iprimus.net.au/Soft
phoneConfigurationServer/Default.aspx?Action=CONFIG&Username=username&Version=1
&MAC\$\$MAC\$&computerid=\$computerid\$" audio:headset:audio\_in\_agc\_enabled="0"
audio:speaker\_phone:audio\_in\_agc\_enabled="0" codecs:speex\_fec:enabled="0"
codecs:speex\_wb:starting\_bandwidths="24000,26000,28800,32800,40000,44000"
codecs:speex\_wb:starting\_bandwidth="44000"
codecs:speex\_wb:usable\_bandwidths="20800,24800,28800,36800,44000"
codecs:speex\_wb:contacts\_last\_upload\_failed="0" proxies:proxy0:contacts\_etag=""
proxies:proxy0:contacts\_last\_upload\_failed="0" proxies:proxy0:contacts\_mod="0"
proxies:proxy0:default\_session\_refresh\_interval\_in\_seconds="60"
proxies:proxy0:domain="anjla2.bw.iprimus.net"

proxies:proxy0:enable\_session\_timer="0" proxies:proxy0:enabled="1"
proxies:proxy0:ice\_enabled="1"
proxies:proxy0:maximum\_time\_before\_reregistration\_in\_seconds="1800"
proxies:proxy0:minimum\_time\_before\_reregistration\_in\_seconds="20"
proxies:proxy0:override\_outbound\_proxy="1"
proxies:proxy0:override\_sip\_listen\_port="0"
proxies:proxy0:override\_stun\_server="1" proxies:proxy0:port\_range\_enable="0"
proxies:proxy0:register="1" proxies:proxy0:reregister\_in\_seconds="3600"

Note the bold highlights. These are the data that you need for your trunk setting. Armed with the data above, you may now set your trunk setting.

## **Peer Details**

Trunk Name: iPrimus

User Details User Context: 02\*\*\*\*\*\*\*

allow=alaw&ulaw disallow=all fromdomain=asnjla2.bw.iprimus.net fromuser=AU\_XXXXXXXX host=asnjla2.bw.iprimus.net insecure=very outboundproxy=sbc01a.syd.voip.iprimus.net.au port=15060 secret=YYYYYYYY type=peer username=AU\_XXXXXXXX context=from-trunk type=user

## **Register String:**

AU\_XXXXXXX@asnjla2.bw.iprimus.net:YYYYYYYY:AU\_XXXXXXX@iprimus/02\*\*\*\*\*\*\*

## B.1.7 MyNetFone

#### Peer Details

Trunk Name: MyFonesip

allow=g729&alaw&ulaw authname=091xxxxx canreinvite=no disallow=all dtmfmode=rfc2833 fromuser=091xxxxx host=sip01.mynetfone.com.au insecure=very nat=yes pedantic=no qualify=yes secret=<password> type=friend username=091xxxxx User Details User Context: 091xxxxx

canreinvite=no context=from-trunk fromuser=091xxxxx insecure=very qualify=no secret=<password> type=friend username=091xxxxx

#### **Register String:**

091xxxxx@sip01.mynetfone.com.au:com.au

For further information on MyNetFone, you may refer to the following link: <u>http://www.myfone.com.au/downloads/sip.conf</u> and <u>http://www.myfone.com.au/downloads/extensions.conf</u>

## B.1.8 Nehos (IAX)

# Peer Details

Trunk Name: Nehos

```
allow=g729
disallow=all
host=iax.ifone.com.au
qualify=yes
secret=<password>
type=peer
username=661xxxx
```

#### **Register String:**

661xxxx:<password>@iax.ifone.com.au

## B.1.9 Nodephone

#### **Peer Details**

Trunk Name: Nodephone

allow=g729 canreinvite=no disallow=all dtmfmode=rfc2833 fromdomain=sip.internode.on.net username=<usernumber> fromuser=<usernumber> host=sip.internode.on.net insecure=very secret=<password> type=peer User Details User Context: 661xxxx

context=from-trunk host=iax.ifone.com.au secret=<password> type=user username=661xxxx

User Details User Context: <usernumber>

context=from-trunk host= sip.internode.on.net secret=<password> type=user username=<usernumber>

## **Register String:**

<usernumber>:<password>@sip.internode.on.net/<usernumber>

## B.1.10 SipMe

## **Peer Details**

Trunk Name: sipme

```
authname=1777xxxxx
dtmfmode=rfc2833
fromuser=1777xxxxx
host=sip.sipme.com.au
insecure=very
secret=<password>
type=peer
username=1777xxxxx
```

#### User Details

User Context: 1777xxxxxx

context=from-trunk
fromuser=1777xxxxx
secret=<password>
type=user

#### **Register String:**

1777xxxxxx:<password>@sip.sipme.com.au/1777xxxxxx

# B.2 – New Zealand VSPs (TRUNKS)

## B.2.1 iTalk

Peer Details Trunk Name: italk

allow=ulaw&g729 canreinvite=no context=from-trunk disallow=all dtmfmode=rfc2833 fromuser=64997xxxxx host=akl.italk.co.nz insecure=very secret=<your password> type=friend username=64997xxxxx

#### User Details User Context:

You may prefix outgoing calls with 0197 to disable outgoing CID.

Inbound route 64997xxxxx required to be forwarded to a destination ext/menu

## **Register String:**

64997xxxxx:<your password>@akl.italk.co.nz/64997xxxxx (Source Ref: Steve Biddle)

## B.2.2 Double Zero 64 (IAX)

Peer Details

Trunk Name: 0064

allow=ulaw&g729
disallow=all
dtmfmode=rfc2833
host=voip.0064.co.nz
insecure=very
secret=<your password>
type=peer
username=your\_Customer\_ID

User Details User Context:; Your\_Customer\_ID

context=from-pstn host=voip.0064.co.nz insecure=very secret=PASSWORD type=user username=CUSTOMER\_ID

## **Register String:**

Your\_Customer\_ID: <your password>@voip.0064.co.nz

# B.3 – OTHER VSPs (TRUNKS)

## B.3.1 Broadvoice

Peer Details

Trunk Name: Broadvoice

allow=ulaw canreinvite=no disallow=all dtmfmode=rfc2833 fromdomain=sip.broadvoice.com fromuser=<usernumber> host=sip.broadvoice.com insecure=very nat=yes secret=<password> type=peer username=<usernumber> User Details User Context: <usernumber>

context=from-trunk type=user

## **Register String:**

<usernumber>@sip.broadvoice.com:<password>:<usernumber>@sip.broadvoice.com/<usernumber>

