Elastix without Tears
The ICT serial following
The Elastix ® IPBX Distribution Development

If you find this book helpful, a **PayPal donation** of $10 or more (US equiv) made to bensharif@gmail.com would be
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elastix®
CUSTOM MADE TELEPHONY

Without Tears

http://www.elastixconnection.com/downloads/elastix_without_tears.pdf

For Elastix® v1.x-x with unembedded freePBX® 2.x

(Document Revision 1.6.2a)
ALWAYS A DRAFT
(This is a live document and constantly under revision)

by

Ben Sharif
(The same author who brought you Asterisk@Home without tears and Trixbox® without tears)

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Photography is my passion
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ℰ*ℰ

Stay tuned for our next episode – Elastix version X

My heartfelt 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,
MBIT
Has the answer

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FOREWORD

Following the release of the Elastix® Distribution of the ever-popular Asterisk® 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. **I will outline everything I know in this document. If it is not in this document it means I do not know it, so don't ask me**

Use this document as a guide, not as 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 VoiceIntegrity or MBIT 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 conditions *(because Australia is where I live – those kind enough to fly me around the world are most welcome to send me an air ticket)* through my experience going through the pains and gains, 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 this not so divine revelation).* I invite others to participate in improving this document. Please send me your feedback; so together we can make this document a valuable resource for our peers in the Elastix community, experienced and new alike.   I welcome all assistance and correction for this document.

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 http://www.elastix.org/ 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. Help may also be obtained from Australian Elastix Forum at Elastix Connection

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I INTRODUCTION

Asterisk^{TM} is a Linux based IPBX application developed by Mark Spencer of Digium™, the company behind Asterisk. Elastix^{TM} 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 as stated in I.III.I below, 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 bill from your local Telco.

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 on 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 v1.6), the core PBX (Made by Digium)
- **vTigerCRM**® and **SugarCRM**®, CRM systems
- **A2Billing**® – Calling Card platform and billing application for Asterisk.
- **Flash Operator Panel**, a screen-based operator's console
- **Hylafax**® a software based FAX System
- **Openfire**® - Jabber Compliant Server for Instant messaging, presence management, SIP Phone
- **Conferencing** control application
- **freePBX**® (embedded and standalone) a web User Interface tool for Elastix.
- **A report system** – the part of Elastic (and freePBX) that provides CDR reporting.
- **A Maintenance system**, also part of Elastix, which provides low level interfaces to some components and real time system information
- **OSLEC** - Software Based Echo Cancellation
- **Postfix**®, a well known mail server.
- **Round Cube webmail** – Webmail Interface
- **CentOS**®, a version of Linux related to a very well known Enterprise Linux (but without the branding and support).

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

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.
- A home network with broadband access, 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 PC or an Elastix Appliance to run the IP PBX. If you have chosen the PC route, 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 III clunker you have in the attic.

- 800Mhz Pentium III PC or better (P4 will give extra comfort).
- 312MB RAM – the more the better
- 8GB minimum hard disk space *(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. This is dependant on how many extensions you are planning)*.

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

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.

> 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 Appendix D). http://www.sipgate.co.uk/faq/index.php?aktion=artikel&type=faq&rubrik=024&id=398&lang=de*

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.

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: *http://forums.whirlpool.net.au/forum-replies.cfm?t=370536*

1 INITIAL PREPARATION AND PLANNING

1.1 FAMILIARISATION

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

1.1.1 Default passwords

Interface	Login	Password
Elastix	admin	palosanto
freePBX	admin	admin
FOP	admin	eLaStIx.2oo7
Calling Cards (A2Billing)	admin	mypassword
MySQL	mysql –p (from root)	eLaStIx.2oo7
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 and 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 and MOH
- 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 training purposes only and to assist new hopeful to get started with Elastix and nothing more.*

1.3 PLAN YOUR SECURITY

This part of the setting is most important and cannot be over stated.

1.3.1 Tim Yardley's Securing Trixbox CE

For starter, the document written by **TimYardley** is a must read. While this document refers to Trixbox CE, it applies equally to Elastix. This document on securing your box can be found here:
http://engineertim.com/wp-content/uploads/2008/12/securing_trixbox_ce_ver1.pdf

1.3.2 Seven Easy Steps to Better SIP Security on Asterisk
(by J Todd *http://blogs.digium.com/2009/03/28/sip-security/*)

In the most part, the points below are those recommended by J. Todd but I have taken the liberty of adding a couple of additions to make it in line with the current versions of Elastix.

1) **Don't accept SIP authentication requests from all IP addresses.** Use the "permit=" and "deny=" lines in sip.conf to only allow a reasonable subset of IP addresses to reach each listed extension/user in your sip.conf file. Even if you accept inbound calls from "anywhere" (via [default]) don't let those users reach authenticated elements!

2) **Set "alwaysauthreject=yes" in your sip_general_additional.conf file.** This should have already been set to "yes", if not you will need to set it to "yes". This option has been around for a while (since 1.2?) but the default is "no", which allows extension information leakage. Setting this to "yes" will reject bad authentication requests on valid usernames with the same rejection information as with invalid usernames, denying remote attackers the ability to detect existing extensions with brute-force guessing attacks. While you are editing **sip_general_additional.conf**, add the following line as well **"allowguest=no"** without the quote marks offcourse.

3) **Use STRONG passwords for SIP entities.** This is probably the most important step you can take. Don't just concatenate two words together and suffix it with "1″ - if you've seen how sophisticated the tools are that guess passwords, you'd understand that trivial obfuscation like that is a minor hindrance to a modern CPU. Use symbols, numbers, and a mix of upper and lowercase letters at least 12 digits long.

4) **Block your AMI manager ports.** Use "permit=" and "deny=" lines in manager.conf to reduce inbound connections to known hosts only. Use strong passwords here, again at least 12 characters with a complex mix of symbols, numbers, and letters.

5) **Allow only one or two calls at a time per SIP entity, where possible.** At the worst, limiting your exposure to toll fraud is a wise thing to do. This also limits your exposure when legitimate password holders on your system lose control of their passphrase - writing it on the bottom of the SIP phone, for instance, which I've seen.

6) **Make your SIP usernames different than your extensions.** While it is convenient to have extension "1234″ map to SIP entry "1234″ which is also SIP user "1234″, this is an easy target for attackers to guess SIP authentication names. Use the MAC address of the device, or some sort of combination of a common phrase + extension MD5 hash (example: from a shell prompt, try "md5 -s ThePassword5000″)

7) **Ensure your [default] context is secure**. Don't allow unauthenticated callers to reach any contexts that allow toll calls. Permit only a limited number of active calls through your default context (use the "GROUP" function as a counter.) Prohibit unauthenticated calls entirely (if you don't want them) by setting "allowguest=no" in the [general] part of sip.conf.

These 7 basics will protect most people, but there are certainly other steps you can take that are more complex and reactive. Here is a fail2ban recipe another method from Sunshine Networks can be found here which might allow you to ban endpoints based on volume of requests. There is discussion on the asterisk-user and asterisk-dev mailing lists of incorporating this type of functionality into Asterisk - let's hear your ideas!

In summary: basic security measures will protect you against the vast majority of SIP-based brute-force attacks. Most of the SIP attackers are fools with tools - they are opportunists who see an easy way to defraud people who have not considered the costs of insecure methods. Asterisk has some methods to prevent the most obvious attacks from succeeding at the network level, but the most effective method of protection are the administrative issues of password robustness and username obscurity.