## B.3.2 FWD aka Pulver(IAX)

#### **Peer Details**

Trunk Name: fwd

```
allow=ulaw&alaw
auth=rsa
canredirect=no
disallow=all
host=iax2.fwdnet.net
inkeys=freeworlddialup
insecure=very
secret=<password>
type=peer
username=65xxxx
```

## **User Details**

User Context: 65xxxx

auth=rsa context=from-trunk inkeys=freeworlddialup type=user

**Register String:** 

65xxxx:<password>@iax2.fwdnet.net

## B.3.3 SipBroker

## **Peer Details**

Trunk Name: sipbroker-out

```
allow=g729&ulaw&alaw
canreinvite=no
disallow=all
dtmfmode=rfc2833
fromdomain=<your existing sip provider>
fromuser=<your user ID of existing provider>
host=sipbroker.com
insecure=very
nat=yes
port=5060
secret=<password for existing provider>
type=peer
```

You will not require user details since you will not be receiving incoming calls from this trunk.

You do not need to register either.

If you are using SipBroker, you must ensure that your sip\_nat.conf is modified to the following;

nat=yes
externip=<Fix Ip address> ; if you have fix IP Or
externhost=<your DNS hostname> ; if you are using Dynamic IP
localnet=192.168.1.0/255.255.255.0

## B.3.4 SipPhone

## **Peer Details**

Trunk Name: sipphone

fromdomain=proxy01.sipphone.com
host=proxy01.sipphone.com
insecure=very
secret=<password>
type=peer
username=1747xxxxxxx

User Details User Context: 1747xxxxxx

canreinvite=no context=from-trunk fromuser=1747xxxxxx insecure=very qualify=no secret=<password> type=user username=1747xxxxxx

## **Register String:**

1747xxxxxx:<password>@proxy01.sipphone.com/1747xxxxxxx

## B.3.5 VOIP Buster

#### **Peer Details**

Trunk Name: VoipBuster

```
allow=alaw&ulaw
context=from-pstn
disallow=all
dtmfmode=inband
fromdomain=sipl.voipbuster.com
fromuser=<your username>
host=sipl.voipbuster.com
insecure=very
nat=yes (necessary in some
cases)
qualify=yes
secret=<your password>
srvlookup=yes
type=friend
username=<your username>
```

User Details User Conext: <leave blank>

If you have an inbound number, an Inbound route is required to forwarded your incoming call to a destination extension or IVR menu:

Inbound Route: DID: <User inbound Number> No Fax

#### **Register String:**

<username>:<password>@sip1.voipbuster.com/<username>
(Source Ref: Florent Chandelier)

# **APPENDIX C**

Details for Asterisk Sound Files can be obtained from the following links:

<u>http://www.voip-info.org/tiki-index.php?page=Asterisk+sound+files</u> <u>http://www.voip-info.org/wiki/view/Asterisk+sound+files+additional</u> <u>http://www.nathanpralle.com/software/ast\_masterlist.html</u>

# C.1 VOICE FOR ASTERISK MESSAGES

Brian, a Whirlpool Enthusiast, posted the following tip in the Whirlpool forum. Brian has asked me to include his discovery in this guide.

The following is an extract of Brian's exploits – verbatim ©

<Quote>

I have just been setting up the Digital Receptionist and I wanted to use a professional voice for messages. I thought I might detail my exploits here for those that might be considering the same thing.

First you need the original recording either from a real professional or a "text to voice" utility. I went with the Realspeak demo <u>http://www.nuance.com/realspeak/demo/default.asp</u>

The "British Daniel" voice was deep and professional. Check this out <u>http://www.cbservers.com.au/whirlpool.wav</u>

Now the demo has a 100-character limit so you may need to do 2 or 3 files and join them together with a program like Accoustica MP3 mixer. <u>http://www.acoustica.com/mp3%2Daudio%2Dmixer/</u>

Save it as a wav file.

It was after this part that I had a lot of problems. Asterisk is very funny about sound files. I tried using all sorts of programs to get the file to the right specs but it failed to work every time. I eventually used "**sox**" an open source command line editing tool and it worked first go (using advice from VoIP-info & digium forums) <u>http://sox.sourceforge.net/</u> then at the command line prompt use this syntax:

sox winwave.wav -r 8000 -c 1 -s -w linwave.wav

you may then upload your message to asterisk.

#### Some further reading;

<u>http://lists.digium.com/pipermail/asterisk-users/2002-December/006635.html</u> <u>http://www.voip-info.org/tiki-index.php?page=Convert+WAV+audio+files+for+use+in+Asterisk</u> <u>http://users.pandora.be/Asterisk-PBX/IVR.htm</u> <u>http://www.voip-info.org/tiki-index.php?page=Asterisk+sound+files</u>

<End quote>

# C.2 AUSTRALIAN VOICE AND LANGUAGE SET FOR ASTERISK

As part of OpenVoice contribution to the Asterisk community, they are offering a free set of Australian prompts for the Asterisk<sup>™</sup> Open Source PBX.

The files (male voice) can be downloaded here: <u>http://www.openvoice.com.au/free.php</u>

Of course, don't forget to add the following line in the iax.conf and sip.conf configuration file under [general]. In zapata.conf configuration file the line should be added under [channels].

language=au ; (or fr, it or whichever language is installed)

# **APPENDIX D (ATAS AND ROUTERS)**

## D.1 ALCATEL SPEEDTOUCH 510/530 ADSL ROUTER

If you are using Telstra Alcatel SpeedTouch 510/530 ADSL Router, you will need to unbind port 5060, otherwise you will not be able to use VOIP since Alcatel, in their wisdom for reasons best known to themselves, have decided that we are not suppose to use this port. So they blocked it in these types of routers.

But, as we know, there are more than just one way to skin a cat <sup>(2)</sup> thus, the solution is to telnet to the ADSL modem and remove this restriction.

I am assuming that you are running a Windows PC to do this.

Click on Start > Run then type:

telnet 10.0.138 [or replace with the correct ip address]

you will notice the following;

```
Trying 10.0.0.138...
Connected to speedtouch
Escape character is '^]'.
Username : root
Password :
------....
... welcome message blah blah blah
...
```

You then should unbind the SIP protocol and reboot the modem thus:

```
=>nat unbind application=SIP port=5060
=>config save
=>system reboot
```

To check whether the SIP protocol is bound in your modem use this command to show the list of bindings (only showing SIP here):

```
=>nat bindlist
Application Proto Port
SIP udp 5060
```

If the line doesn't appear, your phone should work.