Once you are clear on what you want and what you need to do, 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 has 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. See the illustration below if you are not sure what it is.
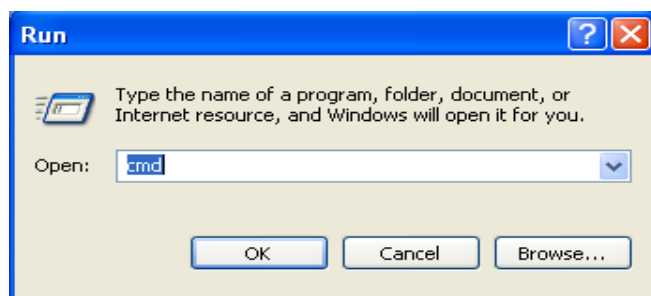
Next a black box will come up. Don't worry if the directory is not the same, as long as it comes up with a prompt.

```
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 Connecti
on
        Physical Address. . . . . . . . . : 00-13-20-B5-5C-9A
        Dhcp Enabled. . . . . . . . . . . : No
        IP Address. . . . . . . . . . . . : 172.22.22.20
        Subnet Mask . . . . . . . . . . . : 255.255.255.0
        Default Gateway . . . . . . . . . : 172.22.22.5
        DNS Servers . . . . . . . . . . . : 172.22.22.2
                                            172.22.22.5

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 SETUP (IPCONFIG for older Centos), 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 **172.22.22.5,** 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 will 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 SETUP (IPCONFIG for older Centos). 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 SETUP (IPCONFIG for older Centos), 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 1.5.2.2. 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
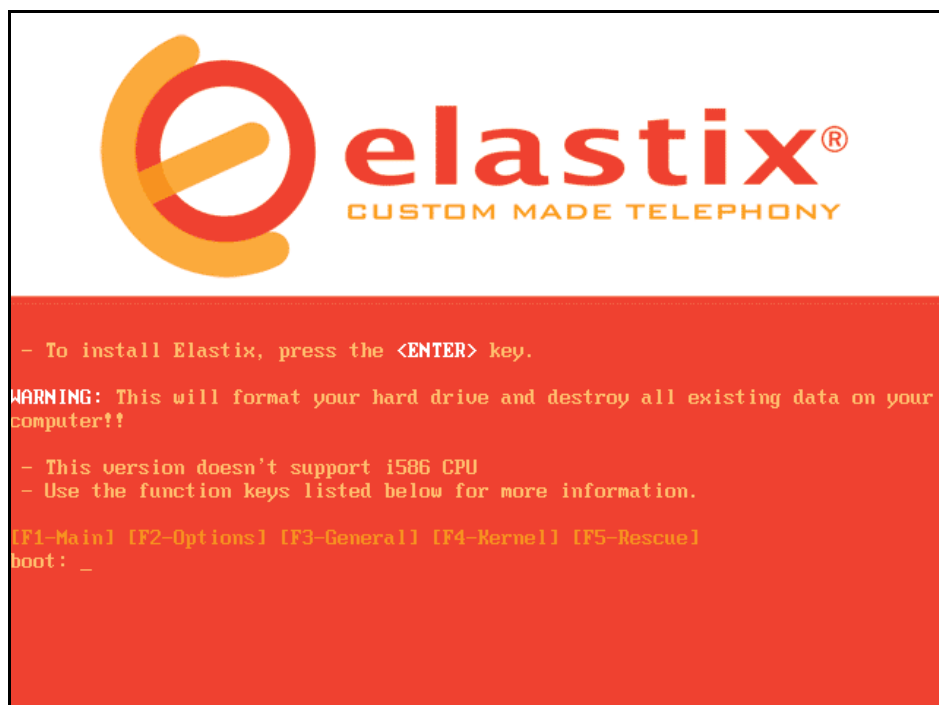
- Press the Enter key to start the installation.

- 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]
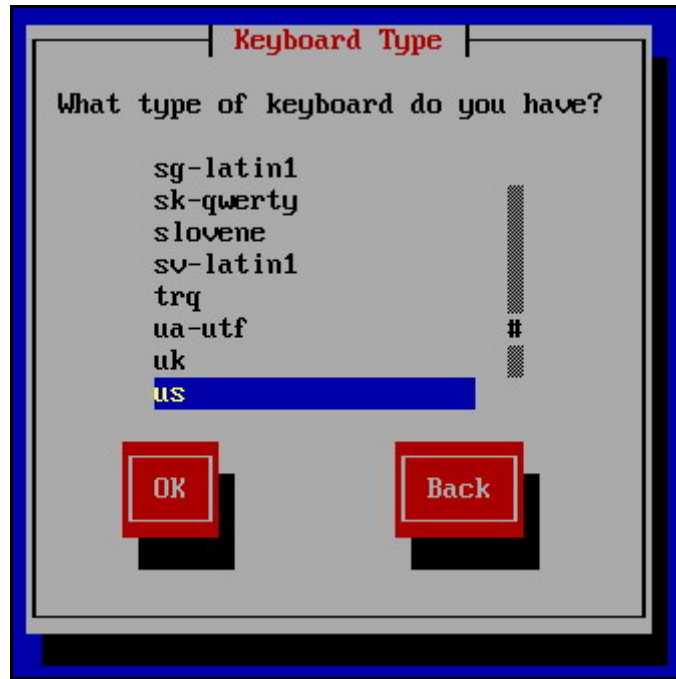
- After system hardware probing is completed, you will be asked to select the Time Zone you are in.
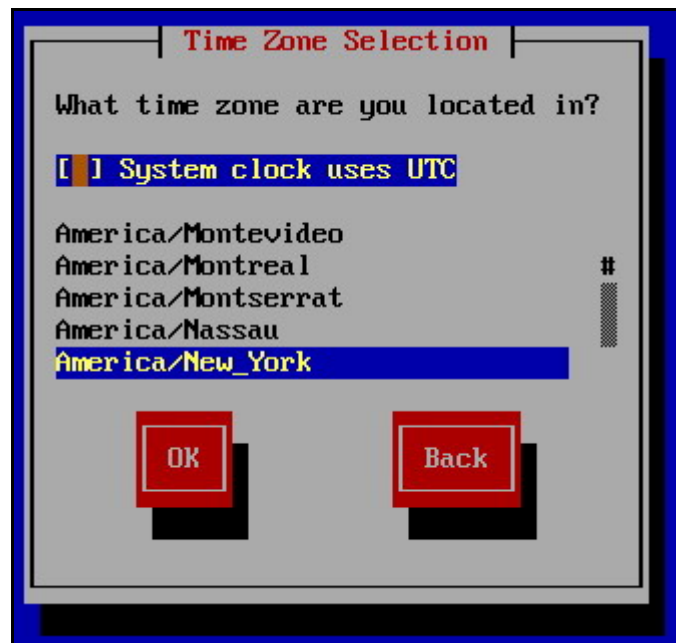
- Use your cursor arrow keys to navigate to the appropriate time zone. In our case, we will select **Australia/NSW**. Use the TAB key to move the focus to **OK** and hit [Enter] to accept (see cheat sheet to set NTP time manually).

- Next you will be asked to enter your root password (remember this password).
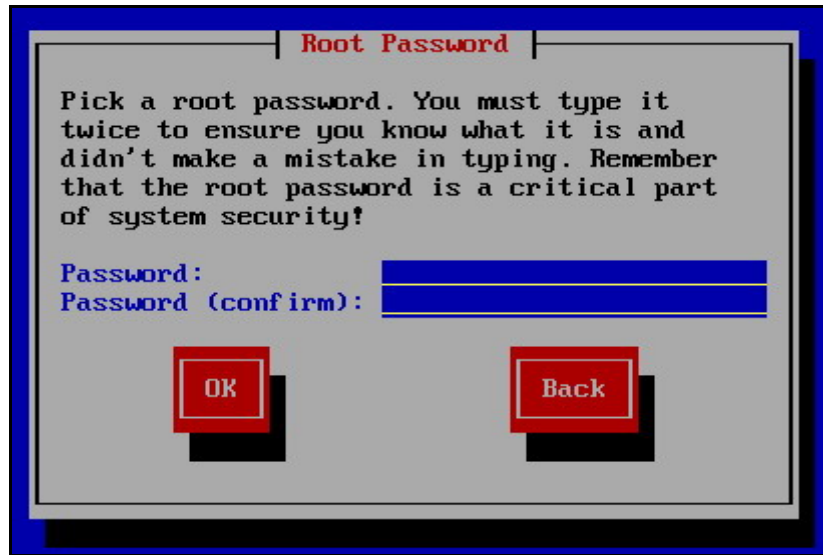
- After entering and confirming your password, installation will now commence by first formatting the Hard Disk/s.

- 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.
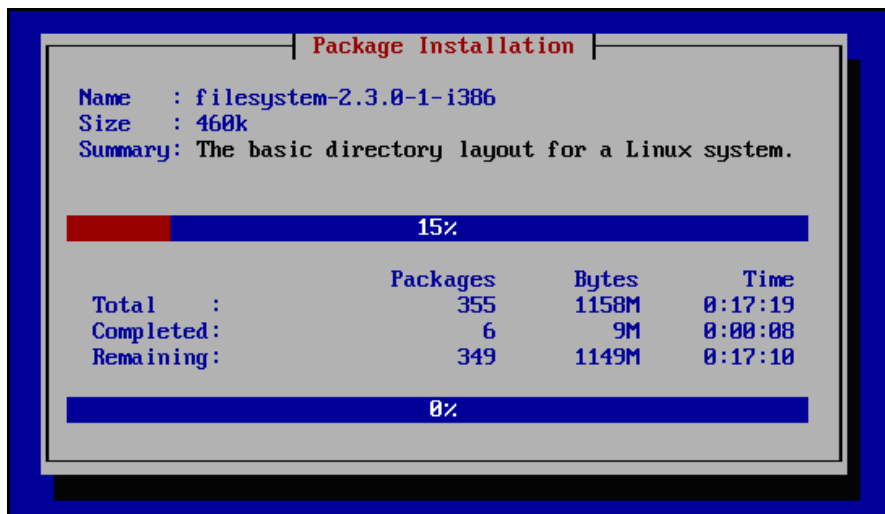
- After Linux is loaded the CD will eject. Take the CD out and wait for the system to reboot.

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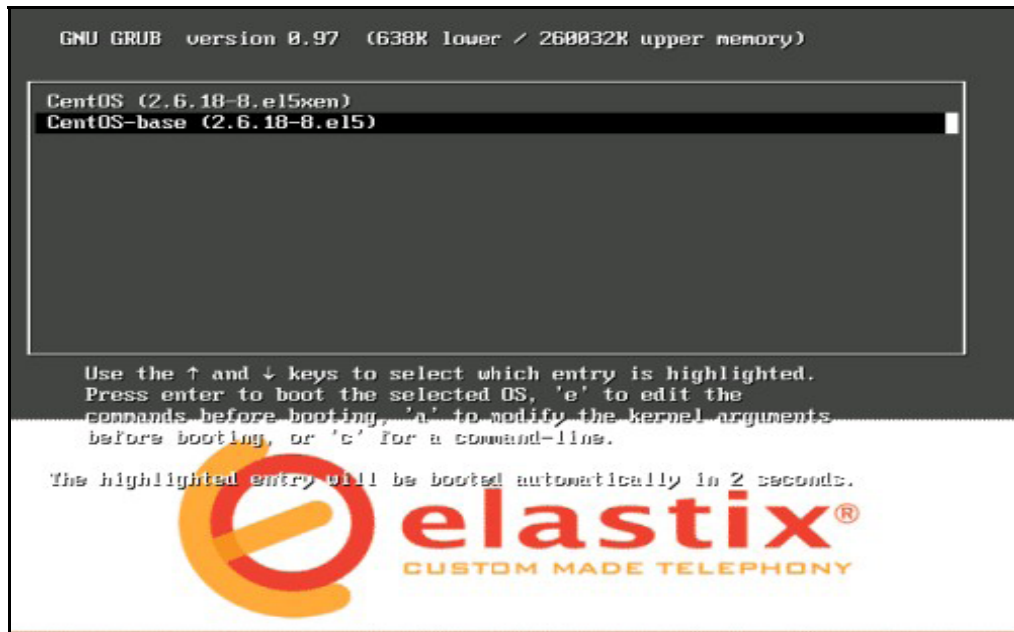
- After going through its initial startup script, Elastix is ready for you to configure and make changes to the system default.

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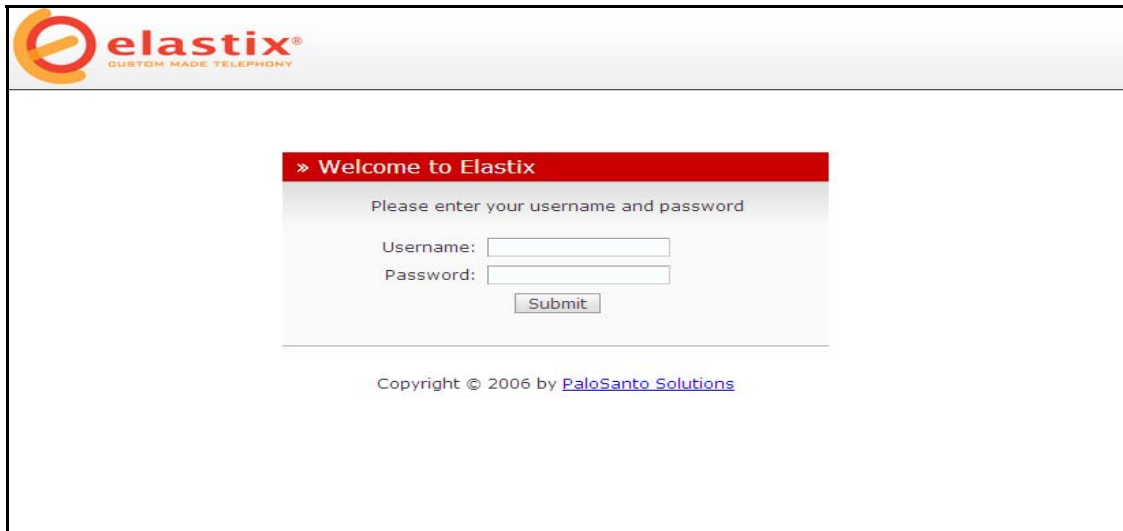
> ⓘ *Elastix uses embedded as well as stand-alone freePBX. To take full advantage of freePBX features, you may want to add extra modules that are not included in the embedded freepbx.*

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.

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.

(i)   *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.
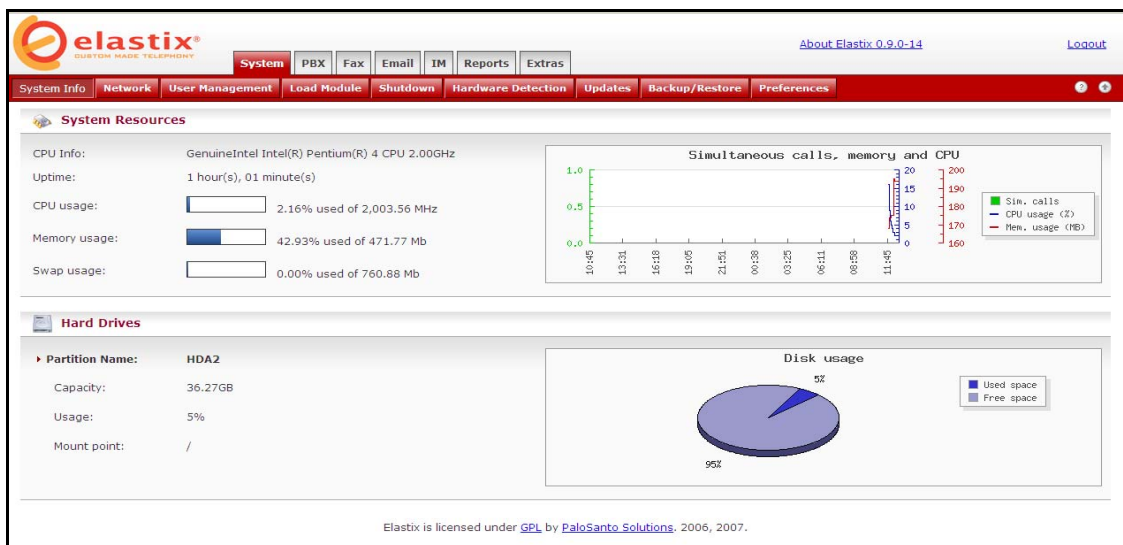
The Admin Dashboard – The Elastix control centre

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.

4.1.1 Change using Elastix GUI

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

The following screen will appear.

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.

4.1.2 Manual method

However if you insist on doing it through the console you may use setup for that purpose.

To change Asterisk IP address from DHCP to Static or vice versa, at the command prompt enter:

`setup`

You will be presented with the following options:

> **Authentication configuration**
> **Firewall configuration**
> **Keyboard configuration**
> **Network configuration**
> **System services**
> **Timezone configuration**
> **X configuration**
>
> **Run Tool**
> **Quit**

In this case you will highlight and select Network configuration and start configuring your IP address.

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.

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

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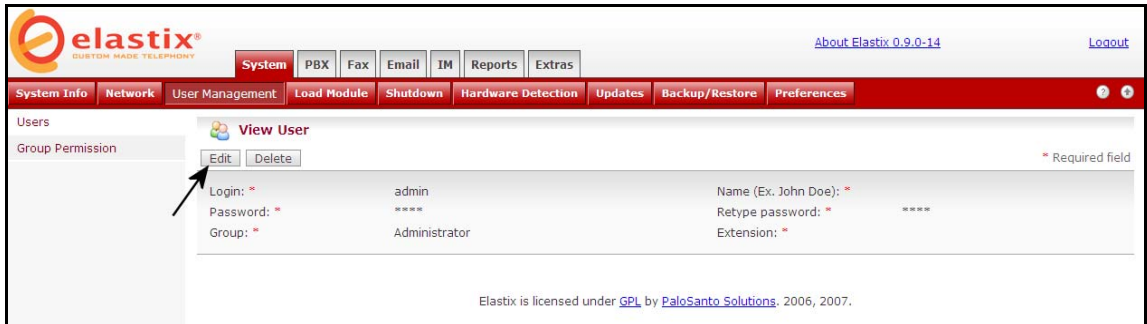
We need to edit the Admin details by changing the default password and other relevant information. At the very least, the default password needs to be changed especially if there is more than one person having access to the system (which is very likely as anyone who knows the box address can get to the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

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

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.
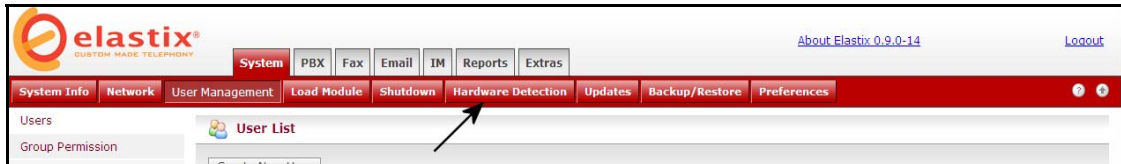
When the following screen appears, click on the Detect New Hardware button. In 99% of the time, your interface cards will be detected. This is the part I love most about Elastix. In my case, my A100P clone card was detected straight away.
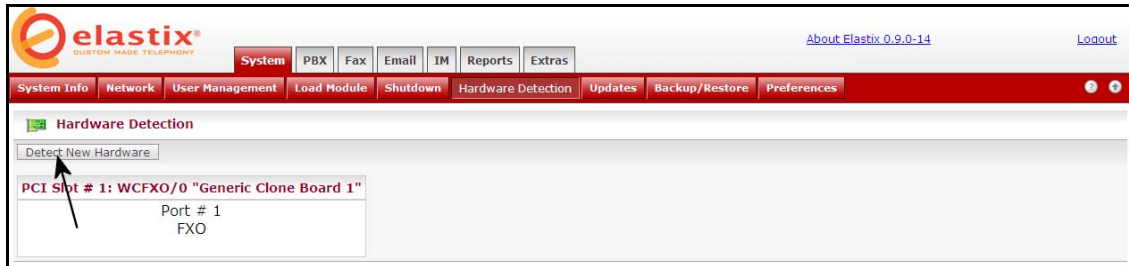
If you are using zaptel instead of DAHDI, all I have to do is add the following 2 lines at the end of zapata.conf

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

(See [Get Under The Bonnet](#) to see how the config file can be edited).

Restart Asterisk

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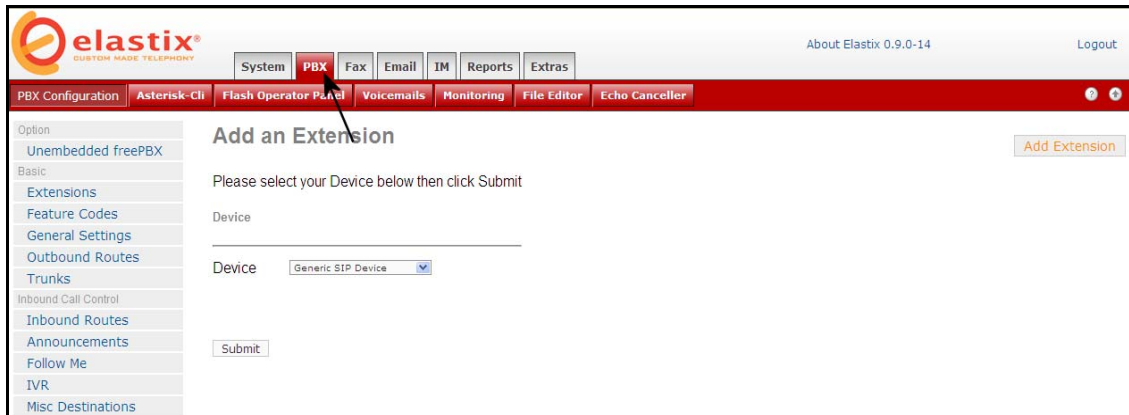
From here, those who are familiar with freePBX will have no difficulties in configuring the PBX portion of Elastix. The choice to do the task using the embedded freePBX or the unembedded version is totally yours. The functionality is similar either way. Naturally not all the modules available for freePBX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

> ⓘ   *To log in to the Unembedded freePBX, use **user**: admin, **password**: admin.*

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.

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

Those fields are as illustrated below.

Country Indications: Australia

Allow Anonymous Inbound SIP Calls? Yes *(if this is set to 'No', all inbound unidentified SIP calls will not be accepted)*. For security, you will have to change this to "no" after you have finished testing your system *(does not work on my freePBX 2.6 but works after I upgraded to freePBX 2.6)*.

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 customise it to your own specific needs such as:

Asterisk Outbound Dial command option:
 "r" which generates 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 values I have as examples above are 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.
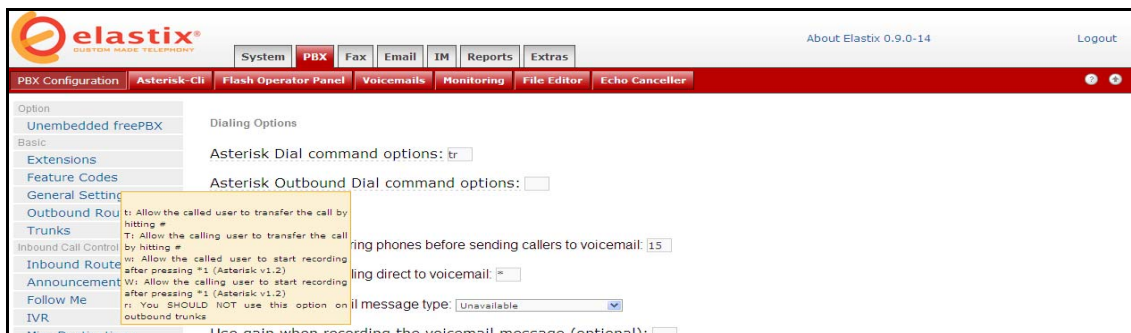
Once done, click on the Submit Change button followed by the "Apply Configuration Changes Here" red bar under the sub-tabs for the changes to take effect.
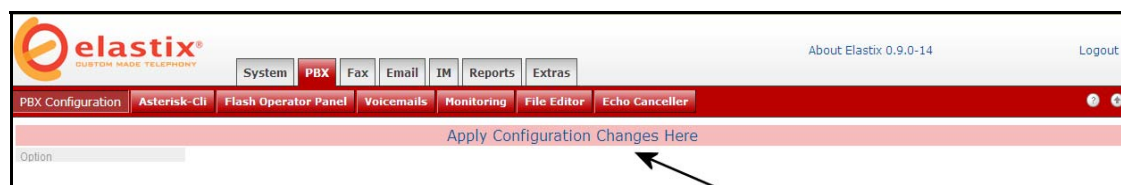
5.1.1 Dial Command Options

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 them alone:

Options:

A(x)	Play an announcement to the called party, using 'x' as the file
C	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.
H	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])	Limit the call to 'x' ms. Play a warning when 'y' ms are left. Repeat the warning every 'z' ms. The following special variables can be used with this option: • LIMIT_PLAYAUDIO_CALLER yes|no (default yes) - Play sounds to the caller. • LIMIT_PLAYAUDIO_CALLEE yes|no - Play sounds to the callee. • LIMIT_TIMEOUT_FILE File to play when time is up. • LIMIT_CONNECT_FILE File to play when call begins. • LIMIT_WARNING_FILE File to play as warning if 'y' is defined. The default is to say the time remaining.
m([class])	Provide hold music to the calling party until a requested channel answers. A specific MusicOnHold

	class can be specified.
M(x[^arg])	Execute the Macro for the *called* channel before connecting to the calling channel. Arguments can be specified to the Macro using '^' as a delimiter. The Macro can set the variable MACRO_RESULT to specify the following actions after the Macro is finished executing • ABORT Hangup both legs of the call. • CONGESTION Behave as if line congestion was encountered. • BUSY Behave as if a busy signal was encountered. This will also have the application jump to priority n+101 if the 'j' option is set. • CONTINUE Hangup the called party and allow the calling party to continue dialplan execution at the next priority. • GOTO:<context>^<exten>^<priority> - Transfer the call to the specified priority. Optionally, an extension, or extension and priority can be specified.
n	This option is a modifier for the screen/privacy mode. 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 if callerID is present, do not screen the call.
o	Specify that the CallerID that was present on the *calling* channel be set as the CallerID on the *called* channel. This was the behaviour of Asterisk 1.0 and earlier.
p	This option enables screening mode. This is basically Privacy mode without memory.
P([x])	Enable privacy mode. Use 'x' as the family/key in the database if it is provided. The current extension is used if a database family/key is not specified.
r	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 party has answered the call.
t	Allow the called party to transfer the calling party by sending the DTMF sequence defined in features.conf.
T	Allow the calling party to transfer the called party by sending the DTMF sequence defined in features.conf.
w	Allow the called party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in features.conf (default *1 in Asterisk v1.2)
W	Allow the calling party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in features.conf (default *1 in Asterisk 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:

70-79	-	Reserved for calls on hold (Definable)
700-799	-	Reserved for calls on hold (Definable)
7777	-	Reserved extension for incoming calls simulation
911	-	Emergency number
999	-	Emergency number in some countries

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:

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".
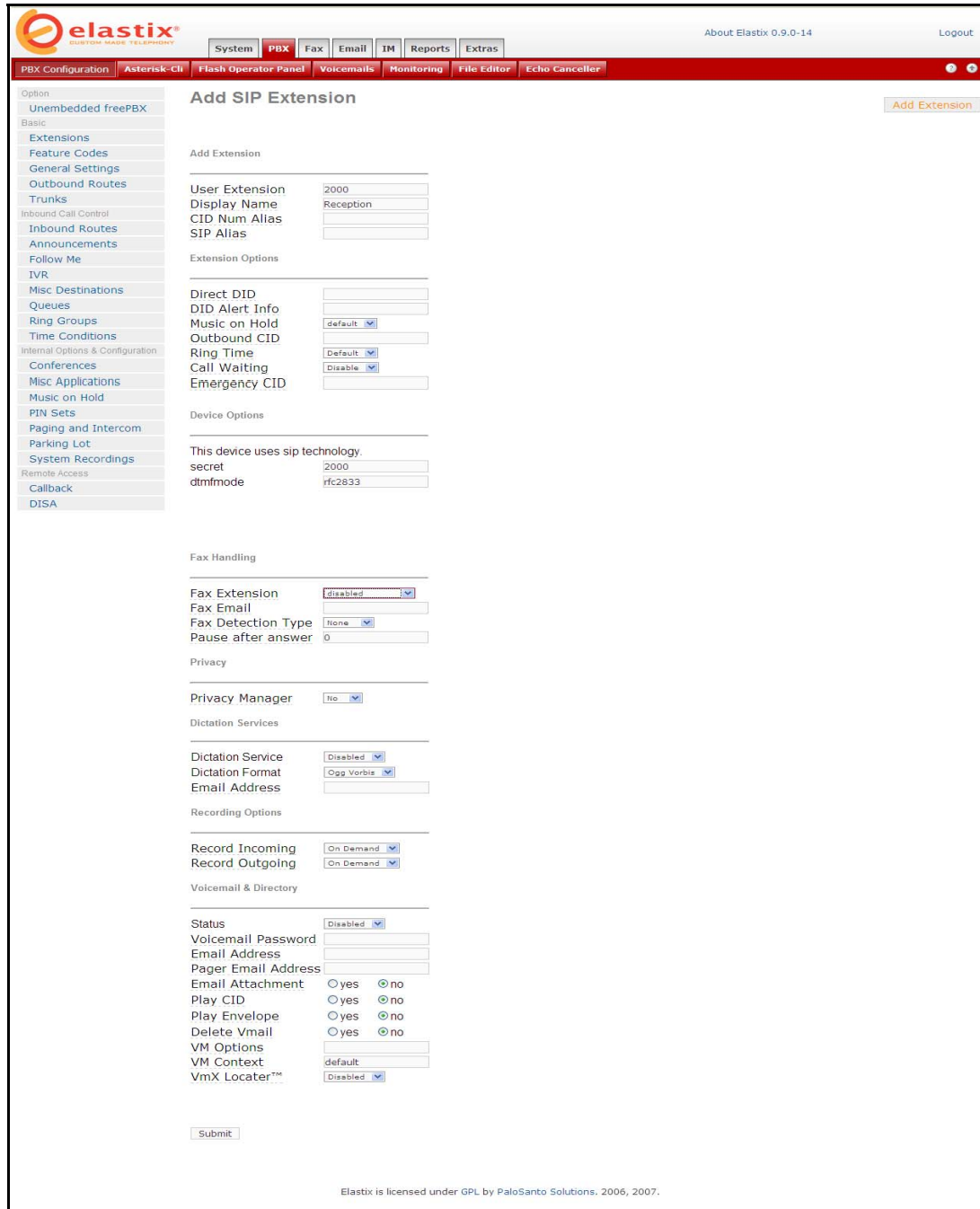
The illustration below is where you create the extension.

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)
- Disable 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.

<u>Add Extension Screen</u>

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

If you have 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. See Users' Suggestions.
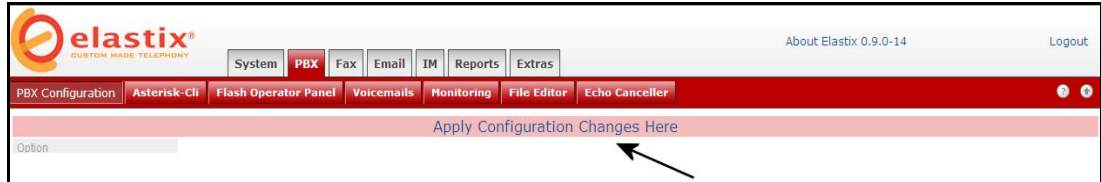
Click on the Add Extension button to add more extensions.

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 one way audio. This also affects any Fax machines attached to the FXS ports; you will not receive faxes at all unless you do as below

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)

> ⓘ *As at the time of writing this document, it is 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 because those extensions were not properly written to the said conf file.*

As part of Elastix 1.x, 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 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

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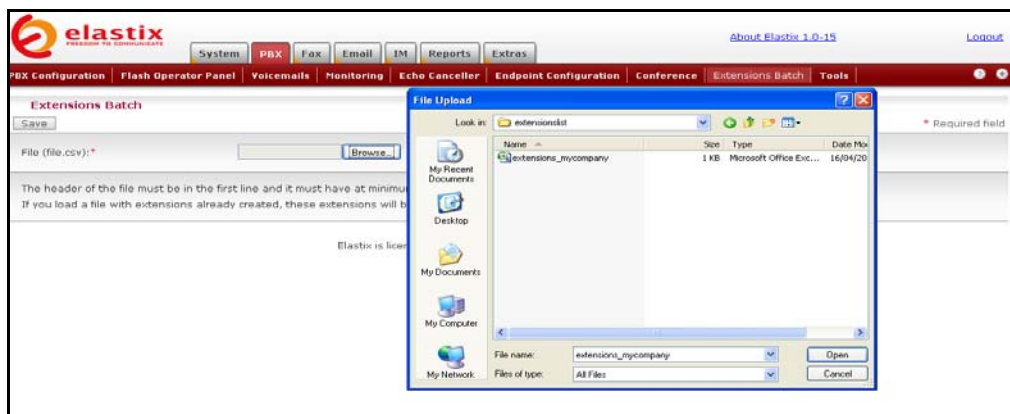
Hopefully it should default the directory where you saved your extensions to, select the Extensions file, and click on **Open** and you should return to the following screen.

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!!
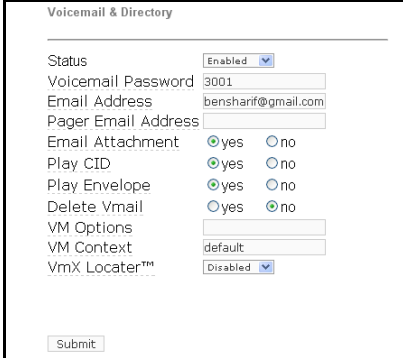
5.2.4 Enable Voicemail

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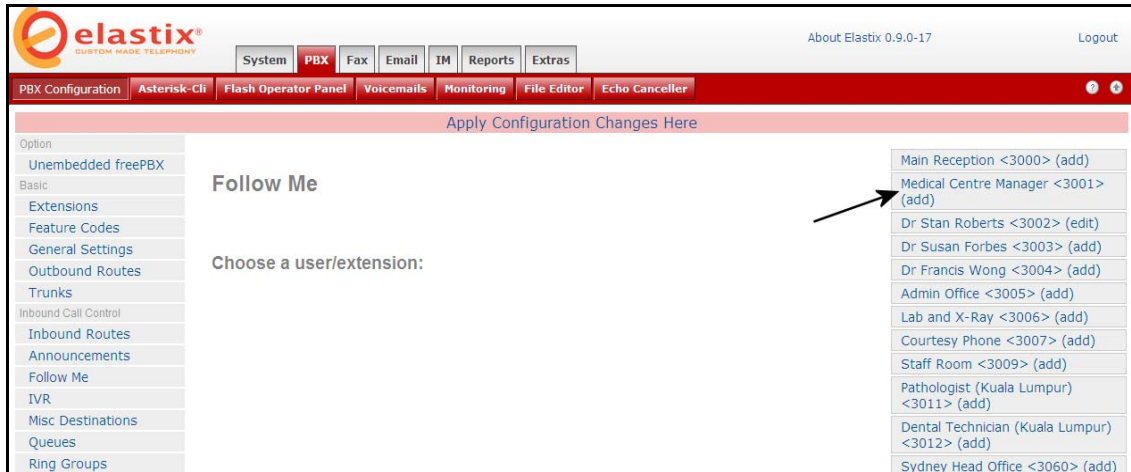
Click submit when done followed by the red bar on top of the screen.

5.3 FOLLOW ME

After setting up the extensions, you need to decide if you want Elastix to call another pre-arranged extension, if the extensions called do not answer. This is where we will do it as per the illustration below:
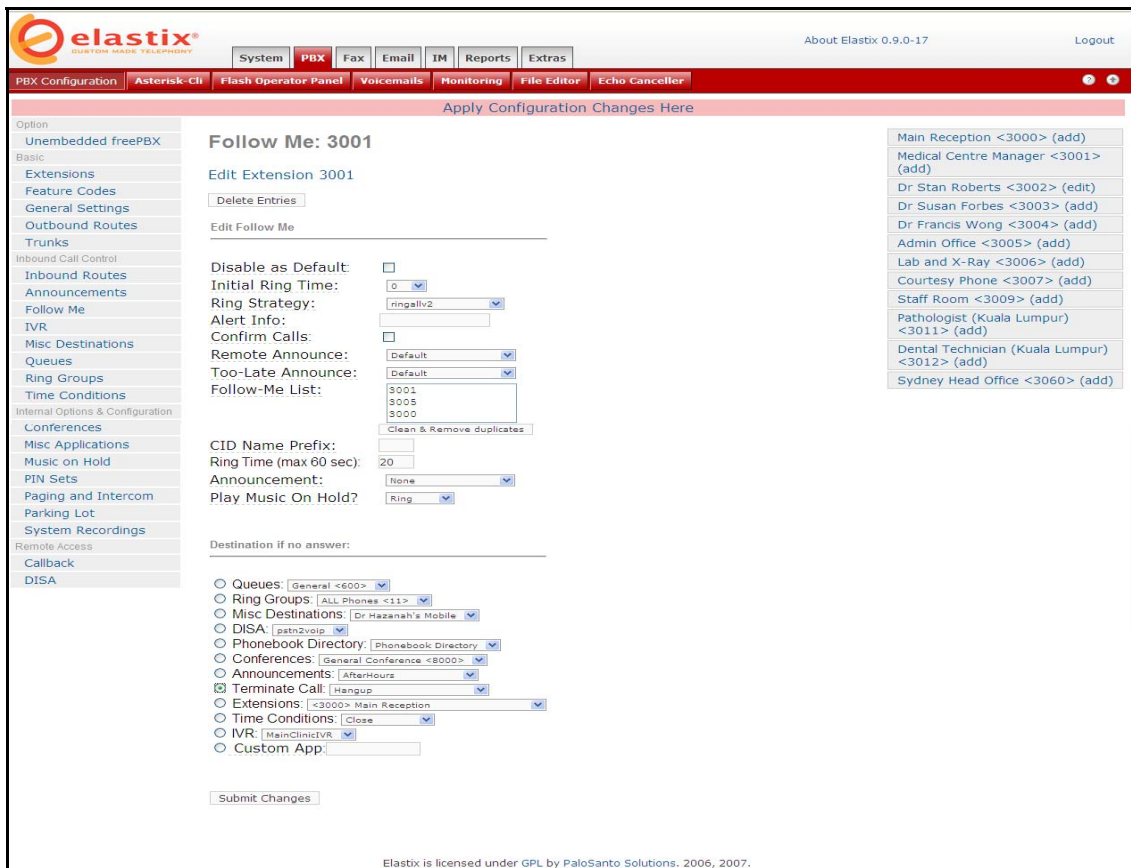
Select the *PBX -> PBX Configuration -> Follow Me*.

You will be presented with the following screen:

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'**

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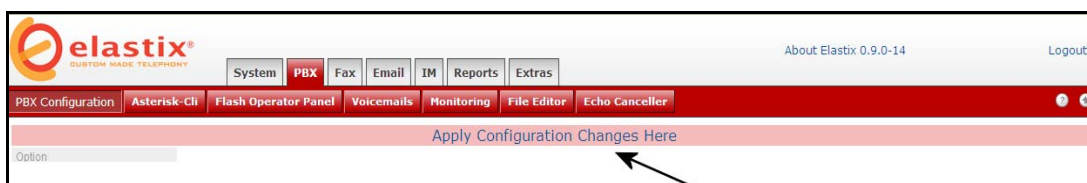
Click "submit" followed by clicking the red bar on top of the screen to finalise this selection.

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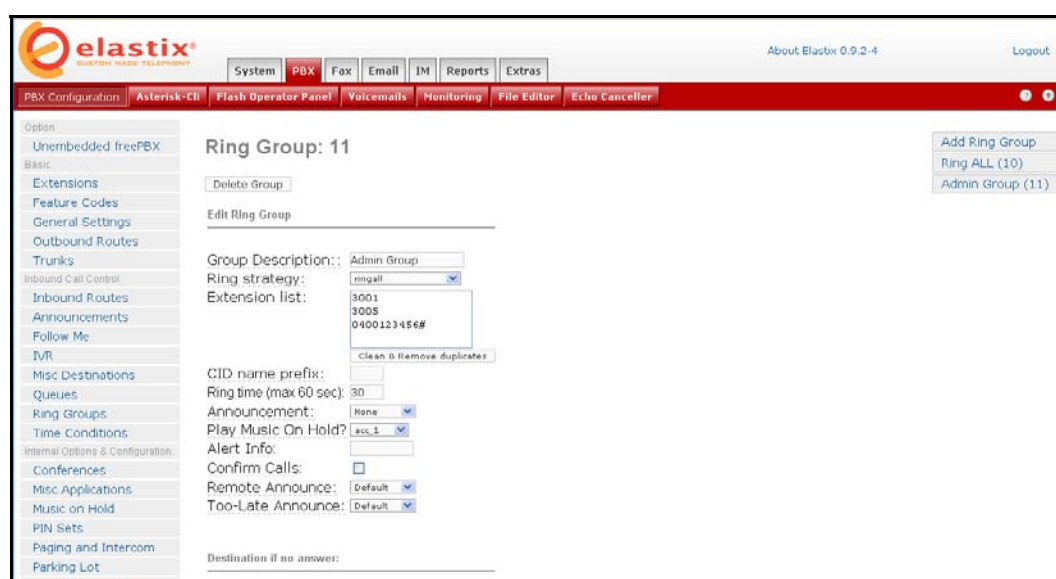
When there is an incoming call to the ring group, the phones nominated in the selected group will ring. You may select different ring group for each of the incoming trunk or you may nominate the same group for all the trunks, in which case you will only need to define only one ring group.

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:

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 sofphone.

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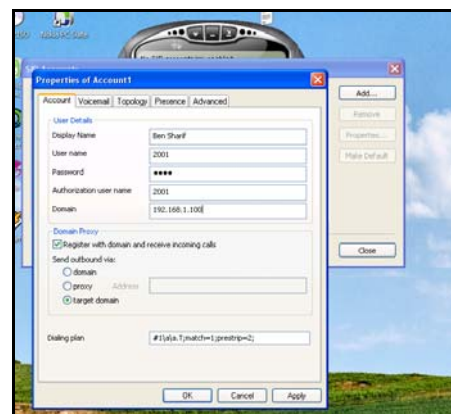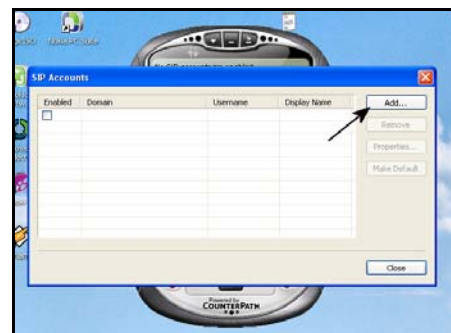
Sometime it is necessary to change the Honor setting in X-Lite for it to work correctly. To do this, you would need to go to the advance setting of X-lite.

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 /sipphone/BOL%20SIPPhone_EN.msi*

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).

Domain/Realm: <leave it blank>
Proxy: Your Asterisk network address e.g. 192.168.1.100
Port: 5060
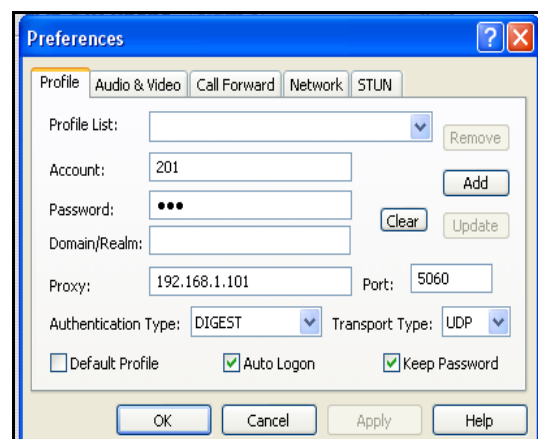
Check the Auto Login and Keep Password.

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
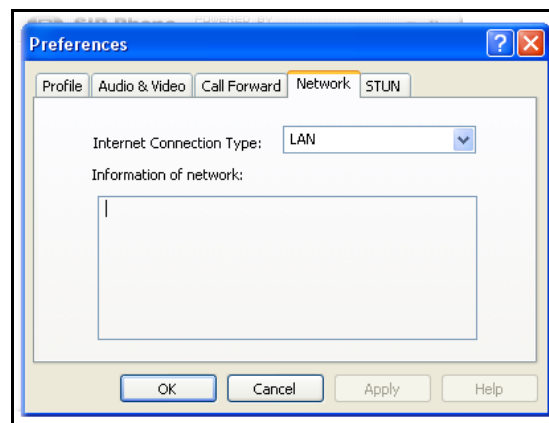
STUN Tab

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 **STUN Servers** later in this document.
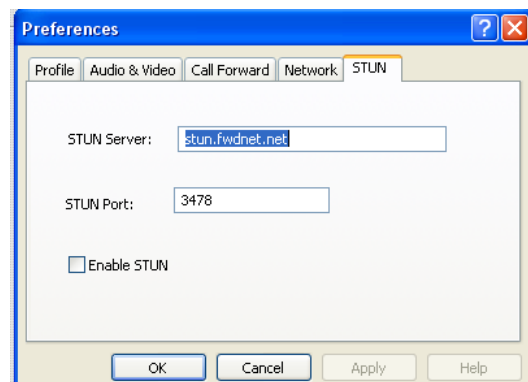
In my case, I do not use STUN and therefore I left that section blank and Enable STUN check box un-ticked.

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.

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.

elastix
FREEDOM TO COMMUNICATE

System | PBX | Fax | Email | IM | Reports | Extras

About Elastix 1.0-14   Logout

PBX Configuration | Flash Operator Panel | Voicemails | Monitoring | Echo Canceller | Endpoint Configuration | Conference | Extensions Batch | Tools

Endpoint Configuration

Endpoint Scan 172.22.22.0/2

Start / Previous (1 - 2 of 2) Next / End

Set	Unset	MAC Adress	IP Adress	Vendor	Phone Type	User Extension	Status
☐		00:0B:82:0D:9E:79	172.22.22.31	Grandstream / Grandstream Networks	-- Unselected --	-- Unselected --	Not Set
☐		00:0E:08:DE:1A:3C	172.22.22.45	Linksys / Sipura Technology	-- Unselected --	-- Unselected --	Not Set

Start / Previous (1 - 2 of 2) Next / End

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These are the VoIP phones identified during the scan. 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.

elastix
FREEDOM TO COMMUNICATE

System | PBX | Fax | Email | IM | Reports | Extras

About Elastix 1.0-14   Logout

PBX Configuration | Flash Operator Panel | Voicemails | Monitoring | Echo Canceller | Endpoint Configuration | Conference | Extensions Batch | Tools

Endpoint Configuration

Endpoint Scan 172.22.22.0/2

Start / Previous (1 - 2 of 2) Next / End

Set	Unset	MAC Adress	IP Adress	Vendor	Phone Type	User Extension	Status
☐		00:0B:82:0D:9E:79	172.22.22.31	Grandstream / Grandstream Networks	GXP2000	Varity <202>	Not Set
☐		00:0E:08:DE:1A:3C	172.22.22.45	Linksys / Sipura Technology	SPA942	Bob <201>	Not Set

Start / Previous (1 - 2 of 2) Next / End

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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.

Endpoint Configuration

Endpoint Scan 172.22.22.0/2

Start / Previous (1 - 2 of 2) Next / End

Set	Unset	MAC Adress	IP Adress	Vendor	Phone Type	User Extension	Status
☐	☐	00:0B:82:0D:9E:79	172.22.22.31	Grandstream / Grandstream Networks	GXP2000	Varity <202>	Configured without incident.
☐	☐	00:0E:08:DE:1A:3C	172.22.22.45	Linksys / Sipura Technology	SPA942	Bob <201>	Configured without incident.

Start / Previous (1 - 2 of 2) Next / End

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.|xx.***)*

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 CIRCLE 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 terms.

Each of these two work 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 needs 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|NXXXXXXXX`

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 What is a Dial Pattern in an earlier chapter.

8.4 LET'S CREATE THE VOIP TRUNKS

To create a new trunk, select the **Trunks** option from the vertical menu on the left.
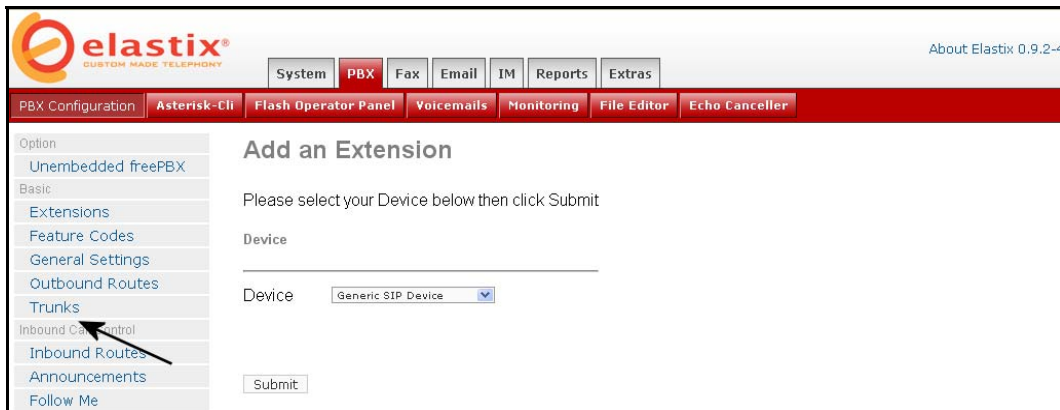
For this example, let's create 2 **SIP** trunks; Pennytel and Oztell.

Pennytel 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.
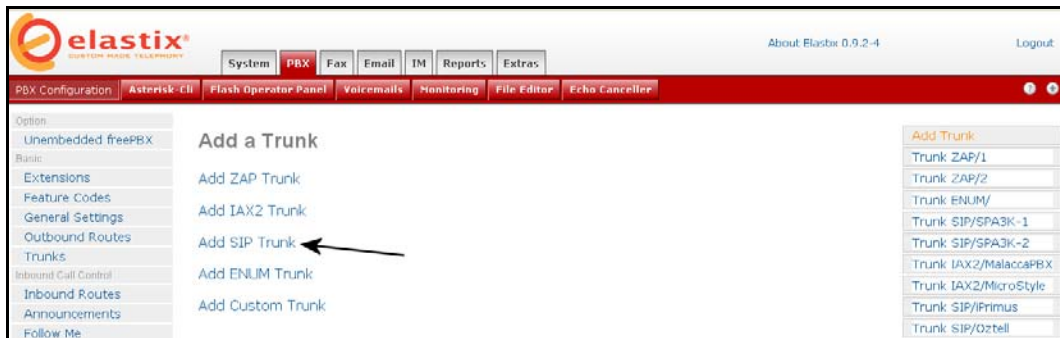
General Setting (This applies to all the trunks):

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

Maximum channels: Leave blank for maximum.

In this example we will create 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 international and mobile calls only.

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 + area code +phone number*. E.g. to call *Kuala Lumpur in Malaysia is 0011+64+3+<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, these 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
601XXXXXXXX	This pattern tells Pennytel to dial all Malaysian Mobile numbers starts with 601 followed by 8 more digits.
60NXXXXXXX	This pattern dials numbers start with 60 followed by a one-digit number in the range of 2-9 followed by 7 more digits.
61+4XXXXXXXX	For dialling Australian Mobile phone. The 61 is the Australian Country 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