Source Reference: <u>http://www.sipgate.co.uk/faq/index.php?aktion=artikel&type=faq&rubrik=024&id=398&lang=de</u>

# D.2 BILLION 7402 VL

Some extra information that will get a Billion 7402 VL FXS to work with asterisk and have NAT port 5060 points to the Asterisk server.

By default, if you enable the FXS ports on the Billion 7402 VL router to be sip enabled, the router automatically forwards the port (5060) to its own IP address and an error will occur if you try to forward that port to the Asterisk box. This is because it is intended for the FXS ports to connect directly to an external SIP provider.

To overcome this issue and make the FXS ports connect to the Asterisk box, complete the following:

From a windows command prompt...

telnet 192.168.XXX.XXX (router I.P. address)

voip set interfacename iplan (this changes it from the WAN to the LAN interface

Next, power reset the router.

Now you can add the NAT to point to the asterisk server and the FXS ports will connect to the Asterisk box.

**NOTE**: - Also codec G729 had to be disabled on the FXS ports for calls to work. Not sure why.

(Source Reference: Sean Mahon)

# D.3 LINKSYS PAP2 AND SIPURA SPA2000

The most common Analog Telephone Adapters (ATA) amongst Australian VOIP users are the Linksys PAP2, Sipura 2000 and Sipura SPA3000 (the Sipura SPA3000 was covered in the chapter relating to Interfacing Asterisk to PSTN). The Linksys PAP2 and the Sipura SPA2000 are essentially identical units encased in different boxes and branded differently. The examples presented here apply to both the Linksys PAP2 and Sipura SPA2000.

#### Note:

The set-up described here is for use with ELASTIX and not for connecting the unit directly to the VOIP service provider.

I will describe the **minimum changes** that are required to get these ATAs to work in conjunction with Asterisk.

As supplied, the unit comes with DHCP enabled. It is best to allocate a fix IP address to the unit and therefore the first thing to do is to disable the DHCP and allocate a fix IP address e.g. 192.168.1.50. You need to do the following:

- 1. Connect the unit to your network via the RJ45 port behind the unit.
- 2. Power up the unit
- 3. Plug in a telephone in the RJ11 port 1 behind the unit (any one will do)
- 4. Dial \*\*\*\* and listen to the instruction that you will hear from the IVR
- 5. Dial 101 and when prompted enter 0 followed by # (this will disable DHCP)
- 6. Dial 111 and when prompted enter 192\*168\*1\*50 followed by # (assuming that you want to allocate 192.168.1.50 as the IP address of the ATA).
- 7. Dial 120 and you will hear the current sub-net mask. If it is not your existing network subnet mask you may want to change it. If it is the same as your current network's subnet mask, do don't need to do anything else). If you need to change the sub-net mask, do the following:
- 8. Dial 121 and at the prompt enter 255\*255\*255\*0 (assuming that is your existing sub-net mask)

Next, connect to the Unit using your browser by entering http://192.168.1.50 in your browser address bar and hit enter.

- 2. Select *Admin Login* and then select *(switch to advanced view)*
- 3. Select the **System** Tab
  - a. Under *Internet Connection Type*, change the *Gateway* address to the address of your router.
- 4. Select *Regional* Tab
  - a. Under Vertical Service Activation Codes, delete all the fields.
- 5. Select Line 1 Tab
  - a. Under SIP Setting, change the SIP Port to 5060
  - b. Under *Proxy and Registration*, change the *Proxy* to reflect the address of your Asterisk box.
  - c. Under Subscriber Information, change the *Display Name* to whatever you like, change the *User ID* to the Asterisk Extension number that you have allocated for this line e.g. 201. Change the *Password* to the password you have allocated to the asterisk extension. Change *Use Auth ID* to *Yes*. Change the Auth ID to the extension number you have allocated e.g. 201, the same as the *User ID*.
  - d. Under *Dial Plan*, change the existing dial plan to (\*x./x.) Change *Enable IP Dialling* to *yes*.
- 6. Select Line 2 Tab
  - a. Under SIP Setting, change the SIP Port to 5061
  - b. Under *Proxy and Registration*, change the *Proxy* to reflect the address of your Asterisk box.
  - c. Under Subscriber Information, change the *Display Name* to whatever you like, change the *User ID* to the Asterisk Extension number that you have allocated for this line e.g. 202. Change the *Password* to the password you have allocated to the asterisk extension. Change *Use Auth ID* to *Yes*. Change the *Auth ID* to the extension number you have allocated e.g. 202, the same as the *User ID*.
  - d. Under *Dial Plan*, change the existing dial plan to (\*\*/\*x./x.) Change *Enable IP Dialling* to yes.
- 7. Set this setting only if you are using STUN Server
  - a. Select SIP Tab
  - b. Handle VIA received: no
  - c. Handle VIA rport: no
  - d. Insert VIA received: no
  - e. Insert VIA rport: no
  - f. Substitute VIA Addr: yes
  - g. Send Resp To Src Port: yes
  - h. STUN Enable: yes
  - i. STUN Test Enable: no
  - j. STUN Server: *stun.xten.com* (I use stun.xten.com, you may use whichever stun server you like)
  - k. NAT Keep Alive: **15**

NAT Support Parameters						
Handle VIA received:	no	<b>~</b>	Ha	ndle VIA rport:	no	×
Insert VIA received:	no	<b>~</b>	Ins	sert VIA rport:	no	×
Substitute VIA Addr:	yes 🛛	<b>*</b>	Se	nd Resp To Src Port:	yes	×
STUN Enable:	yes 🛛	<b>*</b>	ST	'UN Test Enable:	no	✓
STUN Server:	stun.xt	en.com	EX	TIP:		
EXT RTP Port Min:			NA	T Keep Alive Intvl:	15	

These are the minimum changes you need to do to get the ATA working in conjunction with Asterisk. There are other changes that you can do depending on your own requirement. There are various discussions on this topic in the <u>Whirlpool</u>, <u>Voxilla</u> and other forums that you can refer to.

# **APPENDIX E – HARDWARE SPECIFIC**

# E.1 RAID

If you have a machine with 2 sata drives or 2 ide drives you can use this ISO to raid your Elastix setup. The ISO is \_base\_d on the new 1.0 stable.

To use the ISO, place into your new PBX and boot the cd. At the boot screen you would have normally just pressed enter to go straight into a normal install. On this ISO you can also do this but it will just be a single HDD install.

## SATA RAID

At the first boot screen write

linux ks=cdrom:/ks\_sataraid.cfg

## IDE RAID

At the first boot screen write

linux ks=cdrom:/ks\_ideraid.cfg

To download the ISO grab it here

http://www.voipeye.com.au/Elastix-1.0-mbit.com.au.iso

## E.2 How TO INSTALL ELASTIX WITH SOFTWARE RAID1

(Submitted by George Tellos)

This is the standard procedure for novice to build a raid1 machine and install any almost any thin distro. First of all you need two disks and they have to be about the same size. Best would be they are identical.

Don't configure IDE RAID mode at BIOS. If you have hardware raid then follow other instructions. For example on an Dell Vostro 200 configure raid at bios, press ctrl+I at boot, press enter 4 times and you will have a nice little hardware raid1. Here we'll do it manually, works on almost anything!

## Cabling

Connect your disks like this:

- 1. You have two ATA disks. Put one on primary master and the other on primary slave. Put your cd-rom on primary or secondary slave. This way you will have SDA and SDC
- 2. You have two ATA disks. Put one on primary master and the other on primary slave. Put your cd-rom anywhere on the secondary channel. This way you will have SDA and SDB
- 3. You have two SATA disks. Put one on channel 0 and the other on channel 0. Put your cd-rom anywhere you like, channel 2 or on any ATA channel. This way you will have SDA and SDB

#### Starting the procedure

Boot from cd-rom, at the welcome screen type "*advanced*' and press enter. What until it asks you how you like to configure partitions on your disks. Of course we'll use all space so forget your data on those disks.

Choose "*Custom layout*" and continue on the following steps. Note if you have partitions already you have to delete them all by hand using the "*delete*" button ...on the screen, not on the keyboard ;) Just highlight them and select delete.

Use *up* and *down* arrow keys to move through disks, *tab* to move through options and *space* to select/toggle

#### To create the /boot partition and configure raid for this

Step 1

Highlight free space under SDA (first disk) using arrow keys and select NEW using tab.

On the new screen select using only *tab* and *space*: *File system Type*: Software Raid *Allowable Drives*: SDA (unselect SDB using space) *Size*: 512 *Force to be a primary partition*: Select this or the system will never boot

#### Step 2

The same procedure for SDB but with <u>one difference</u> Allowable Drives: SDB (unselect SDA)

#### Step 3

Select RAID Mount point. **/boot** File System Type: ext3 Raid Level: RAID1 Raid members: sda1 and sdb1. It's already selected. Leave them alone!

#### To create swap partition and configure raid for this

Step 1

Highlight free space under SDA (first disk) using arrow keys and select NEW using tab.

On the new screen select using only *tab* and *space*: *File system Type*: Software Raid *Allowable Drives*: SDA (unselect SDB using space) *Size*: 2048 *Force to be a primary partition*: DON'T select this!

#### Step2

The same procedure for SDB but with <u>one difference</u> *Allowable Drives*: SDB (unselect SDA)

#### Step 3

Select RAID *Mount point*: nothing! *File System Type*: **swap**  *Raid Level*: RAID1 *Raid members*: sda2 and sdb2. It's already selected. Leave them alone!

#### To create root partition and configure raid for this

Write down the free space of the smallest drive.

Step 1

Highlight free space under SDA (first disk) using arrow keys and select NEW using tab.

On the new screen select using only tab and space: *File system Type*: Software Raid

Allowable Drives: SDA (unselect SDB using space) Size: type the number you wrote down or just toggle the option "Use all available space" if you have identical disks Force to be a primary partition: DON'T select this!

#### Step2

The same procedure for SDB but with <u>one difference</u> *Allowable Drives*: SDB (unselect SDA)

#### Step 3

Select RAID Mount point: I File System Type: ext3 Raid Level: RAID1 Raid members: sda3 and sdb3. It's already selected. Leave them alone!

We're done! Press OK to continue installing the system as usual.

#### Wrap it up

On first boot, login and type: watch cat /proc/mdstat to see the status of the raid.

See if all raid members are *active* and nothing is *unusable* or the machine will blow up! Just kidding.

Finally press *ctrl+c* to exit this screen.

## Some FAQ

Why we mirror the swap partition? No reason. We mirror them all!

Why we use twice as much space than the default setup? Because we have big disks and because free space is never enough!

Will the system format the disks and build the raid on it's own? Yes, don't worry. Just don't be too quick to reboot the first boot.

That's all folks,

# **APPENDIX F** (USERS' CONTRIBUTIONS)

# F.1 ELIMINATING ECHO PROBLEMS IN SPA-3000

Tony, another participant in the Whirlpool Forum, provided the following recommendation.

Echo in the SPA-3000 is a common problem. In reality, most of the time the SPA-3000 isn't causing the echo, it's just making it more noticeable. This is due to the fact that the SPA-3000 passes calls from the PSTN to the LINE1 by converting it to VoIP internally and then backs to analogue. This process does not produce any echo, however is can add about 30ms of latency to the call. This added latency can make echo that was previously unnoticed suddenly become annoying. A pure digital system has no echo (the TX and RX path are 100% separated). It's the interaction of the Digital and Analogue that cause problems.

This may help getting rid of that pesky echo on your Sipura SPA-3000 PSTN line:

1. Make sure you are running the latest firmware (3.1.7) and you have everything back to factory defaults or at least undo all the previous tweaking.

2. Switch \*off\* all echo can in all your devices. There are 6 places in the sipura to switch off echo can.

- PSTN Line -> "Echo Canc Enable",
- "Echo Canc Adaptive Enable",
- "Echo Supp Enable" and
- Line 1 -> "Echo Canc Enable",
- "Echo Canc Adaptive Enable",
- "Echo Supp Enable".

The idea is that we want to hear how bad the echo is with different configs.

3. Unplug everything from your phone line except the SPA-3000. This includes all the extension cables even with nothing connected to them. These can cause impedance problems that lead to echo.

4. Set the Impedance on your lines.

- PSTN -> "Port Impedance" = 220+820||120nF as a starting point.
- Regional -> FXS Port Impedance = "220+820||115nF" as a starting point.

5. In the PSTN tab set -

- "Tip/Ring Voltage Adjust: = 3.1V" and
- "Operational Loop Current Min = 16mA".

Doesn't seem to affect echo, but I believe that these are the correct numbers for Australia.

6. Turn down the jitter buffers!

- "PSTN -> Network Jitter Level: = low",
- "PSTN -> Jitter Buffer Adjustment: = disable".

This reduces the delay across your SPA-3000.