```
username=8880XXXX
type=peer
secret= Password
insecure=very
host=sip.pennytel.com
dtmfmode=rfc2833
disallow=all
allow=alaw&ulaw&gsm
canredirect=no
canreinvite=no
```

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= Password
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_general_custom.conf.

```
useragent = Elastix
```

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

My *sip_general_custom.conf* looks like this;

```
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

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+NXXXXXX	Using the Oztell 06 rate to dial any NSW number.
0661+NXXXXXXX	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+NXXXXXX
0661+NXXXXXXX
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.

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

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.

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

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:

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

*** For more trunk settings, please refer to APPENDIX_B**

Now that you know what the trunk will do, you may proceed to the next chapter 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.

You will then be presented with the following screen.
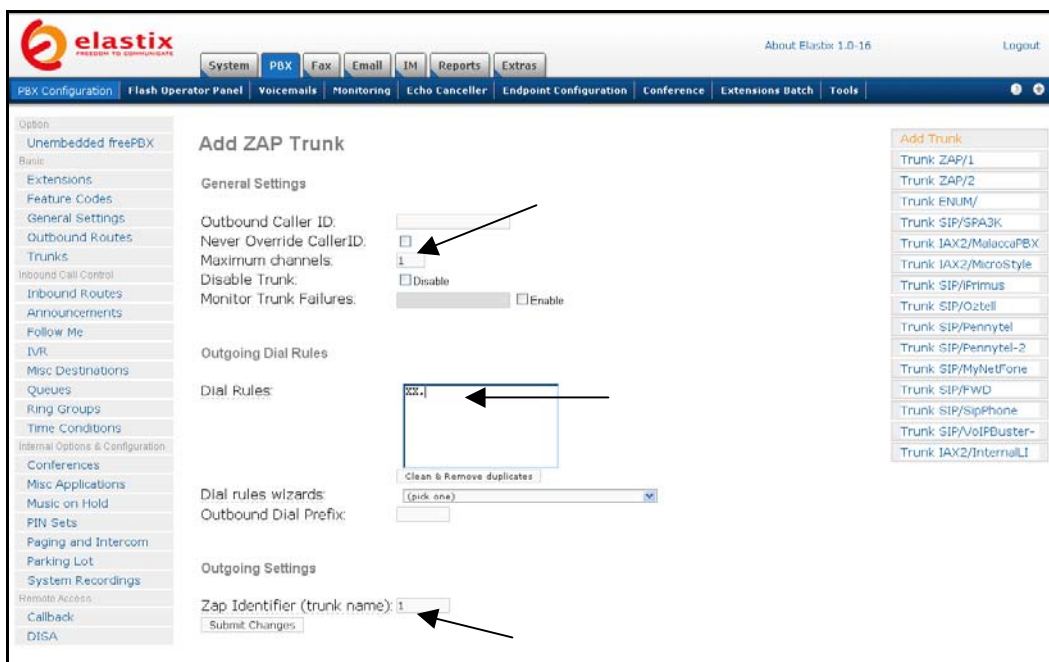
For simplicity, simply do the following in the related fields:

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 confirmation bar on top and you are done.

9 CALL 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 Routes 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.

If you are using freePBX v2.4 or later, 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 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).

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* and later, 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.

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 (upgrading from earlier version will show the zap channels as *zapchan1* or *zapchan2* etc. which have been depreciated).

9.2 TIME CONDITIONS

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.

From freePBX 2.5 onwards, the time condition needs to be associated with a time group. You can no longer define the time in the Time Condition screen. The initial Time Group needs to be defined in freePBX as Elastix do not have that option on the initial GUI.

Alternatively you may create the time condition as illustrated above without the time group and submit the change. Once the changes have been submitted, a link will appear enabling you to go to the time group (see illustration below).

Clicking on the link will take you to a screen where you will be able to create the time group that you need. While you are at it, you may as well create all the time groups that you need (see illustration below).

You may create several Time Groups and give each of the Time Group a Short Name to identify it. These time groups can then be assigned to each individual time condition that you have created (or will create) and use for the Inbound Route if you choose to do so.

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.

- Create a *Time Group*:
    - o   Name the Time Group DayNight
    - o   Time to start: 08:00
    - o   Time to finish: 20:00
    - o   Week Day start: Monday
    - o   Week Day finish: Sunday
    - o   Submit the changes
- 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** (same as the time group for simplicity)
- Select DayNight for your time group
- 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 to 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. Create an OfficeMorning time condition.
    - Create an **OfficeMorning** Time Group:
        - o Time to start: 08:00
        - o Time to finish: 13:00
        - o Week Day start: Monday
        - o Week Day finish: Friday
    - 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 Group*, select *OfficeMorning*
    - 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
    - Create an **OfficeAfternoon** Time Group
        - o Time to start: 14:00
        - o Time to finish: 17:30
        - o Week Day start: Monday
        - o Week Day finish: Friday
    - 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 Group,* select *OfficeAfternoon*
    - 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.

> (i) *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 4 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.
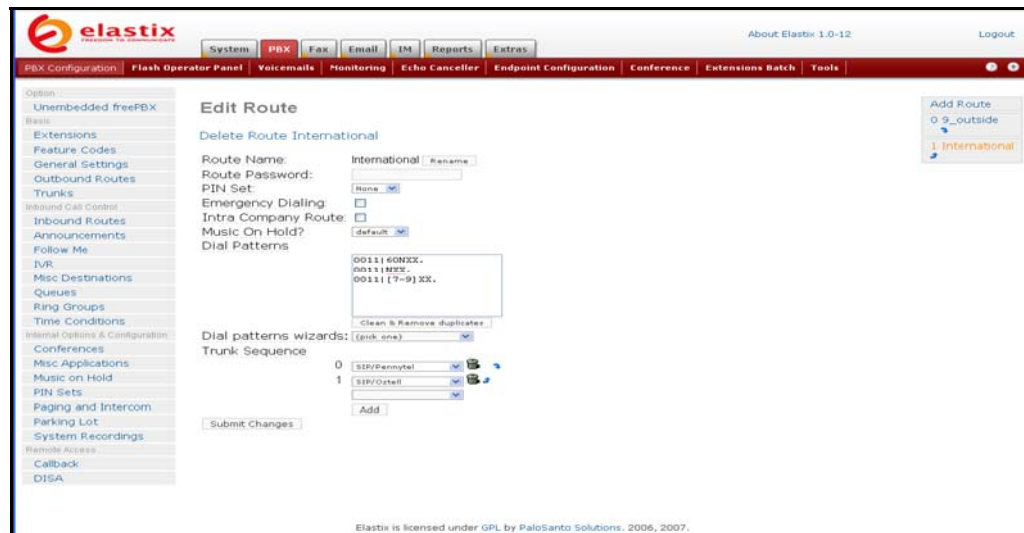
Click on the Add Route in the menu on the right of the screen.

9.3.3 International
You will need the following patterns 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]XXXXXXX	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 mobiles – illustrated below.

Dial Patterns

```
0|4XXXXXXXX
```

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]NXXXXXXX	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.
2. 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+NXXXXXXX (for local calls without having to dial the 02)
4. On the second line enter 61+NXXXXXXXX (for dialling Australian numbers where you need to dial the prefix (eg. 03)
5. On the next line enter ZXXXXXXXX. (Including the dot) for dialling International numbers – see example below:

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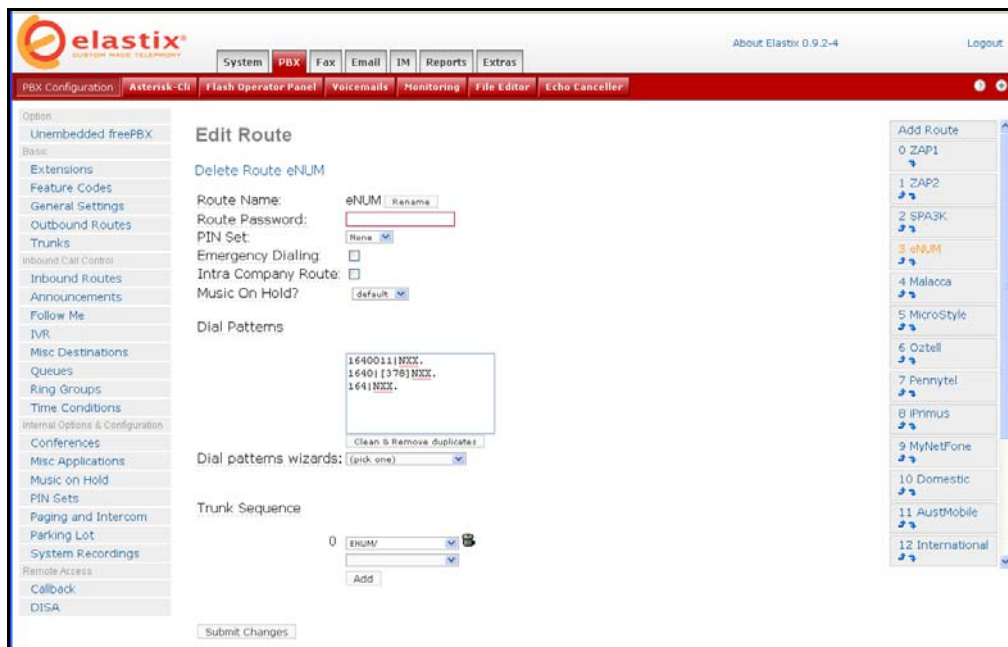
Click here for a Step-by-Step instruction on how to set up an ENUM account.

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

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.
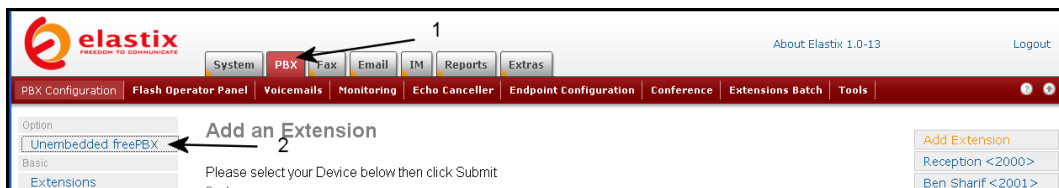
Default Login/Password is **admin/admin**
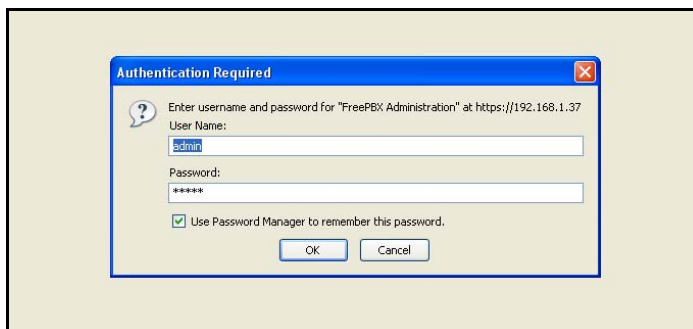
On entering freePBX, select the following:

Admin Tab → Module admin → Custom Contexts (as per the illustration below). At this stage, you will notice that Custom Contexts is not installed.
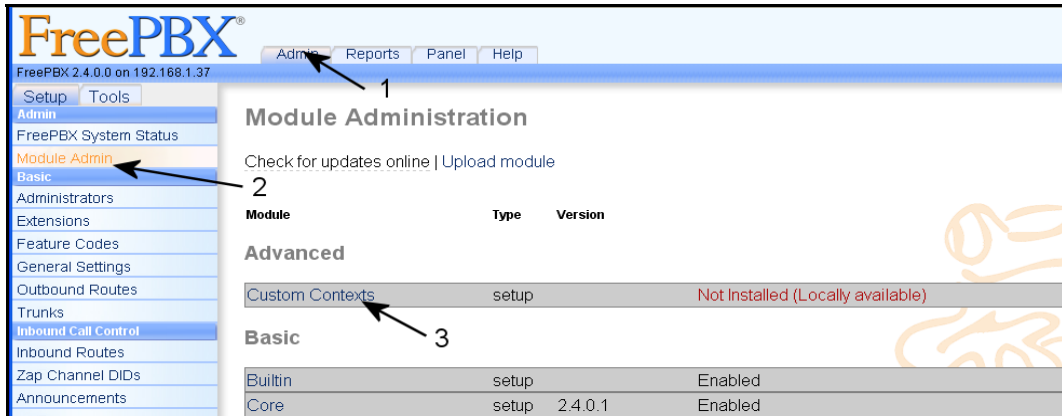
Select Install and click on Process.
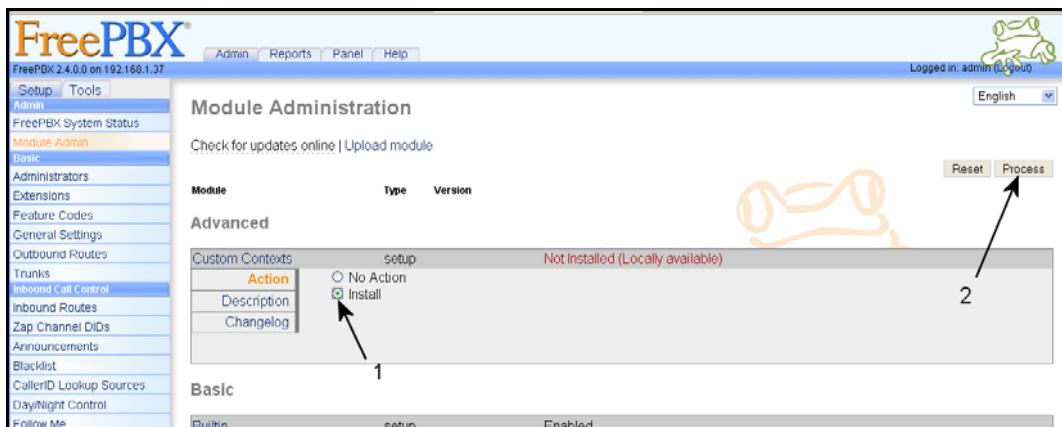
Confirm you action by clicking on the confirm button
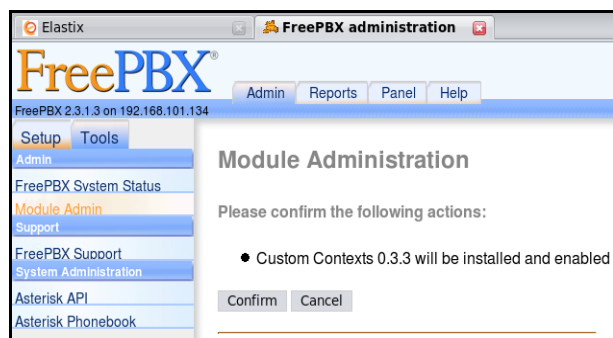
Click on the orange bar to complete the installation.
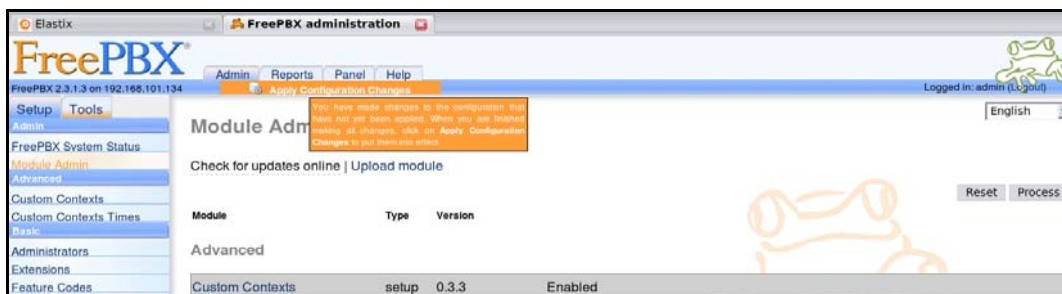
10.3 ADD AND CONFIGURE NEW CUSTOM CONTEXT

Add New custom Context: Admin → Setup → Custom Context
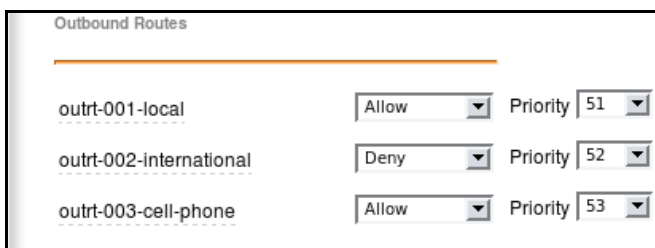
To edit the custom-context, only select: Allow or Deny each context.

To control any outbound call, select the route and from this selection, select Allow or Deny.

10.4 APPLY A CUSTOM CONTEXT TO EXTENSION.

To finish, you use the custom context on Edit Extensions, Device Options.

Device Options

This device uses sip technology.

secret	100
dtmfmode	rfc2833
canreinvite	no
context	from-internal
host	dynamic
type	friend
nat	yes
port	5060
qualify	yes
callgroup	
pickupgroup	
disallow	
allow	
dial	SIP/100
accountcode	
mailbox	100@device
Custom Context	

ALLOW ALL (Default)
allow cell calls

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

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.
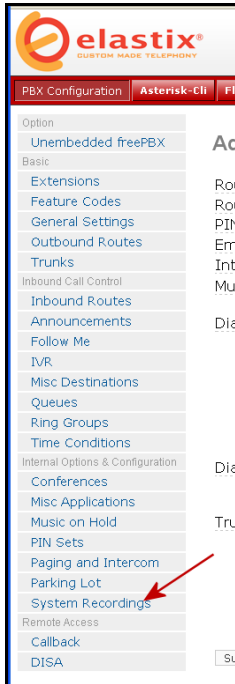
Clicking on the System recording label as highlighted by the red arrow will open the following "System Recordings" screen.

There are two methods that you can capture your recording:

- By recording directly using your telephone
- Uploading pre-recorded material
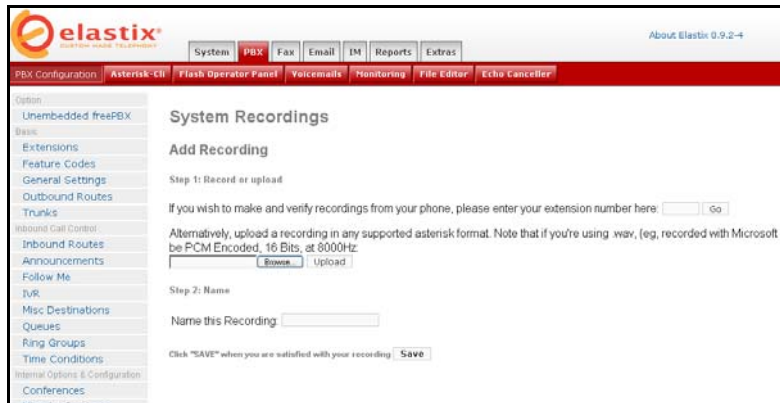
11.1 DIRECT RECORDING

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

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.

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

"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.

Step 2: Verify

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.