- "LINE1 -> Network Jitter Level: = low",
- "LINE1 -> Jitter Buffer Adjustment: = up and down".

If you are using a poor quality VoIP service as well as the PSTN then you could change the

• "LINE1 -> Network Jitter Level: = medium".

7. Set the preferred codec for the PSTN to be g711a and lock it in.

- "PSTN -> Preferred Codec = g711a",
- "PSTN -> Use Pref Codec Only = yes".

Obviously adjust this if you're accessing your PSTN line via VoIP from a remote network. Set the LINE1 to allow g711a as well as whatever else your prefer.

• "LINE1 -> Use Pref Codec Only = no".

The g711a is fast to encode and decode. Using this codec again reduces your latency and may make the echo less obvious or easier to catch with the echo canceller.

8. Power cycle the SPA-3000 (pull the power plug). Believe it or not, this sometimes fixes the problem. Especially after you have changed the physical phone wiring.

9. Make some test calls. The telco test number 1800801920 is a good one to start with. It has a recorded voice telling you your local phone number. While it's talking, talk back and work out how much echo you are getting. Talk loud, talk soft.

10. Look at what you have got. If you can hear an echo then the problem could be that your probably sending to much power down the line. This is probably reflecting back somewhere as an echo. If you're close to the exchange and have good wires then this is probably the case. You need to crank back on the power. Go to PSTN -> "SPA To PSTN Gain" and turn it down. Be aware that at some point if you turn it down to much, the SPA sorts does a double negative and turns it way up. I believe the range of this variable is about -127 -> 127 (from my testing). Turn it down, down, down until the person can still hear you but reduced echo.

Note: if you enable "Echo Supp Enable" then you will negate these parameters. It seems that the Sipura echo suppression is actually just an automatic gain control. It's really annoying - leave it off.

11. Make a test call to someone with a known good phone out via the SPA's PSTN line or get someone to call in to the PSTN line. Best if it's just a boring old Telstra phone hard wired to a socket on the wall. Don't call a mobile!

If the remote party is hearing echo, it could be that your phone is so loud that it's feeding back into the microphone. Turn down the PSTN To SPA Gain until you can comfortable hear the person, no more. If the remote user can still hear echo, try using a different phone plugged into the SPA. Go for the basics first, a cruddy old Telstra phone is what I use for testing. If this solves the problem you may have a bad phone or an impedance miss match between your phone and the SPA.

- Try changing the Regional -> FXS Port Impedance to "600".
- Try changing the FXO port impedance to "600" or "global".

If this doesn't help, change it back. The impedance will only affect what the \*remote\* party hears, it won't help echo you are hearing.

12. After you have the echo down to a reasonable level, go back into the "PSTN" tab and switch on the "Echo Can Enable = yes". Check to see if the echo has improved. If the echo is tolerable at this level, leave the adaptive echo canceller off. You should have the echo level down to a level that can be stomped on by the echo canceller. If you are using a sip device to talk through your PSTN line, you should probably do all the echo cancellation at that device and leave it switched off in the SPA.

The adaptive echo canceller is a lot more aggressive but also can cancel out some of the incoming conversation. In particular if you're calling in a loud environment then the voice going down the line from your end can trick the echo canceller to start canning some of the real conversation. It makes the incoming party sound a bit scratchy. Leave it off unless you really need it.

The "Echo Supp Enable" switches on automatic gain controls. This means the Sipura will be constantly turning up and down the volume of the call for you and the remote party to try and keep the sound levels at a "good" level. Sipura's definition of good may not suit your situation. The constant variation of the volume annoys me so I recommend leaving it "off".

13. Test call. Talk loud, talk soft. Is your conversation clear at both ends? Should be a lot better. If echo re-appears, at a later date, remember to try power cycling the SPA before you tweak with anything. It may also be at the other end of the call.

In the end it basically boils down to this -

If you are hearing the echo, the real problem is at the \*other\* end of the connection. In a normal phone conversation, the latency is so low that you don't notice it. Your brain automatically tunes out to its own voice when you're talking (as long as the delay between talking and hearing isn't to long). When you insert the SPA, the delay is increased and sometimes the echo can become noticeable.

The only way you can counter this is to reduce the latency (jitter buffer) and/or reduce the gains so the echo volume is reduced below the level that it is noticed. At this point, the SPA-3000 echo canceller should be able to kill off the rest.

## F.2 IMMEDIATE ANSWER OF PSTN CALLS

By default, calls will be answered after about 3 rings thus allowing time for Asterisk to detect the CID. Some users may want the PSTN calls to be answered by Asterisk immediately especially if there is a common phone connected in parallel and to avoid someone answering the call using the parallel phone.

To do this you need to change one setting in the zapata.conf file. By default the setting is set to:

Immediate = no

To force Asterisk to answer immediately, change the line to read:

Immediate = yes

If you are not using Fax, it is also a good idea to disable fax extension in your General setting.

This may or may not work satisfactorily as Asterisk needs a couple of rings to detect CID.

# **APPENDIX G** (PUBLICATIONS AND REFERENCES)

## G.1 DUNDI – A GUIDE TO A SIMPLE SETUP

You will find a guide to a simple DUNDi set-up at the GoldfishNet Blog: <u>http://blog.thegoldfish.net/dundi-tutorial-for-asteriskhome/</u>

## G.2 THE FUTURE OF TELEPHONY BY O'REILLY PUBLISHING

You may download the pdf version of this publication here: http://voipspeak.net/images/stories/orielly/AsteriskTFOT.zip

O'Reilly has licensed the book under the creative commons license and as such, you are free to download and distribute it.

# G.3 OTHER ASTERISK FORUMS, TUTORIALS AND WIKIS

One of my favourite site is <u>http://www.elastixconnection.com.au/</u>

Others that I refer to are:

<u>http://www.voip-info.org/wiki/view/Asterisk%40home+Handbook+Wiki</u> <u>http://sourceforge.net/forum/?group\_id=123387</u> http://sourceforge.net/docman/?group\_id=121515

The following are two tutorial sites that I recommend everyone who is serious about Elastix and Asterisk generally.

<u>http://mundy.org/blog/index.php?p=81</u> <u>http://www.asteriskguru.com/tutorials/</u> <u>http://www.freewebs.com/voipformula/VoIP-HOWTO.html</u>

## **G.4** VOICE OVER **IP** – **P**ER CALL BANDWIDTH CONSUMPTION

The following is a reference from Cisco Systems that's worth a look.

http://www.cisco.com/en/US/tech/tk652/tk698/technologies\_tech\_note09186a0080094ae2.shtml

## G.5 USB PHONE SUPPORT

Most USB Phones are pre-configured for use with Skype and as a result some USB Phone users are experiencing difficulties using the USB phones dial pads to dial out using providers other than Skype especially when using SIP. Special driver are usually required.