System Recordings

Add Recording

Step 1: Record or upload

If you wish to make and verify recordings from your phone, please enter your extension number here: [   ]

Alternatively, upload a recording in any supported asterisk format. Note that if you're using .wav, (eg, recorded be PCM Encoded, 16 Bits, at 8000Hz:

[   ] [Browse...] [Upload]

Step 2: Name

Name this Recording: [   ]

Click "SAVE" when you are satisfied with your recording [Save]

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 organisations, 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.
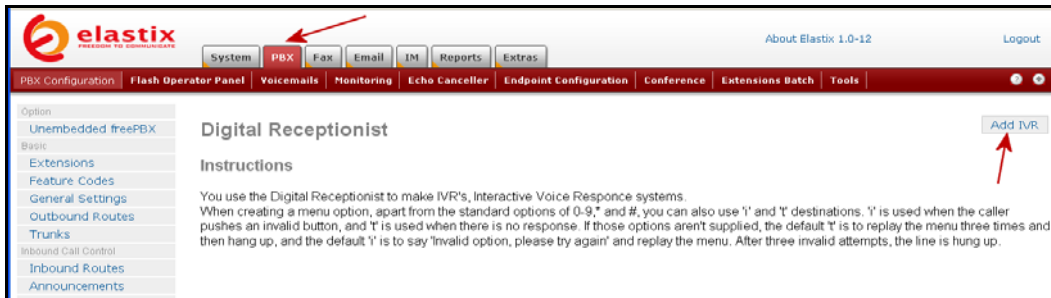
Click on the **Add IVR** option on the top right on the screen (see arrow) and the new Unnamed IVR configuration screen will be presented to you.
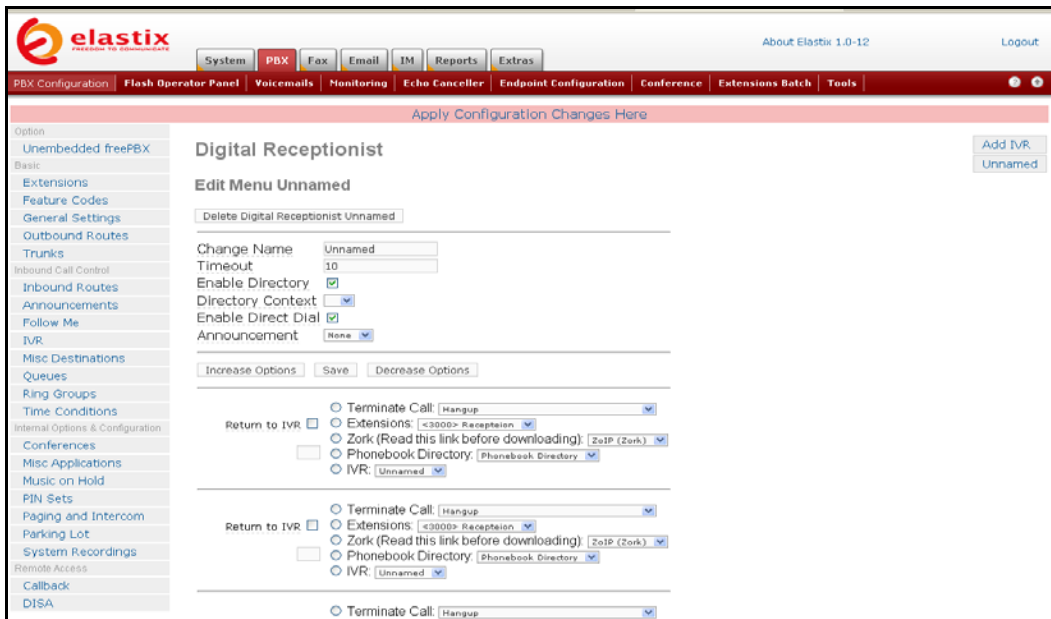
We will then fill in the following fields:

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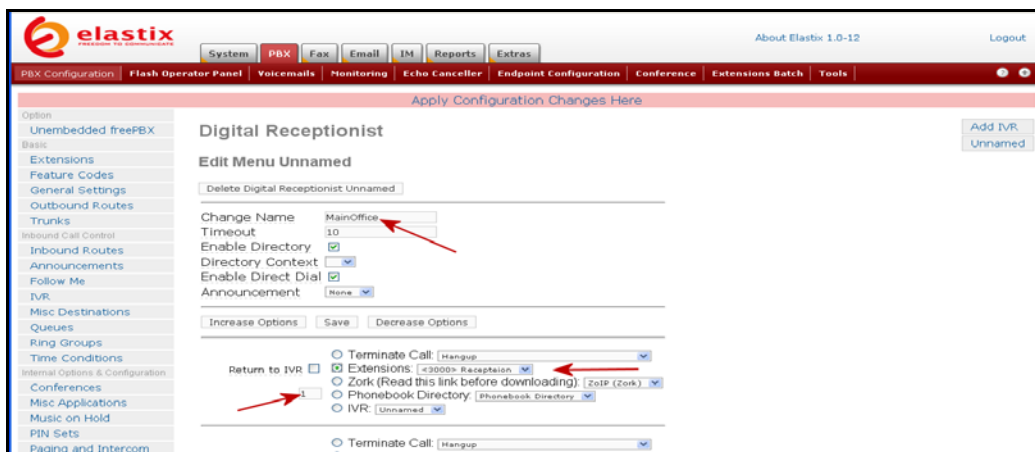
Then proceed to enter the selection options that you require for this IVR when the caller presses the appropriate option.

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.

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.

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 Installing Other Languages in the chapter referring to Customised Voice 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 haven't 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 define 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 left-hand side. You will enter the following screen.
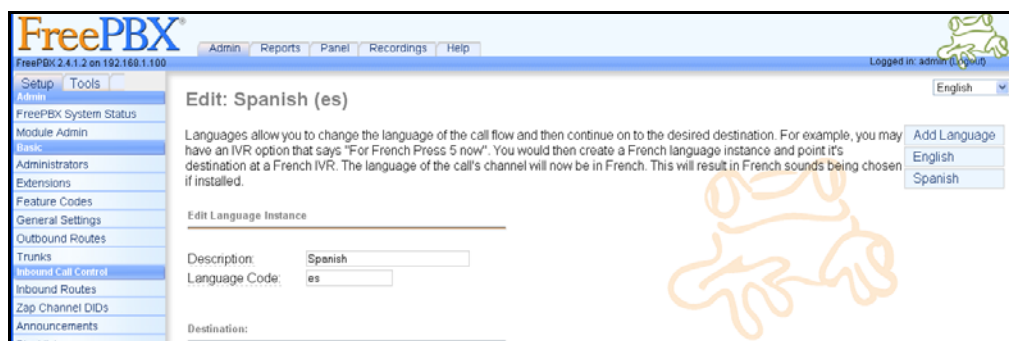
This is where you define your languages codes. The example above shows that I have defined the Spanish code where the following fields were populated:

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 extension. 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 ☺

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
- DentalIVR
- PathoIVR

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 PathoIVR

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 How to restrict outgoing calls in the Calls Handling Notes chapter of this document.

You also install the *Custom Context Module* 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 calls.

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.
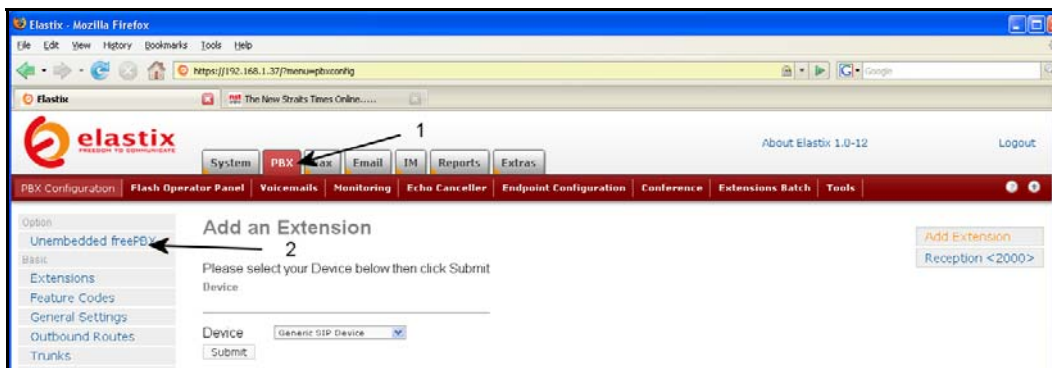
The freePBX system status screen will be displayed.
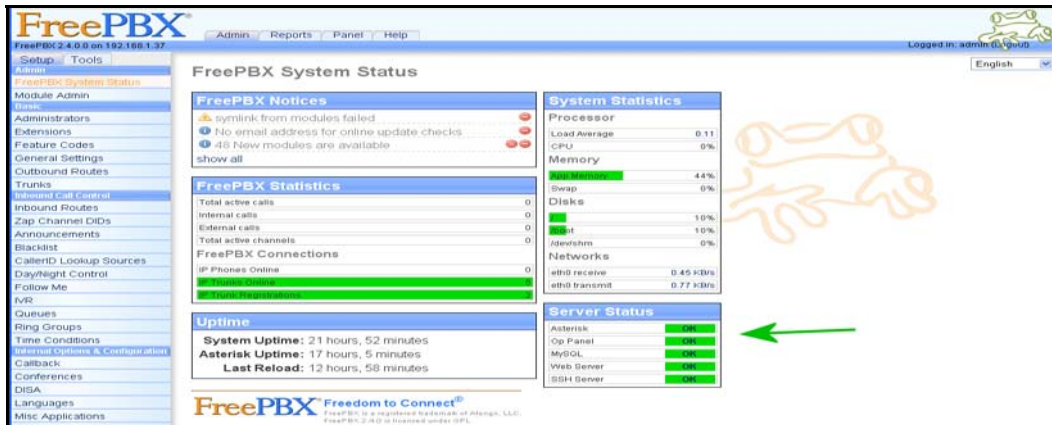
The server status should all display OK and green as illustrated above.

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 device, 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 it is time to start the installation again.

Next, while on the freePBX screen, Tools → Asterisk info → Full Report.
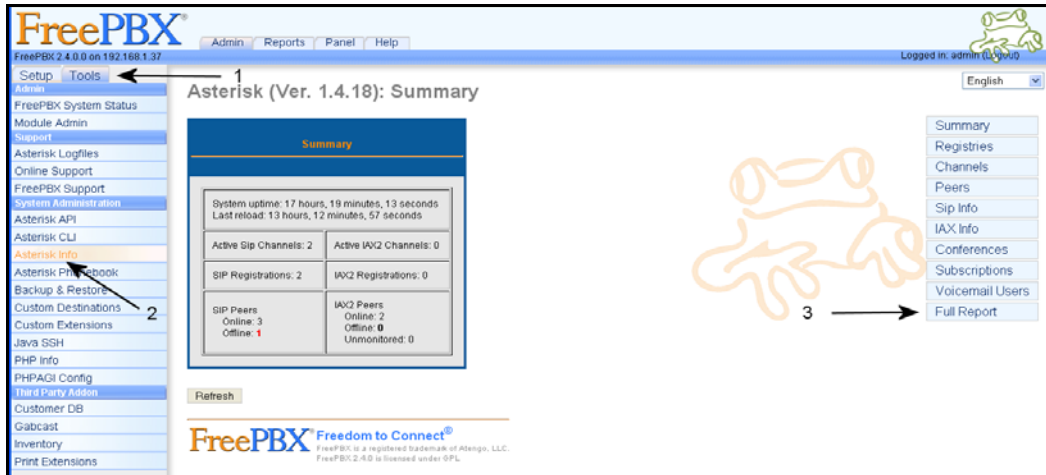
When presented with the full report page, scroll down to the SIP (or IAX if you are using IAX) section and ensure that you are registered to your trunks. You will only be able to make and receive external calls – hopefully - if you are registered.

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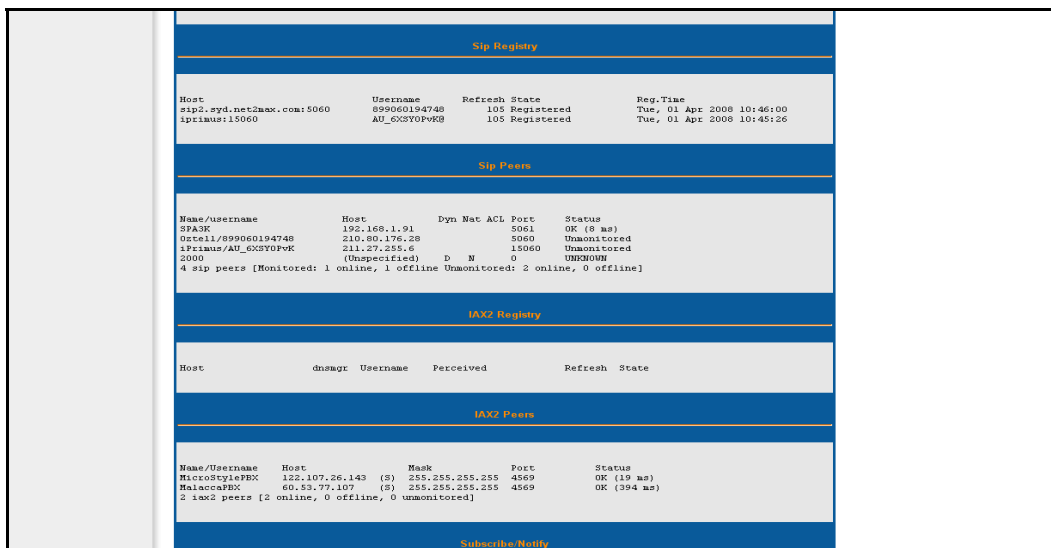
If all trunks are registered, you can be almost certain that at the very least, your Elastix installation is working (although not necessarily that you can make or receive calls).

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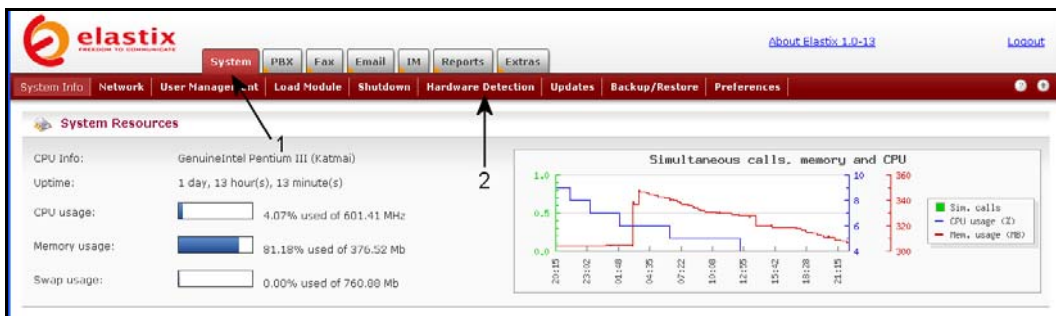
In the following screen, click on Detect New Hardware. Elastix will then detect the hardware installed.

A100P card installed

A400P with 2 FXO and 1 FXS modules installed

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 `zapata-channels.conf` was not included in the `zapata.conf`.

(**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 `zapata.conf` if it is not already there.

```
#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 *(the latest version of Elastix is using DAHDI and this file is no longer relevant)*.

```
# 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
```

Reboot after this is done.

14.1.2 Changes for DAHDI Users

DAHDI users need to edit the .conf file *etc/asterisk/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. Attention should also be given to *etc/asterisk/dahdi_channels.conf*.

chan_dahdi.conf

Please take note of the following changes and suggestions for chan_dahdi.conf.

```
[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
language=au ;(default is us)   < -- Add this line for Australian user
```

dahdi_channels.conf

In dahdi_channels.conf, ensure that **context=from-zaptel** appears in all the fxo channels as illustrated below.

```
; Autogenerated by /usr/sbin/dahdi_genconf on Sat Apr 18 14:18:44 2009
-- do not hand edit
; Dahdi Channels Configurations (chan_dahdi.conf)
;
```

```
; This is not intended to be a complete chan_dahdi.conf. Rather, it is
intended
; to be #include-d by /etc/chan_dahdi.conf that will include the
global settings
;

; Span 1: WCTDM/4 "Wildcard TDM400P REV E/F Board 5" (MASTER)
;;; line="1 WCTDM/4/0 FXSKS  (EC: OSLEC)"
signalling=fxs_ks
callerid=asreceived
useincommingcalleridonzaptransfer=yes
group=0
context=from-zaptel
channel => 1
callerid=
group=
context=default

;;; line="2 WCTDM/4/1 FXSKS  (EC: OSLEC)"
signalling=fxs_ks
callerid=asreceived
useincommingcalleridonzaptransfer=yes
group=0
context=from-zaptel
channel => 2
callerid=
group=
context=default

;;; line="3 WCTDM/4/2 FXOKS  (EC: OSLEC)"
signalling=fxo_ks
callerid="Channel 3" <4003>
mailbox=4003
group=5
context=from-internal
channel => 3
callerid=
mailbox=
group=
context=default
```

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.
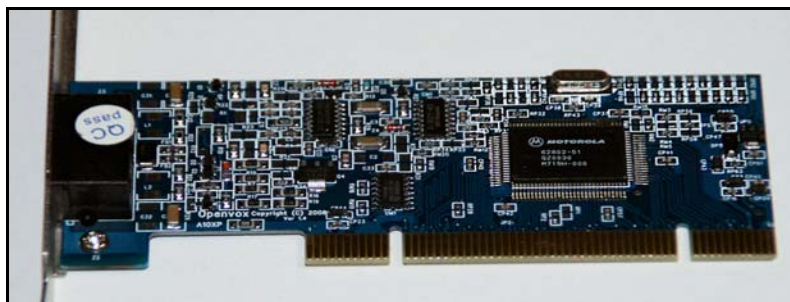
OpenVox A100P Card

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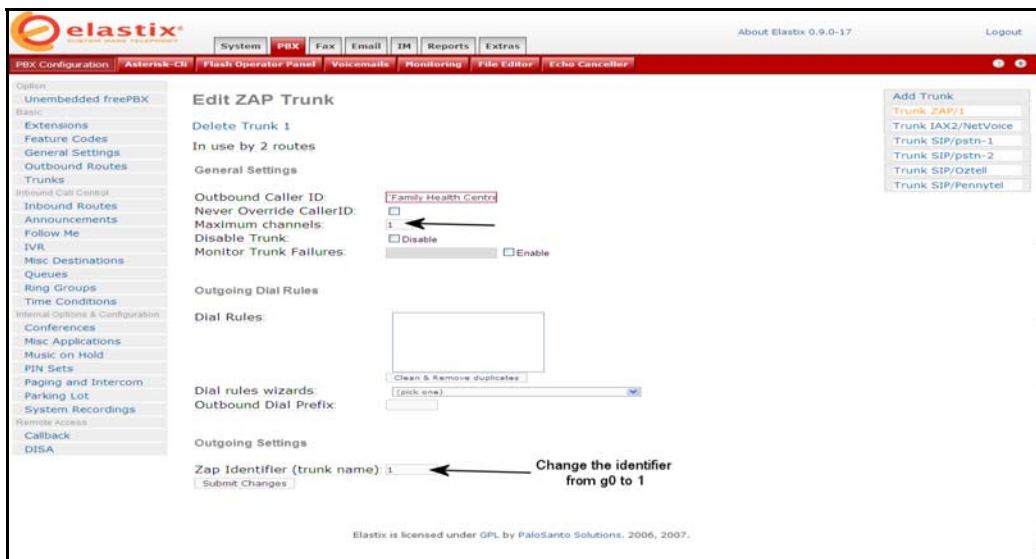
Once the card is configured, you must add a route for **Incoming Calls** or asterisk will not answer this line

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 **/etc** 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
```

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 `zapata-channels.conf`. 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 ;(For Australia, yours may be different. Default is us)
defaultzone = au  ;(that's for Australia, yours may be different)
```

For **DAHDI** users, these changes can be made in */etc/dahdi/system.conf*.

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/asterisk/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/asterisk/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=asreceived
group=0
context=from-zaptel
channel => 1
context=default

;;; line="2 WCTDM/0/1"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-zaptel
channel => 2
context=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.

Note for TDM410 users
As suggested in /usr/share/doc/zaptel-1.4.12.1/README, users of TDM410 should be using the file **wctdm24xxp** instead of **wctdm**.

ⓘ 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
```
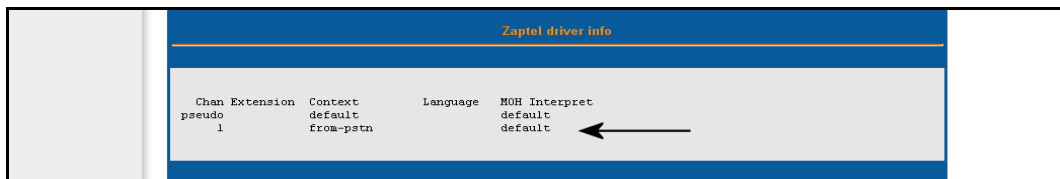
You should see the following in amongst the messages;

```
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 or DAHDI 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 lose 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 – sends caller ID to fxs
port (extension)
hanguponpolarityswitch = yes   (may have to ask your telco for the polarity reversal)
```

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 simplified installation and configuration instruction 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.

Change the settings

Ensure that you have set your SPA3102 in the Bridge Mode *(SPA3102 users)*.