The following are links, which may interest USB phone users, where appropriate drivers may be found. Users may have to try different drivers to find a suitable one.

http://www.yealink.com/english/support.htm http://www.packetizer.com/products/ http://www.sednacomputer.com/products/se-p1k.html http://www.welltech.com/product\_e\_06.htm http://www.waytec.com.au/download.htm http://www.pc-telephone.com/free-phone.htm http://www.sjlabs.com/usbphone/SJphoneDriverATCOM-AU-100.exe

# NOTES

## DELETE CALL RECORDS FROM CDR

There will be time when you need to delete the CDR records from your call details record. One example is, those calls you made white testing will need to be deleted before going live where you will need clean record.

There 2 ways which this can be done.

- 1. From the Reports GUI itself
- 2. Doing it directly from the SQL database

## **Deleting CDR Record from Reports GUI**

As illustrated below, do the following from the reports screen:

- 1. Select the date range
- 2. Click on filter
- 3. Click Delete

DR Report Channels U	Usage Dilling st s + 01 Apr 3 s + 09 Apr 3	2008	Field: Destination	×	Filter	Delete	• •
CDR Report Lise Start Date: End Date:	st 8 * 01 Apr : 8 * 09 Apr :	2008	Field: Destination	M	Filtor	Delete	
Start Date: End Date:	n: * 01 Apr : n: * 09 Apr 2	2008	Field: Destination	M	Filtor	Delete	
					2 3		
#_Export					14 Start / Previous	(1 - 9 of 9) Ne:	et 🗈 End 🕅
Date Sou	urce Des	stination	Src. Channel	Código de Cuenta	Dst. Channel	Status	Duration
2008-04-09 18:18:06 2003	01 200	11	SIP/2001-09d2cbf8		Local/0400364780@from-internal-3622,1	NO ANSWER	0
2008-04-09 18:18:08 6032	320565331 s		51P/Pennytel-09d10688			NO ANSWER	0
2008-04-09 18:18:07 6032	320565331 040	0364780	.ocal/0400364780@from-internal-3622,2 SIP/Pennytel-09d10688			NO ANSWER	0
2008-04-09 18:15:34 2003	01 FMP	R-2013	Local/FMPR-2013@from-internal-1d40,2			FAILED	0
2008-04-09 18:15:34 2003	01 FMP	R-2013	Local/FMPR-2013@from-internal-1d40,1			NO ANSWER	0
2008-04-09 18:15:34 2003	01 FMG	SL-0418164640#	Local/FMGL-0418164640#@from-internal-863a,2		Local/0418164640@from-internal-6154,1	ANSWERED	0
2008-04-09 18:15:37 6032	320565331 041	8164640	Local/0418164640@from-internal-6154,2		SIP/Pennytel-09d9b1f8	ANSWERED	0
2008-04-09 18:15:33 2001	01 han	gup	SIP/2001-09d95670		Local/FMGL-0418164640#@from-internal-863a,1	NO ANSWER	0
2008-04-09 18:18:07 2001	01 040	0364780	Local/0400364780@from-internal-3622,1			NO ANSWER	0
					14 Start / Previous	(1 - 9 of 9) Ne:	it in End M

## Delete from MySQL database

Alternatively, you can do it manually;

Login to the shell (SSH) as root. You can do this using Putty.

Enter these commands manually, each on a new line.

```
mysql -p
Enter password: eLaStIx.2007
use asteriskcdrdb;
delete from cdr;
exit;
```

Your CDR will now be empty.

## **EXTENSION NUMBERS TO AVOID USING**

Unless you are prepared to edit and change some codes, its best to avoid the following extension numbers:

200	-	Park Notify
300-399	-	Reserved for speed dial
666	-	Reserved for FAX testing
70-79	-	Reserved for calls on hold
700-799	-	Reserved for calls on hold
7777	-	Reserved extension for incoming calls simulation

## HOW TO RESTRICT OUTGOING CALLS – MANUAL METHOD

This requirement often manifests itself in an office environment where management restricts staff from making International, long distance, or even local calls. In many organizations, staffs are confined to making internal calls only.

There are 2 ways that this can be handled. One is by using the Custom Context module or by creating the necessary context manually.

Here, we will create the procedure manually.

It is somewhat complicated, but I will try to explain it here and provide an example.

- 1. Set up all your trunks and outbound routes as normal. At this stage, users can make unrestricted calls. This is the normal way of doing it with freePBX. You probably would have already done this.
- Through Config Edit of Trixbox©, open extensions\_additional.conf and locate the [outbound-allroutes] section of the file. You will notice something like my example below.

```
[outbound-allroutes]
include => outbound-allroutes-custom
include => outrt-001-PSTN
include => outrt-002-SPA3K
include => outrt-003-Oztell
include => outrt-004-Pennytel
include => outrt-005-Domestic
include => outrt-006-MobileAust
include => outrt-007-International
include => outrt-008-12Number
include => outrt-009-13N1800Numbers
include => outrt-010-e164
include => outrt-011-SingaporeLink
include => outrt-012-KualaLumpurLink
include => outrt-013-MalaccaLink
exten => foo,1,Noop(bar)
```

; end of [outbound-allroutes]

Highlight them, copy and paste in a notepad text file somewhere and will get back to this a little later.

Alternatively you can paste it directly to extensions\_custom.conf. Do not do this if Trixbox© is active as it may cause some problem while people using it.

Through Config Edit of Trixbox©, open extensions\_custom.conf and create a section like my example below. You can get the codes from [from-internal-additional] of extensions\_additional.conf. So will have to do some juggling because you can't cut and paste this from another .conf file without having to close one (you can cut and paste from here if you like).

```
[from-restricted]
; These are all the applications that you will require
;
include => app-cf-busy-off
include => app-cf-busy-off-any
include => app-cf-busy-on
include => app-cf-off
include => app-cf-off-any
include => app-cf-on
include => app-cf-unavailable-off
include => app-cf-unavailable-on
include => app-calltrace
include => app-callwaiting-cwoff
include => app-callwaiting-cwon
include => app-dialvm
include => app-directory
include => app-dnd-off
include => app-dnd-on
include => app-echo-test
include => app-recordings
include => app-speakextennum
include => app-speakingclock
include => app-userlogonoff
include => app-zapbarge
include => app-vmmain
include => ext-group
include => ext-fax
include => ext-meetme
include => ext-findmefollow
include => ext-paging
include => ext-queues
include => ext-test
include => ext-local
include => parkedcalls
; #### OutBound Routes ########
; # Below are all restricted routes #
```

Now copy your [outbound-allroutes] that was saved to the text file directly underneath the last line of the above context. Remember to delete the [outbound-allroutes] label. Unfortunately you can't cut and paste my example because they are my routes NOT yours.