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,.4/2,.4/.2,.4/2,.4/.2,.4/2,.4/.2,.4/2,.4/.2,.4/2,.4/.2,.4/2,.4/.2,.4/2)`
CWT8 Cadence: `30(.2/.2,.2/4.4)`
Hook Flash Timer Min: `.07`
Hook Flash Timer Max: `.13`
FXS Port Impedance: `220+820||120nF`

🗑 ***Delete all the Vertical Service Activation Codes.***

Line 1 tab
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|0011xxxxxxxx.|0[23478]xxxxxxxx|09xxxxxx|1100`
`|122[135]|1222xxxxxxx|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+NXXXXXXXX` *(for example)*
                `0011+ZXXXXXXXXXX.`
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)*
```
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

14.7.5 What if I use SPA3102 in Europe
(Thanks George)

If you have bought your SPA3102 from Europe and you are using it in Europe, these are the Basic Instructions for SPA3102

1) Add an extension for your SPA

     i) Go to extensions and make a new extension for SPA3102 FXS.

          Example configuration:

          -Extension number: 2300 (number it yourself)
          -Display name: 2300 (name it yourself)
          -Secret: 2300 (put your password)

Save and reload

ii) Go to SPA3102 config (admin mode), Line1 tab and register your new extension.

Example configuration

Proxy and registration
-Proxy: IP of your Elastix machine
-Register: yes
-Make calls without reg: yes
-Ans calls without reg: yes

Subscriber info
Display name: 2300 (or your name as above)
User id: 2300 (or your extension number as above)
Password: 2300 (or your password as above)

Step 1 done!

2) Add a trunk of your SPA3102

   i) Go to SPA3102 config (admin mode), PSTN tab and register your new trunk.

   Example configuration

   Proxy and registration
   -Proxy: IP of your Elastix machine
   -Register:yes
   -Make calls without reg: yes
   -Ans calls without reg: yes

   Subscriber info
   User id: PSTN (or your trunk name)
   Password: 2300 (or your password)

   Dial plans
   -Dial plan1: (xx.)
   -Dial plan2: (S0<:1029384756>) (replace 1029384756 by your number)

   Voip-to-pstn gateway setup
   -Voip-to-pstn enable: yes
   -Voip caller auth method: http digest

   Voip users and passwords
   -Voip user1 auth id: PSTN (or your trunk name as above)
   -Voip user1 dp: 1
   -Voip user1 password: 2300 (or your password as above)

   Pstn-to-voip gateway setup
   -Pstn-to-voip gateway enable: yes
   -Pstn ring thru line1: yes
   -Pstn CID for voip CID: yes
   -Pstn caller default dp: 2

ii) Go to trunks and make a new SIP trunk for SPA3102 FXO

Example configuration

-Outbound caller id: 1029384756 (replace 1029384756 by your number)

Outgoing settings
-Trunk name: PSTN

Peer details
-allow= ulaw
-canreinvite= no
-context= from-pstn
-disallow= all
-dtmfmode= inband
-host= dynamic
-nat= no
-port= 5061
-secret= 2300 (or your password as above)
-type= peer
-username= PSTN

Add an inbound route for your SPA3102
-DID number: 1029384756 (replace 1029384756 by your number)

Add an outbound route for your SPA3102
-add your dialplan with the wizard
-Trunk sequence: SIP/PSTN

Save and reload as usual

14.8 SPA3000/SPA3102 REMOTE MANAGEMENT

The SPA3000/3102 can be managed remotely. To enable remote management the following must be set as per the illustration below:

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*

15 CALLBACK AND DISA

15.1 CALLBACK

Callback is where you make a call to your IP-PBX and when reached you will be disconnected, but it does not end there. Your PBX will in turn call your mobile and reconnect you relieving you of the cost of the lengthy Mobile phone call that you will otherwise be up for.

Lets take this step by step.

1. Setup DISA

a. DISA name:MyMobile
b. Response Timeout:10
c. Digit Timeout:5
d. Caller ID:0400123456 *(My Mobile Number)*
e. Context:from-internal
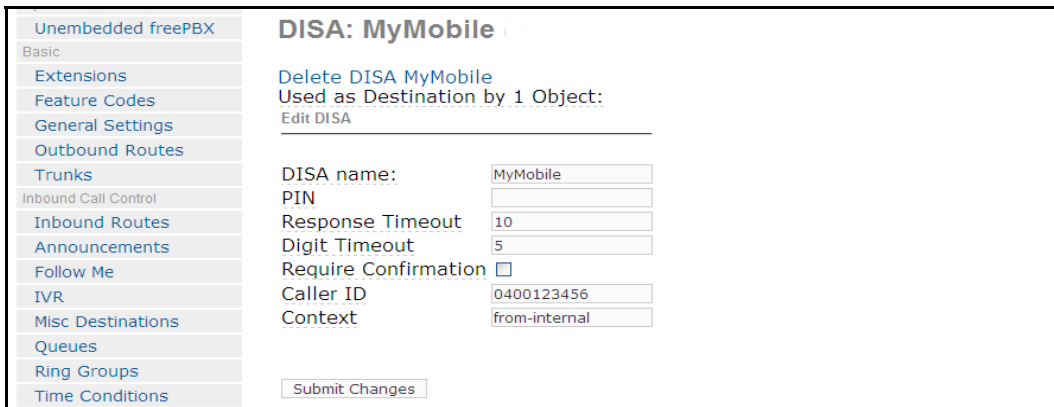
Click Submit Changes button

2. Setup Callback

a. Callback Description:MyMobile
b. Callback Number: 0400123456 *(My mobile Number)*
c. Delay Before Callback:10
d. Destination after Callback: IVR – Residence (or Office IVR)

Click Submit Changes button

3. Inbound Routes
a. Description: Callback-MyMobile
b. DID Number:61247324100 *(My DID number)*
c. Caller ID Number: 0400123456 *(My mobile Number)*

d. Set Destination to: Callback - MyMobile

Click Submit Changes button then Click on the bar at the top & follow on screen prompts
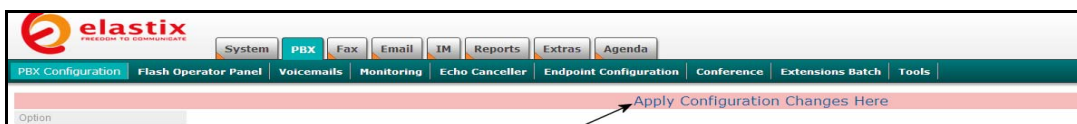
Now enable send caller ID on your mobile and call your DID number. When connected you will get one beep and then followed by silence. Hang up your mobile and wait for approximately10 seconds and your mobile will ring.

When you answer your mobile, you will hear your IVR playing with the various options. One of the silent options in my IVR is DISA. If I need to make an external call using my PBX, If I know the option and select it, I will be then get DISA where I can make an external call at no cost to my Mobile.

15.2 DISA

DISA (**D**irect **I**nward **S**ystem **A**ccess) 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.

This can be achieved through Elastix as illustrated in the screenshot below.

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.

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 after entering your password, you will then get a dialtone. You may start dialling the telephone number.

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 SOFTPEDIA and another is HUDLite from Fonality. HUD will integrate nicely with ELASTIX and the support for HUD is already integrated within ELASTIX.

16.1 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.
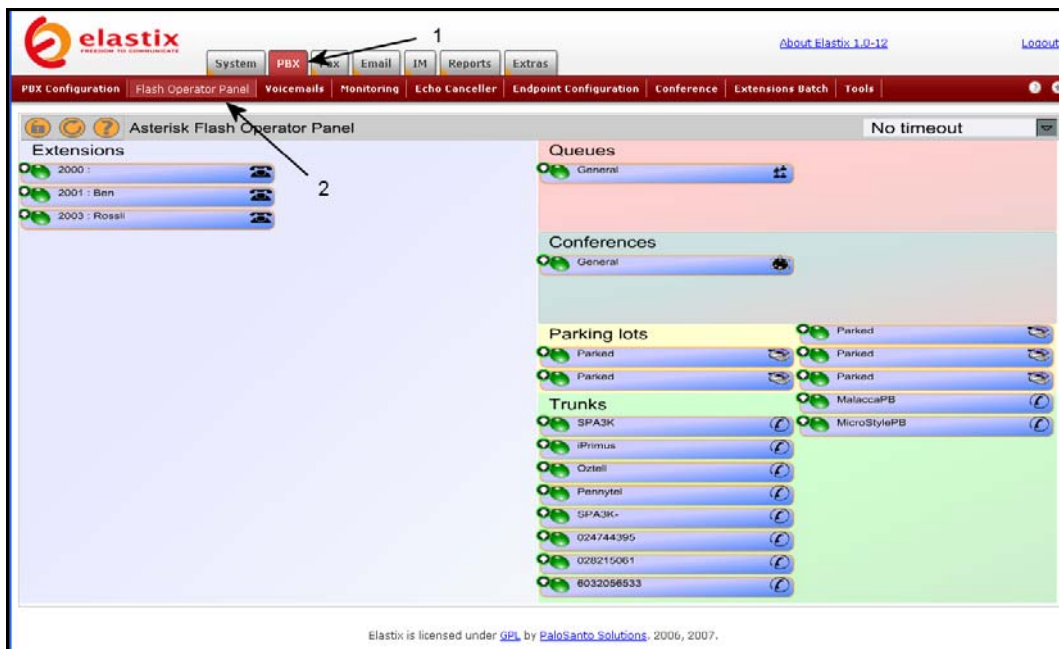
The following information is displayed on FOP:

- 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: *http://www.asternic.org/*

16.1.1 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.2oo7`
*(in FOP that comes with ELASTIX, the default password is "*eLaStIx.2oo7*")*

Replace the "eLaStIx.2oo7" 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.

16.1.2 Hang-up a Call

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

16.1.3 Transfer a Call

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

16.1.4 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.

16.1.5 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.

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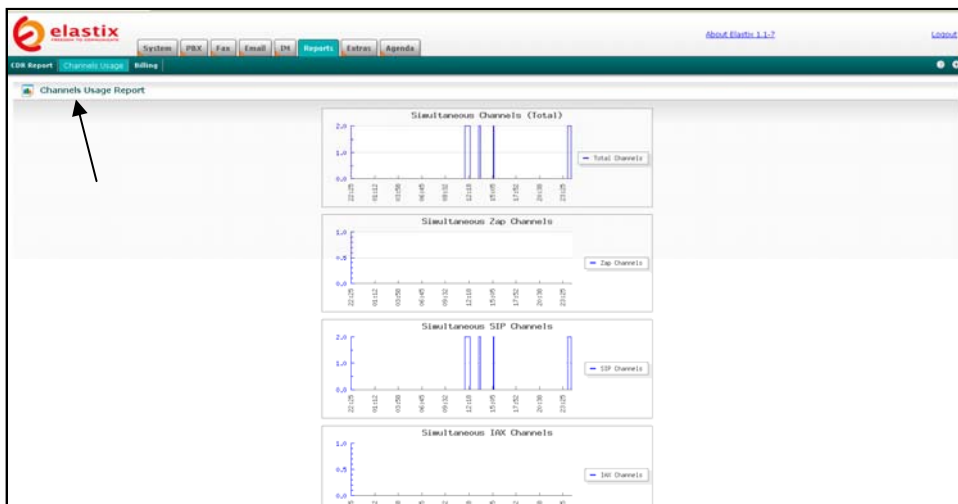
Like the CDR Report, there is no special setup or configuration to be conducted for this facility as it is simply an information gathering tool.

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)*

Parking lot is managed through PBX -→ PBX Configuration -→ Parking Lot.
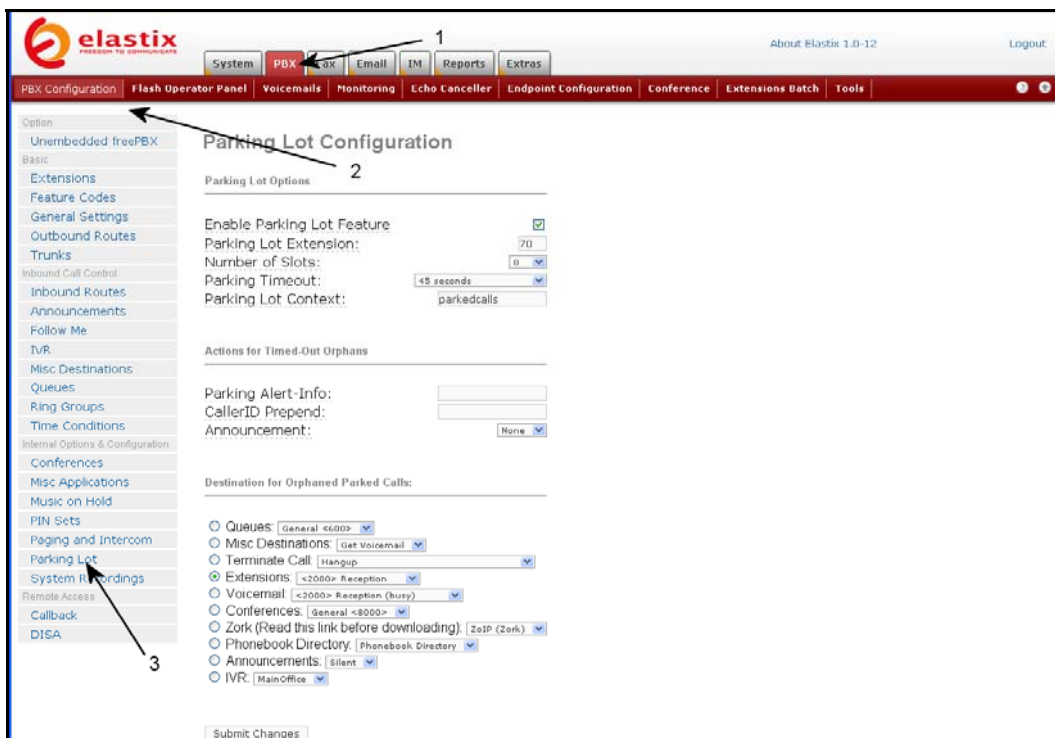
18.1 CALL TRANSFER - MANAGED

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 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 locked from the inside"* ☺ 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 *http://mundy.org/blog/index.php?p=66* relating to filtering incoming calls and also some discussions here *http://forums.whirlpool.net.au/forum-replies.cfm?t=418909*

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.

If after receiving the message 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

There are 2 type of conference facilities in Elastix, the Meet-Me Conference and another one managed by Conference Manager.

Both conference methods can be configured individually. The Meet-Me conference can be configured using FreePBX and Elastix Conference option. However the more controllable Conference Manager option can only be configured through Elastix.

The Meet-me 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 Meet-me 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 the meet-me 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 MEET-ME 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:
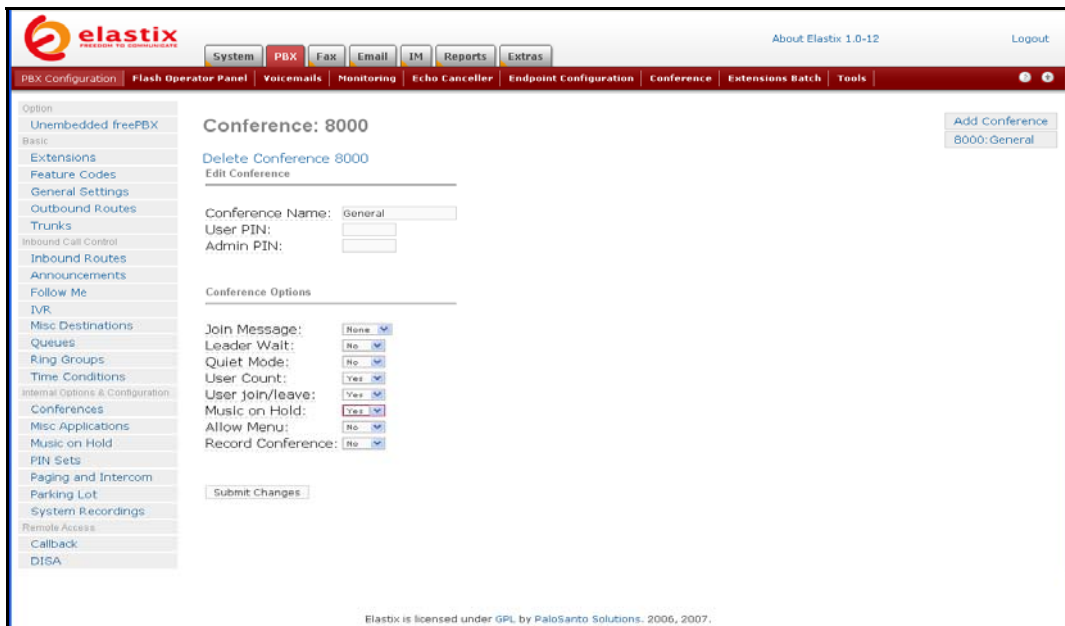
- This conference room number is **8000**
- I call this conference room **General**
- Users are not required to enter a PIN number to join the conference.
- Admin PIN number is **not required** *for the conference to wait for the leader to conduct the conference.*
- I don't have a joining message greeting 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. You may include ext: 8000 in your IVR choice for external callers to join in 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.

19.2 ELASTIX CONFERENCE MANAGER

This method can only be setup through Elastix. 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.
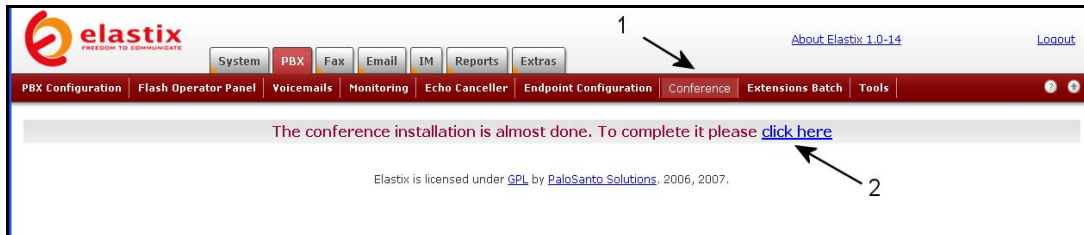
To activate this module, it is necessary to click on the "click here" link.

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.

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.
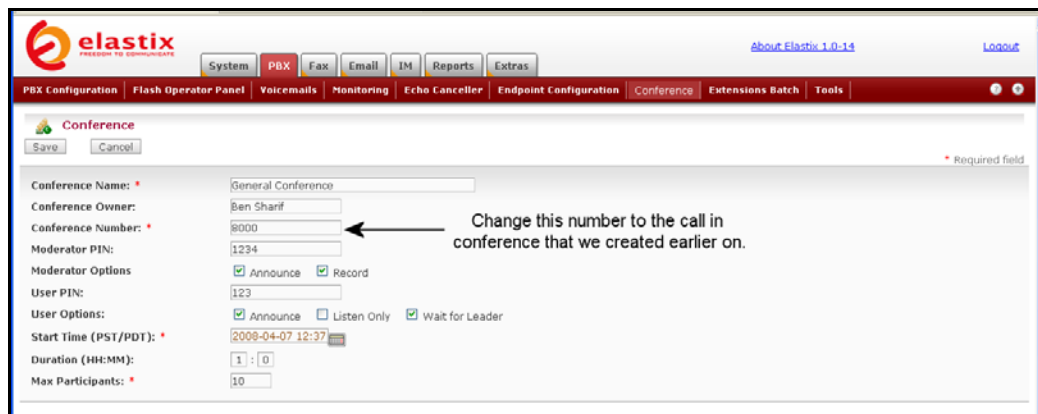
In the fields, we enter the following:

Conference Name: General Conference
Conference Owner: <Your Name>
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 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.

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

Conference Name: Marketing
Conference Owner: <Your Name>
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.

19.3 CONTROLLING AND MODERATING THE CONFERENCE

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

The Elastix team has taken all the hard work away 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).
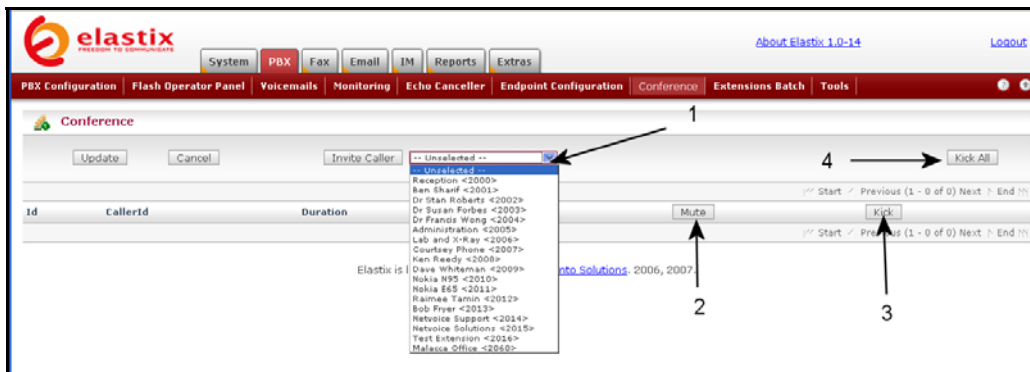
Here you can do the following:

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

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

19.4 HANDLING MULTIPLE CALL-IN CONFERENCE

Using the conference manager, a number of conferences can be handles by creating various conference rooms.

Do not use the meet-me conference GUI of freePBX instead use the Elastix conference Tab to do this.

The process of creating conference room is similar to the previous chapters on Conference Manager. The conference rooms can be identified by the conference numbers allocated by the conference manager or define our own conference number.

Once all the rooms have been created, internal users can be either invited to the conference or can ring in to join the various conferences identified by the room numbers. Participants must be informed of the conference number so they can choose which conference to join.

For internal users, they can dial 5555 (a pre-defined extension in *extensions_custom.conf*) and will be prompted to select the conference number to join.

Since external users will not be able to dial custom extension 5555, an IVR selection can be created for this purpose to direct the callers to the conference selection prompt.

To do this we have to first create a Custom Destination through freePBX. 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: `conferences,5555,1`
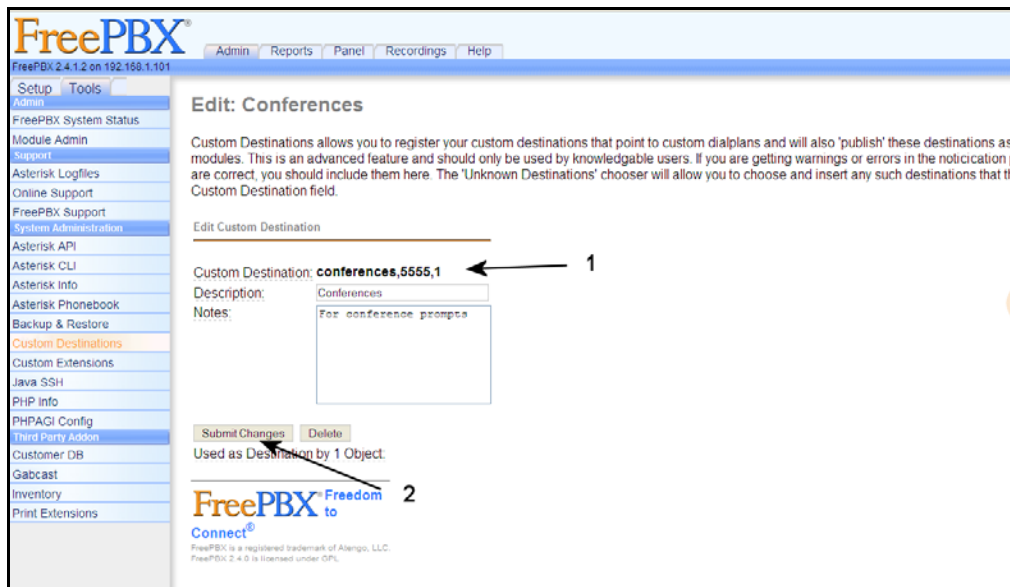
Click Submit Change when done.

Conferences will now appear in the dropdown list of Custom Application selection of the IVR and you can include a selection for Conference in your IVR.
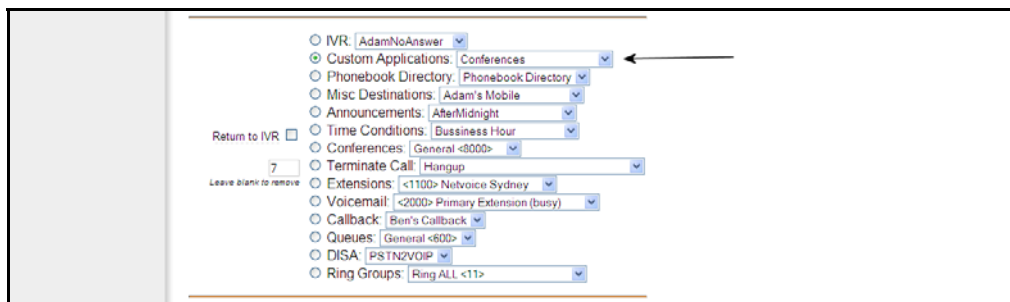
If the situation requires, you may also direct incoming calls straight to this option.

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;