Your final handiwork will look something like the example below:

```
[from-restricted]
;
;
These are all the applications that you will require
;
include => app-cf-busy-off
include => app-cf-busy-off-any
include => app-cf-busy-on
```

```
include => app-cf-off
include => app-cf-off-any
include => app-cf-on
include => app-cf-unavailable-off
include => app-cf-unavailable-on
include => app-calltrace
include => app-callwaiting-cwoff
include => app-callwaiting-cwon
include => app-dialvm
include => app-directory
include => app-dnd-off
include => app-dnd-on
include => app-echo-test
include => app-recordings
include => app-speakextennum
include => app-speakingclock
include => app-userlogonoff
include => app-zapbarge
include => app-vmmain
include => ext-group
include => ext-fax
include => ext-meetme
include => ext-findmefollow
include => ext-paging
include => ext-queues
include => ext-test
include => ext-local
include => parkedcalls
; #### OutBound Routes ########
; # Below are all restricted routes #
; new outbound-restricted-routes
include => outbound-allroutes-custom
;include => outrt-001-PSTN
include => outrt-002-SPA3K
include => outrt-003-Oztell
include => outrt-004-Pennytel
include => outrt-005-Domestic
include => outrt-006-MobileAust
include => outrt-007-International
include => outrt-008-12Number
include => outrt-009-13N1800Numbers
include => outrt-010-e164
include => outrt-011-SingaporeLink
include => outrt-012-KualaLumpurLink
include => outrt-013-MalaccaLink
exten => foo,1,Noop(bar)
; end of outbound-restricted-routes
Now add the following codes underneath that to finish it off.
;
exten => h,1,Hangup
exten => s,1,Macro(hangupcall)
exten => h,1,Macro(hangupcall)
```

4. You may now comment out the outbound routes you do not want the restricted extensions to use.

E.g. If you comment out include => outrt-001-PSTN, the restricted extension cannot use PSTN.

5. After you have done that, you need to go back to FreePBX and edit the extension that you want to restrict by change the context to **from-restricted** instead of from-internal.

Now go enjoy yourself with it.

# **BUGS REPORTS**

Software is not software if it is release without any bug. It just goes against nature's logic, not to mention our Mr. Murphy <sup>(2)</sup>

## **MISSING DIRECTORIES.**

The following directories are missing from Elastix 1.0RC

/var/www/html/recordings /var/www/html/\_asterisk

You can either use WinSCP to create these directories or use Nano.

If you use Nano, do the following:

su asterisk cd /var/www/html mkdir recordings cd /var/www/html mkdir \_asterisk

When you start Elastix or when you do amportal restart and if you see the following errors means that you will need to create those directories.

chown: cannot access `/var/www/html/recordings': No such file or directory chmod: cannot access `/var/www/html/recordings': No such file or directory chown: cannot access `/var/www/html/\_asterisk': No such file or directory.

## **MISSING ENTRIES IN ZAPATA.CONF**

The following entries may be missing from zapata.conf. If they are missing, simply add them.

#include zapata\_additional.conf
#include zapata-channels.conf

## VOICEMAIL CANNOT BE DELETED FROM GUI

As at Ver.1.0-RC2 Voicemail cannot be deleted from the GUI. To delete the voicemail, one must delete it after listening to it on the phone.

## **C**ALL RECORDING NOT FUNCTIONING

As at Ver.1.0-RC2 Call recording will only record external calls only and when it does, the date and time of the recording were wrong. Therefore, no matter how many calls were recorded, you will only be able to see one call and the previous calls are deleted by the system. To overcome this, upgrade to Elastix 1.0 Stable then update again to Elastix 1.0-16

yum update elastix.

## **SYMLINK ERROR**

(Symlink from modules failed)

On the freePBX system status, a Symlink Error is reported.

The steps below will eliminated this error.

```
mv /etc/asterisk/sip.conf /etc/asterisk/sip.conf.old
mv /etc/asterisk/iax.conf /etc/asterisk/iax.conf.old
mv /etc/asterisk/extensions.conf /etc/asterisk/extensions.conf.old
mv /var/lib/asterisk/bin/fax-process.pl /var/lib/asterisk/bin/fax-
process.pl.old
```

/var/lib/asterisk/bin/retrieve\_conf

## ALL PHONES UNUSABLE IF INTERNET CONNECTION IS LOST.

This only happens if users are using **SIP Trunks and/or using Dynamic DNS** service. Luckyly, **those using IAX trunks only are spared from this issue.** 

Some say it is not a bug but to me if users have to make modification to get all the phones to work locally and accept PSTN calls, something is terribly wrong and therefore it is a bug and I think it is a fundamental flaw in the design of Asterisk.

Asterisk is assuming that when a call is being made, the call is going to be made via a trunk **IF** a sip trunk is recorded and enabled. Therefore Asterisk will scan for SIP trunks availability. Once Asterisk finds a trunk it will start sip. If it does not find an available trunk it will give up and start sip then you can make a call. However if you have more than one SIP trunks, it will scan for all enabled SIP trunks until the verification times out. If you also happen to have 10 SIP trunks, all 10 will be verified first before you can even use your phones.

In normal cases where internet connection is available, this will only take a couple of seconds to complete, but if internet connection is out, the verification process will take for ever till it just die away and all your extensions rendered useless.

If no SIP trunk is recorded or enabled at all, then it will start sip immediately and you can make a call. This issue only appears for those using SIP trunks, especially multiple SIP trunks.

This is not a major issue if you only have 1 sip trunk because when Asterisk failed on that trunk, it will give up and start sip. The only thing you notice will be a slight delay between your dialling and the other phone ringing. Normally it is just a slight irritation - nothing more.

The problem starts when you have multiple SIP trunks. If you have 2 SIP trunks, the delay before Asterisk gives up is a little longer and you can still dial internal number after a long delay. In many cases the delay is enough for users to think that nothing is happening, but if you keep waiting it will dial.