```
relayhost = mail.optusnet.com.au
```
← *being my ISP smtp server*

* *Optus blocks outgoing port 25 on cable to stop people running their own mail servers. They force relaying through their server*

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*.

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

# 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 → 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/cgi-bin/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=vm@asterisk       ; 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 entries 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/cgi-
bin/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 8th.
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 users 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: *http://www.soft-switch.org/foip.html* and : *http://www.voip-info.org/wiki-Asterisk+fax*

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 by using Hylafax, 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 is 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.*

Since the introduction of DAHDI, it has become necessary to also install NVFaxdetect if you wish to share your Voice lines with Faxing *(if you are using zaptel instead of DAHDI you are spared this extra chore).*

Configuring Elastix Fax Module

Click on the FAX tab and the following screen will appear.

Next is to Click on the New Virtual Fax selection – see arrow.

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".**

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.

You will be presented with the usual create new extension screen as illustrated below:

We have populated the various fields:

- 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)
- Dictation service: Disabled
- Incoming/outgoing recording: Never

Press submit when done.

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)

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 (**set Fax Detect to NVFax** if you are using the current version of Elastix) leaving your final destination to the voice call if it is not a fax call.

This is where the difference is if you are using DAHDI instead of Zaptel. If you are using DAHDI, chances are you will be, you will have to install NVFaxdetect. You can get the 32 bit version here http://dl.dropbox.com/u/4264251/Fax.rar.

After unpacking the .rar file you will have 3 files;

```
app_nv_backgrounddetect.so
app_nv_faxdetect.so
app_nv_faxemail.so
```

Copy these 3 files to your /usr/lib/asterisk/modules directory and reboot your Elastix system box.

Description: Voice and FAX
DID Number: 61247364000
Caller ID Number:
Fax Handling

Fax Extension: Sydney FAX <1100>
Fax Email:
Fax Detection Type: Zaptel
Pause After Answer: 0
Privacy

Privacy Manager: No
Options

Alert Info:
CID name prefix::
Music On Hold? default
Signal RINGING: ☐
CID Lookup Source

Source: None
Set Destination

○ Queues: General <600>
○ Misc Destinations: Ben's Mobile
○ Terminate Call: Hangup
○ Extensions: <1100> Sydney FAX
○ Voicemail: <2000> Reception (busy)
○ Conferences: General <8000>
○ Zork (Read this link before downloading): ZoIP (Zork)
○ Phonebook Directory: Phonebook Directory
○ Announcements: Silent
○ Custom Contexts: Full Internal Access
○ DISA: VoIP2PSTN
○ Ring Groups: Ring ALL <11>
● IVR: MainOffice

.
The last thing that you need to do is make a little change in **zapata.conf** or if you are using DAHDI, you will have to make this change in **etc/asterisk/chan_dahdi.conf**.

1. Use nano to edit zapata .conf (or chan_dahdi.conf)
2. Ensure that the line `faxdetect=incoming` (see illustration below) is not commented out.
3. If the line is not there then add faxdetect=incoming" to your zapata.conf or chan_dahdi.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

The Fax Visor is used to check how many faxes have been received and to check their contents..

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.

21.3 HYLAFAX FAX CLIENT

Using your windows PC to send fax is possible if you have the Hylafax Fax client installed on your PC.

For my purpose, I am using "Yet Another Java HylaFAX Client", **YajHFC** in short *thanks to Dave Edwards for pointing out this extremely useful client to me*, which can be downloaded from here *http://prdownload.berlios.de/yajhfc/yajhfc-0_4_0-FOPPlugin-0_1_3-setup.exe*

After downloading, it is just a matter of installing the client on a Windows PC that is on your LAN. Ideally this PC would have a fixed Local IP address (in my case it is 192.168.1.20) as this address will be required by the Elastix fax module.

21.3.1 Preparing your PC to be the fax client

Before you can operate YajHFC on your PC, you will have to ensure that Elastix knows about your PC so it can communicate with each other.

Under the Elastix GUI Fax -- > Fax Client Tab, add the IP address of your PC (in my case it is 192.168.1.20 as illustrated below) and apply the change.

21.3.2 Preparing YajHFC

Once installed, it will be recognised by your PC as a YajHFC Fax/printer.

Click on the YajHFC icon to start the client and you will get the following screen:
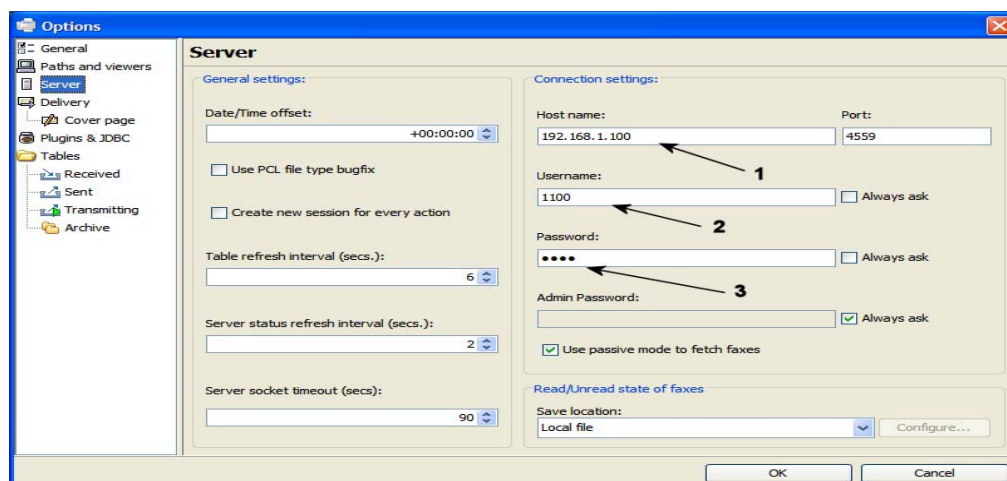
All you need to do to get started is to fill in the three fields as illustrated above:

Host name: 192.168.1.100 (my Elastix Box IP address – yours may be different)
Username: 1100 (My fax extension – yours may be different)
Password: The password for the fax extension

You may leave the rest of the settings at default. You can always explore once you are comfortable that the client is working correctly.

Once that is done, the client will initialise and you will get the following screen.
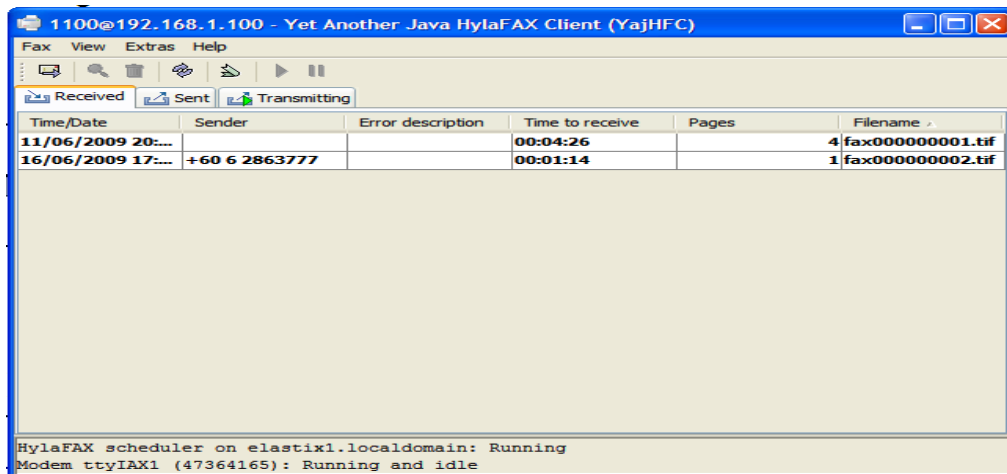
If you have already received some faxes prior to installing this client, they will be shown (as above) otherwise the screen will be blank.

Your fax client is now ready to receive a fax transmission.

21.3.3 To send a Fax from your PC

To send a word document from your PC, all you have to do is open the Word document and print it to the Fax printer (see illustration below).
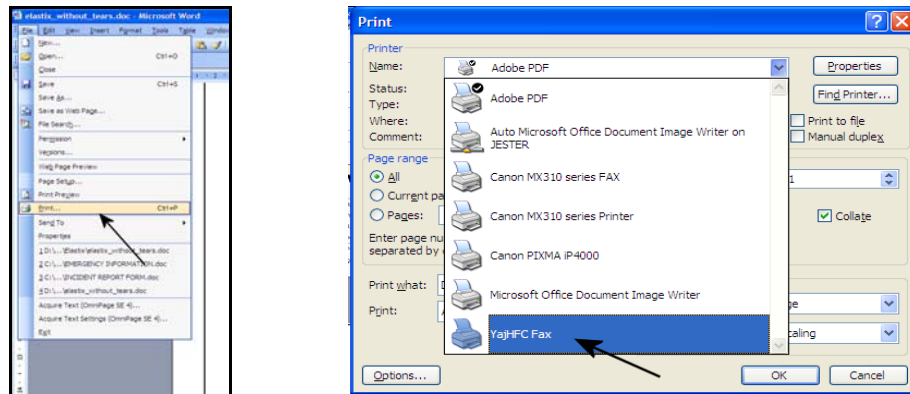
You will then be required to enter the fax phone number you wish to send the fax to.

Once done click on send (see illustration below)
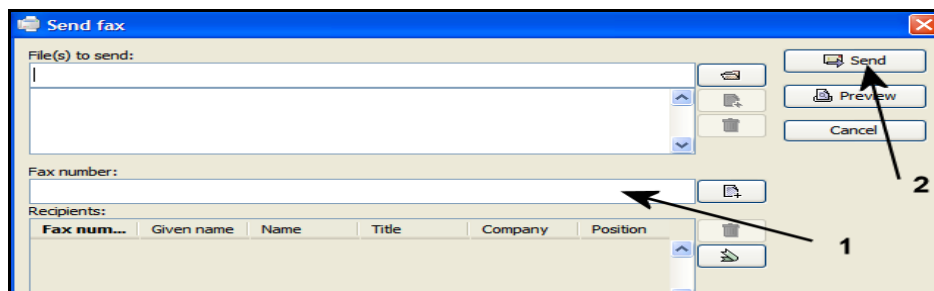
22 WEATHER REPORTS

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: *http://www.bom.gov.au/*

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 *http://www.speech.cs.cmu.edu/flite/doc/index.html*.

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 -O 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 -O 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.

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

```
; ################################
; 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,Playback(/tmp/weather1)
exten => *61,5,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,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`
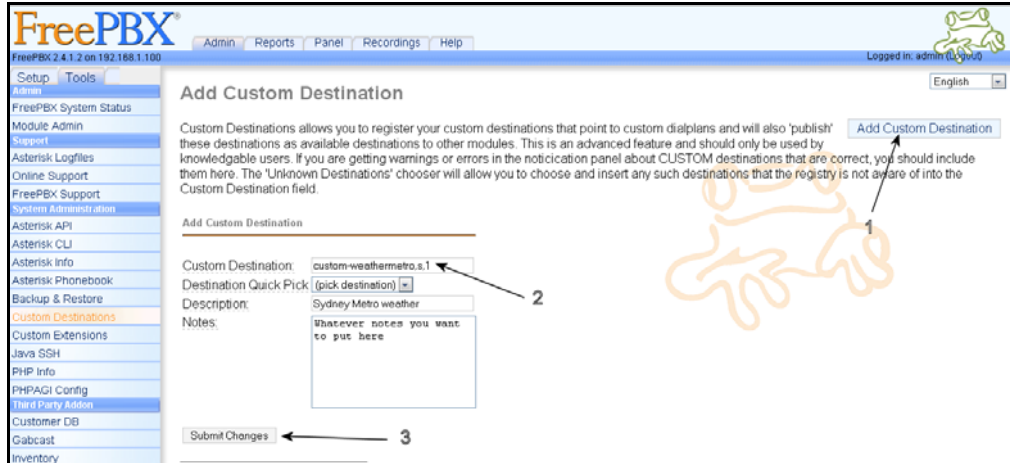
You can create another one for custom-weathernsw using similar format.

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` or `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.*
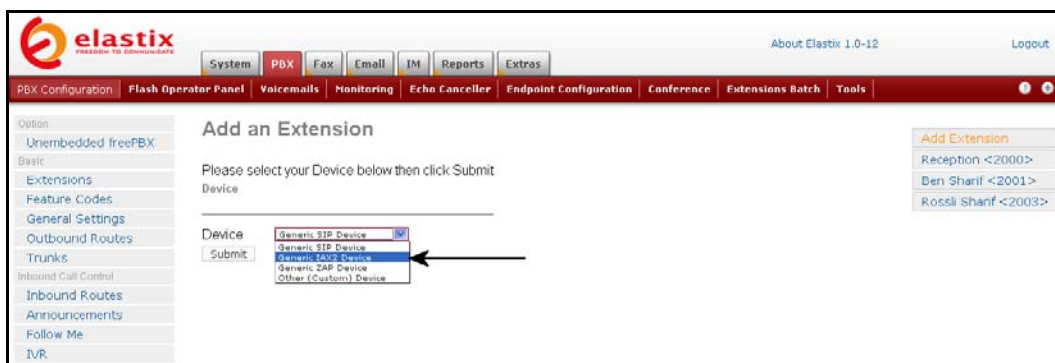
As mentioned above, you may create a SIP extension instead, if you so wish but you must ensure that you set "**nat**" field to "**yes**" (in the device option) instead of "never" as illustrated in the screen below - see red arrow.

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

Changes made in iax.conf will be rewritten over every time asterisk initializes itself and therefore any changes that you may need to make should be done in one of the cusom files. In this case the changes I needed was done in *iax_custom.conf*.

The following is the content of my *iax.conf* and *iax_custom.conf* file.

24.2.1 iax.conf

```
;--------------------------------------------------------------------------;
;Do NOT edit this file as it is auto-generated by FreePBX. All modifications to
;
;this file must be done via the web gui. There are alternative files to make
;
;custom modifications, details at: http://freepbx.org/configuration_files
;
;--------------------------------------------------------------------------;
;

[general]

; These will all be included in the [general] context
#include iax_general_additional.conf
#include iax_general_custom.conf
#include iax_registrations_custom.conf
#include iax_registrations.conf

; These should all be expected to come after the [general] context
;
;iax_custom.conf is the proper place to start creating new contexts that you
;might have a need for.  Dundi IAX trunks is one example of when this file is
needed.
;
#include iax_custom.conf
#include iax_additional.conf
;
;iax_custom_post.conf will allow you to modify FreePBX generated IAX setups so
;that you can add additional parameters to a auto-generated setup.
;if you have a auto-generated context of [foobar] and need to add a parameter
;to it then create this line [foobar](+) and place your additions on the next
line
;
#include iax_custom_post.conf
```

24.2.2 Iax_custom.conf

```
bindport = 4569  ; Port to bind to (IAX is 4569. Forward this port to your Asterisk box)
externhost = netvoice.selfip.com  ;
;externip = 122.107.29.48
localnet = 192.168.1.0/255.255.255.0
bindaddr = 0.0.0.0      ; Address to bind to (all addresses on machine)
language=au
delayreject = yes
jitterbuffer = yes
mailboxdetail = yes
```

24.3 SIP_NAT.CONF

*(**Note:** Users of the later versions of Elastix should include the content of **sip_nat.conf** in **sip_general_custom.conf** amd should ignore sip_nat.conf altogether)*

The only other .conf file that requires attention is the *sip_nat.conf* in case you are upgrading from older version of Elastix and 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> ;  < -- Do not use this for Pennytel – leave it out
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:
http://www.asteriskguru.com/tools/idefisk_beta.php

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.
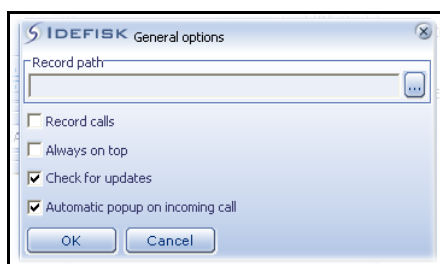
The next option to take care of is the Account option. Apart from that, there is nothing else critical to be done for you to use IDEFISK.