However if you have 3 or more trunks, the delay becomes so long for Asterisk to cycle through all your SIP trunks and the phone system becomes unusable, just gave up and dies.

This is where the fundamental flaw is.

Asterisk should dial the number without scanning for SIP trunk (even if you have SIP trunks) unless the number dialed is part of an outbound route that requires SIP trunk.
IAX is spared this hassle why not SIP?

I tested this by adding 1 sip trunk then 2 sip trunks etc. The delay becomes progressively longer and longer the more sip trunks I added.... which brought me to the above conclusion.

Question is; how do we get around this? There is a way but it is a **kludge**. It works and you don't have to fiddle with codes or create a local DNS or BIND and what not that the normal digger would not know how.

This is a project you should try.

**Hint:** Create 2 boxes. Link both boxes using SIP. Make the other box a SIP trunk of your live box. Make sure that this SIP trunk appears as the first trunk in your sip\_additional.conf. Comment out externhost from sip.conf or sip\_nat.conf..

#### VIRTUAL FAX LIST DOES NOT DISPLAY EXISTING VIRTUAL FAX.

An error in one of the lines of */var/www/html/libs/paloSantoFax.class.php* (in line 146) caused this problem.

Change the line from

if(is\_array(\$result))

to

```
if(isset($result))
```

Once this is done, the display will return.

## **CHEAT SHEET**

amportal restart	To restart Elastix after configuration change without rebooting. However certain changes will need rebooting.
asterisk -rvvvv	To get to asterisk CLI
Disable default voicemail message	<ul> <li>This works in custom context e.g.</li> <li>Voicemail(s2000@default) - will not play default message</li> <li>Voicemail(su2000@default) - will not play default message and instead play your unavailable custom message.</li> </ul>
Enable Call waiting by default	Edit ENABLECW=yes in amportal.conf
grep "AUTO FXO" /var/log/messages	Tells you what mode your TDM400 is running on.
/etc/init.d/ntpd stop ntpdate ntp.netspace.net.au /etc/init.d/ntpd start	To set the NTP time manually from the command line.
rpm -qa   grep asterisk	Tells you what version of Aterisk is installed.
rpm -qa   grep zaptel-modules	Tells you what Zaptel modules you have installed
service asterisk restart	To restart asterisk if required when something goes wrong
service network restart	To restart network service if connection goes down and did not come up again even though internet has been restored.
Set System time on system clock	To set the system clock under Linux, you need touse the "date" command. Example: To set the current time and date to May 12, 2007:10:15.30 seconds pm, type ``date 0512 <b>2215</b> 2007. <b>30</b> ' (The time,in <b>bold</b> , is in 24 hour notation). To see what the current local time is, run "date" with no arguments.
core show features	This command run on the Asterisk CLI to tell you features in use.
core show translation	This command runs on the Asterisk CLI to tell you codecs in use.
sip reload	To reload SIP. This command is to be run from Asterisk CLI

#### Some Linux commands in common use

chown <newowner> <filename></filename></newowner>	To change file ownership to the new owner
touch <filename></filename>	To create a blank file using the <filename></filename>

#### **USERS SUGGESTIONS**

# TO RESOLVE THE PROBLEM ON EMBEDED FREE**PBX 2.5.**X ON BLANK EXTENSION FIELDS

Saleh, an Elastix User and a regular on the Elastix forum solved the Extensions GUI problem of displaying blank when freepbx was upgraded to FreePBX 2.5 with the following solution:

The *index.php* which resides in */var/www/html/modules/pbxadmin* needs to be edited as shown below.

# vi /var/www/html/modules/pbxadmin/index.php

and after this line'

\$GLOBALS['recordings\_save\_path'] = "/tmp/";

please add this tow lines

\$GLOBALS['\_guielement\_tabindex'] = 1; \$GLOBALS['\_guielement\_formfields'] = 0;

Reload elastix and the problem will be resolved.

#### **BIBLIOGRAPHY**

The real people behind this guide (Real Names used or aliases if real names unknown): I have also made references to a number of other online publications and forums whenever I need confirmations and further clarifications. Some of the information in this document were sourced from these people and places.

1	The Elastix Team	They keep on getting better
2	Asterisk Guru	http://www.asteriskguru.com
3	Brian	Advise tips and tricks- Voice for Asterisk
		A friend from a long way back. From the days I learned to use a PC and according to Bob, I still can't use it. Bob has
4	Bob Fryer - Voiceintegrity	been helping me putting this document together and providing me with valuable materials and contents. Thanks heap Bob
5	Colin Swan	Major assistance, advice, and basically a place to go when things go wrong.
6	Graham Foote	Always giving me tips and corrections
7	Greg Hind	Advise, Tips and tricks
8	Jack Zimmermann	Helping me with e164
9	Jeffrey Borg	Advise, Tips and tricks
10	lusyn.com	X100P Patch for UK Caller ID <u>http://www.lusyn.com</u>
		Another of my source for help when things go wrong. Mark
11	Mark Brooker - MBIT	is a very competent VoIP consultant. No job is too small or too big for Mark.
12	Matt (daggo)	Advise, Tips and tricks
13	Neilmc	Advise, Tips and tricks
14	Nathan Poyner	Assistance with Scripts and also proofing the documents
15	Openvoice	Source for Australian Voice
16	Peter Quodling	Testbed and resident cynic
17	Rehan	Help with Raid configuration.
18	Rob Thomas	freePBX Developer and Super Hero – without which there will be no Elastix
19	Sammv/sz	Taken time to point out some of my mistakes
20	Sean Mahon	Solution for Billion 7402 BL
21	Shaun ( <i>Ewing</i> )	Advise, Tips and tricks on codecs and dial plan etc
22	Sofoklis Sflomos	Changes and assistance with Raid configuration
23	Stefan Keller-Tuberg	Helping me out with Cron Jobs and Scripts
24	SteveM	Advise, Tips and tricks on Webmin
25	Steven D	Help with detecting some error and proofing the document
26	Thunderbird1/Chippy	Giving me clues to some obscure processes.
27	voip-info.org	A site I often refer to when I require further clarification.
28	Vladimir de la Cruz	Assistance with Custom Contexts
29	Voxilla.com	http://voxilla.com
30	Ward Mundy	My hero <a href="http://mundy.org">http://mundy.org</a> my place of salvation ©.

## CHANGES MADE SINCE LAST UPDATE

Changes made to Muti-Language IVR

To resolve the problem on embedded freePBX 2.5.x "Problem: on extension options all fields are empty"

**Elastix Without Tears** 

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