24.4.2 Accounts options
Next click on the Account Options and set up your account as per the image below.

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 it is illegal. If you are recording conversation without the consent of all parties, you may be breaking the law – so be warned. It is also a good idea to indicate when the call is being recorded. To do this insert the following line in /*etc/asterisk/features_general_custom.conf*.

```
Courtesytone = beep
```

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.
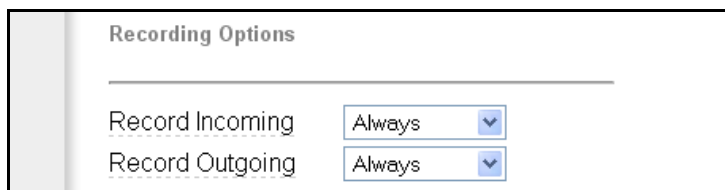
To upgrade Elastix to the latest version, you simply log on to the console (or ssh using putty) and issue the following command at the root ptompt:

```
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.

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.
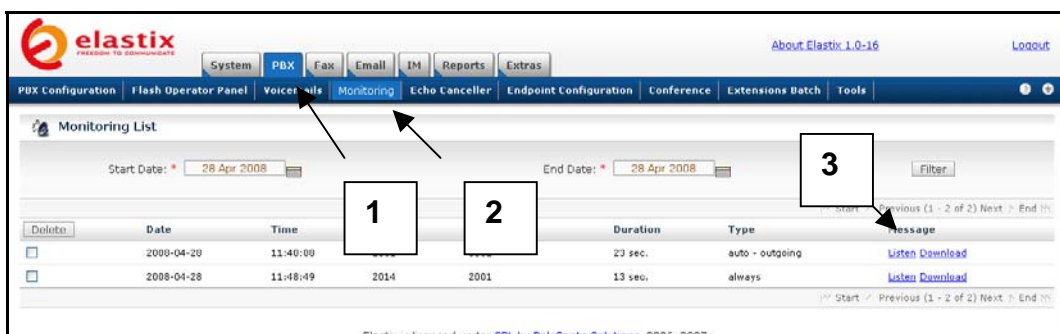
25.5 ARI INTERFACE

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 older Elastix distribution (download here) – it is however installed in the latest distribution.

If you want to listen to all the calls, you will need to log in as *admin* with the default password of *ari_password* (or *password*), otherwise you can only listen to your recordings only.
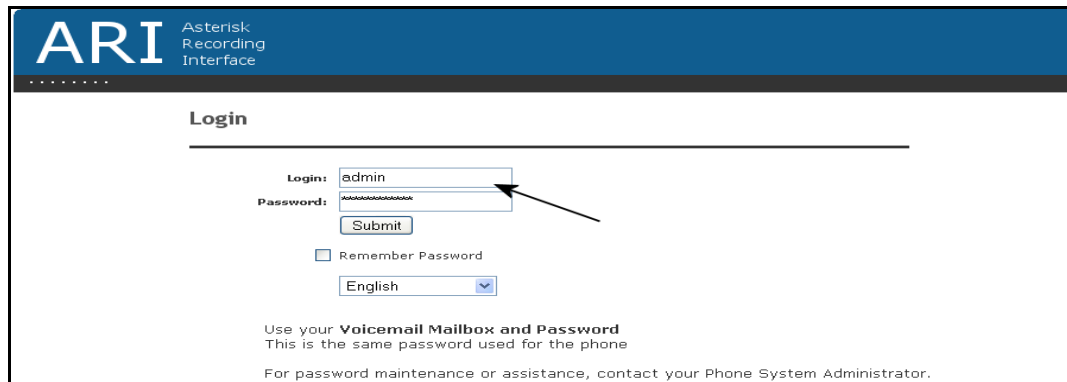
To monitor the calls for any individual extension, log in using the *extension number* and the *extension password*.

To play the recording, simply click on the **Play** link related to the extension that you want to listen to, in the screen that follows.

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.

If you get the following error chances are **main.conf.php** is missing.

"ARI does not appear to have access to the Asterisk Manager. ()
Check the ARI 'main.conf.php' configuration file to set the Asterisk Manager Account.Check
/etc/asterisk/manager.conf for a proper Asterisk Manager Accountmake sure [general] enabled =
yes and a 'permit=' line for localhost or the webserver".

To overcome this you may create **main.conf.php** and place the following in the file.

```php
<?php

/**
 * @file
 * site-specific configuration file.
 */

###############################
# AMP or standalone settings
###############################
#
# From AMP.  Used for logon to database.
#
$AMP_FUNCTIONS_FILES = "../admin/functions.php;../admin/functions.inc.php";
$AMPORTAL_CONF_FILE = "/etc/amportal.conf";

#
# Host for Asterisk Manager Interface
#
$ASTERISKMGR_DBHOST = "127.0.0.1";

#
# Database options for older legacy AMP installations (pre-FreePBX)
#   - $LEGACY_AMP_DBFILE only needs to be set if using a database like sqlite
#
$LEGACY_AMP_DBHOST = "127.0.0.1";
$LEGACY_AMP_DBENGINE = "mysql";
$LEGACY_AMP_DBFILE = "";
$LEGACY_AMP_DBNAME = "asterisk";

#
# Database cdr settings
#   - Only need to update these settings if standalone or an older AMP version
(pre-FreePBX) is used
#   - $ASTERISKCDR_DBFILE only needs to be set if using a database like sqlite
#   Options: supported database types (others are supported, but not listed)
#     'mysql' - MySQL
#     'pgsql' - PostgreSQL
```

```
#       'oci8' - Oracle
#       'odbc' - ODBC
#
$ASTERISKCDR_DBHOST = "127.0.0.1";
$ASTERISKCDR_DBENGINE = "mysql";
$ASTERISKCDR_DBFILE = "";
$ASTERISKCDR_DBNAME = "asteriskcdrdb";
$ASTERISKCDR_DBTABLE = "cdr";


#
# Standalone, for use without AMP
#   set use = true;
#   set asterisk_mgruser to Asterisk Call Manager username
#   set asterisk_mgrpass to Asterisk Call Manager password
#
$STANDALONE['use'] = false;
$STANDALONE['asterisk_mgruser'] = "";
$STANDALONE['asterisk_mgrpass'] = "";
$STANDALONE['asteriskcdr_dbuser'] = "";
$STANDALONE['asteriskcdr_dbpass'] = "";

################################
# authentication settings
################################
#
# For using the Call Monitor only
#   option: 0 - use Authentication, Voicemail, and Call Monitor
#           1 - use only the Call Monitor
#
$ARI_NO_LOGIN = 0;


#
# Admin only account
#
$ARI_ADMIN_USERNAME = "admin";
$ARI_ADMIN_PASSWORD = "password";
#
# Admin extensions
#   option: Comma delimited list of extensions
#
$ARI_ADMIN_EXTENSIONS = "";


#
# Authentication password to unlock cookie password
#   This must be all continuous and only letters and numbers
#
$ARI_CRYPT_PASSWORD =
"z1Mc6KRxA7Nw90dGjY5qLXhtrPgJOfeCaUmHvQT3yW8nDsI2VkEpiS4blFoBuZ";

################################
# modules settings
################################
#
# modules with admin only status (they will not be displayed for regular users)
#   option: Comma delimited list of module names (ie
voicemail,callmonitor,help,settings)
#
$ARI_ADMIN_MODULES = "";

#
# disable modules (you can also just delete them from /recordings/modules
without problems)
#   option: Comma delimited list of module names (ie
voicemail,callmonitor,help,settings)
#
$ARI_DISABLED_MODULES = "";

#
# sets the default admin page
#   option: Comma delimited list of module names (ie
voicemail,callmonitor,help,settings)
#
$ARI_DEFAULT_ADMIN_PAGE = "callmonitor";

#
```

```
        # sets the default user page
        #   option: Comma delimited list of module names (ie
        voicemail,callmonitor,help,settings)
        #
        $ARI_DEFAULT_USER_PAGE = "voicemail";


        #
        # enables ajax page refresh
        #   option: 0 - disable ajax page refresh
        #           1 - enable ajax page refresh
        #
        $AJAX_PAGE_REFRESH_ENABLE = 1;


        #
        # sets the default user page
        #   option: refresh time in 'minutes:seconds' (0 to inifinity) : (0 to 59)
        #
        $AJAX_PAGE_REFRESH_TIME ="01:00";
        ###############################
        # voicemail settings
        ###############################
        #
        # voicemail config.
        #
        $ASTERISK_VOICEMAIL_CONF = "/etc/asterisk/voicemail.conf";


        #
        # To set to a specific context.
        #   If using default or more than one context then leave blank
        #
        $ASTERISK_VOICEMAIL_CONTEXT = "";


        #
        # Location of asterisk voicemail recordings on server
        #    Use semi-colon for multiple paths
        #
        $ASTERISK_VOICEMAIL_PATH = "/var/spool/asterisk/voicemail";


        #
        # valid mailbox folders
        #
        $ASTERISK_VOICEMAIL_FOLDERS = array();
        $ASTERISK_VOICEMAIL_FOLDERS[0]['folder'] = "INBOX";
        $ASTERISK_VOICEMAIL_FOLDERS[0]['name'] = _("INBOX");
        $ASTERISK_VOICEMAIL_FOLDERS[1]['folder'] = "Family";
        $ASTERISK_VOICEMAIL_FOLDERS[1]['name'] = _("Family");
        $ASTERISK_VOICEMAIL_FOLDERS[2]['folder'] = "Friends";
        $ASTERISK_VOICEMAIL_FOLDERS[2]['name'] = _("Friends");
        $ASTERISK_VOICEMAIL_FOLDERS[3]['folder'] = "Old";
        $ASTERISK_VOICEMAIL_FOLDERS[3]['name'] = _("Old");
        $ASTERISK_VOICEMAIL_FOLDERS[4]['folder'] = "Work";
        $ASTERISK_VOICEMAIL_FOLDERS[4]['name'] = _("Work");


        ###############################
        # call monitor settings
        ###############################
        #
        # Location of asterisk call monitor recordings on server
        #
        $ASTERISK_CALLMONITOR_PATH = "/var/spool/asterisk/monitor";


        #
        # Extensions with access to all call monitor recordings
        #   option: Comma delimited list of extensions or "all"
        #
        $CALLMONITOR_ADMIN_EXTENSIONS ="";
        #
        # Allow call monitor users to delete monitored calls
        #   option: 0 - do not show controls
        #           1 - show controls
        #
        $CALLMONITOR_ALLOW_DELETE = 1;


        #
        # Allow for aggressive matching of recording files to database records
```

```
#       will match recordings that are marked several seconds off
#    option: 0 - do not aggressively match
#             1 - aggressively match
#
$CALLMONITOR_AGGRESSIVE_MATCHING = 1;


#
# Limits log/recording file matching to exact matching
#       will not try to look through all the recordings and make a best match
#       even if there is not uniqueid
#       requires that the MYSQL_UNIQUEID flag be compiled in asterisk-addons
#       (in the asterisk-addon Makefile add the following "CFLAGS+=-
DMYSQL_LOGUNIQUEID")
#
#       * use if there are or will be more than 2500 recording files
#
#    option: 0 - do not exact match
#             1 - only exact match
#
$CALLMONITOR_ONLY_EXACT_MATCHING = 0;

###############################
# conference page settings
###############################
#
# Meetme extension prefix
#   for this module to function, the user has to have
#   a meetme conference room {prefix}{extension}
#
$CONFERENCE_WEBMEETME_PREFIX = "";

#
# url to web meetme conference room
#   example: "http://example.mycompany.com/webmeetme"
#
$CONFERENCE_WEBMEETME_URL = "";

###############################
# help page settings
###############################
#
# help feature codes
#   list of handset options and their function
#
$ARI_HELP_FEATURE_CODES = array();
//$ARI_HELP_FEATURE_CODES['*411'] = _("Directory");
//$ARI_HELP_FEATURE_CODES['*43'] = _("Echo Test");
//$ARI_HELP_FEATURE_CODES['*60'] = _("Time");
//$ARI_HELP_FEATURE_CODES['*61'] = _("Weather");
//$ARI_HELP_FEATURE_CODES['*62'] = _("Schedule wakeup call");
//$ARI_HELP_FEATURE_CODES['*65'] = _("festival test (your extension is XXX)");
//$ARI_HELP_FEATURE_CODES['*77'] = _("IVR Recording");
//$ARI_HELP_FEATURE_CODES['*99'] = _("Playback IVR Recording");
//$ARI_HELP_FEATURE_CODES['666'] = _("Test Fax");
//$ARI_HELP_FEATURE_CODES['7777'] = _("Simulate incoming call");

$ARI_HELP_FEATURE_CODES['*72'] = _("Call Forward All Activate");
$ARI_HELP_FEATURE_CODES['*73'] = _("Call Forward All Deactivate");
$ARI_HELP_FEATURE_CODES['*74'] = _("Call Forward All Prompting Deactivate");
$ARI_HELP_FEATURE_CODES['*90'] = _("Call Forward Busy Activate");
$ARI_HELP_FEATURE_CODES['*91'] = _("Call Forward Busy Deactivate");
$ARI_HELP_FEATURE_CODES['*92'] = _("Call Forward Busy Prompting Deactivate");
$ARI_HELP_FEATURE_CODES['*52'] = _("Call Forward No Answer/Unavailable
Activate");
$ARI_HELP_FEATURE_CODES['*53'] = _("Call Forward No Answer/Unavailable
Deactivate");
$ARI_HELP_FEATURE_CODES['*70'] = _("Call Waiting - Activate");
$ARI_HELP_FEATURE_CODES['*71'] = _("Call Waiting - Deactivate");
$ARI_HELP_FEATURE_CODES['*78'] = _("Do-Not-Disturb Activate");
$ARI_HELP_FEATURE_CODES['*79'] = _("Do-Not-Disturb Deactivate");
$ARI_HELP_FEATURE_CODES['*97'] = _("My Voicemail");
$ARI_HELP_FEATURE_CODES['*98'] = _("Dial Voicemail");

###############################
# settings page settings
```

```
################################
#
# protocol config.
#    config_file options: semi-colon delimited list of extensions
#
$ASTERISK_PROTOCOLS = array();
$ASTERISK_PROTOCOLS['iax']['table'] = "iax";
$ASTERISK_PROTOCOLS['iax']['config_files'] =
"/etc/asterisk/iax.conf;/etc/asterisk/iax_additional.conf";
$ASTERISK_PROTOCOLS['sip']['table'] = "sip";
$ASTERISK_PROTOCOLS['sip']['config_files'] =
"/etc/asterisk/sip.conf;/etc/asterisk/sip_additional.conf";
$ASTERISK_PROTOCOLS['zap']['table'] = "zap";
$ASTERISK_PROTOCOLS['zap']['config_files'] =
"/etc/asterisk/zapata.conf;/etc/asterisk/zapata_additional.conf";

# Settings for Follow-Me Select Boxes in seconds
#

$SETTINGS_PRERING_LOW = 4;
$SETTINGS_PRERING_HIGH = 30;
$SETTINGS_LISTRING_LOW = 6;
$SETTINGS_LISTRING_HIGH = 60;

$SETTINGS_FOLLOW_ME_LIST_MAX = 5;
$SETTINGS_ALLOW_VMX_SETTINGS = true;
#
# For setting
#    option: 0 - do not show controls
#            1 - show controls
#
$SETTINGS_ALLOW_CALLFORWARD_SETTINGS = 1;
$SETTINGS_ALLOW_VOICEMAIL_SETTINGS = 1;
$SETTINGS_ALLOW_VOICEMAIL_PASSWORD_SET = 1;

#
# password length
#    setting: number of characters required for changing voicemail password
#
$SETTINGS_VOICEMAIL_PASSWORD_LENGTH = 3;

#
# password exact length
#    option: 0 - do not require exact length when setting the password
#            1 - require exact length when setting the password
#
$SETTINGS_VOICEMAIL_PASSWORD_EXACT = 0;

#
# voicemail email option descriptions
#
$SETTINGS_VOICEMAIL_EMAIL_OPTION_DESCRIPTIONS = array();
$SETTINGS_VOICEMAIL_EMAIL_OPTION_DESCRIPTIONS['attach'] = _("Email voicemail as
attachment");
$SETTINGS_VOICEMAIL_EMAIL_OPTION_DESCRIPTIONS['saycid'] = _("Say caller id in
recording emailed");
$SETTINGS_VOICEMAIL_EMAIL_OPTION_DESCRIPTIONS['envelope'] = _("Say envelop
(date/time) in recording emailed");
$SETTINGS_VOICEMAIL_EMAIL_OPTION_DESCRIPTIONS['delete'] = _("Delete voicemail
when emailed");
$SETTINGS_VOICEMAIL_EMAIL_OPTION_DESCRIPTIONS['nextaftercmd'] = _("Play next
message after deleting current message");
$SETTINGS_VOICEMAIL_EMAIL_OPTION_DESCRIPTIONS['review'] = _("Ask caller to
review their voicemail before sending");
$SETTINGS_VOICEMAIL_EMAIL_OPTION_DESCRIPTIONS['maxmessage'] = _("Maximum time
in seconds a voicemail will record");

#
# Default
#    option: ".wav" - wav format
#            ".gsm" - gsm format
#
$ARI_VOICEMAIL_AUDIO_FORMAT_DEFAULT = ".wav";

#
```

```
# For setting
#   option: 0 - do not show controls
#           1 - show controls
#
$SETTINGS_ALLOW_CALL_RECORDING_SET = 1;


$SETTINGS_ALLOW_PHONE_SETTINGS = 1;


?>
```

Hopefully this will fix the problem. You will never know if you don't try.

```
# For setting
#   option: 0 - do not show controls
#           1 - show controls
```

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	stun1.noc.ams-ix.net
stun.fwd.org	stun.voipbuster.com
stun01.sipphone.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™ 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 AstTapi, which is a Microsoft TAPI 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™ Outlook and for that matter from other TAPI compliant applications.

Download AstTapi from *http://sourceforge.net/projects/asttapi/*

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.
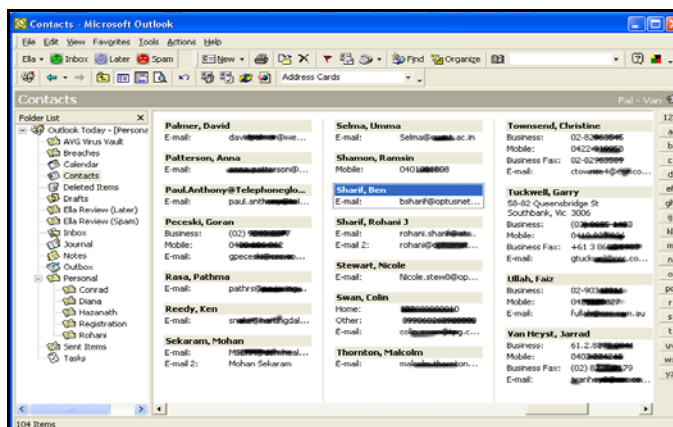
Click on the Telephone icon.

The New Call dialog box will be displayed:

Select Dialling Options.
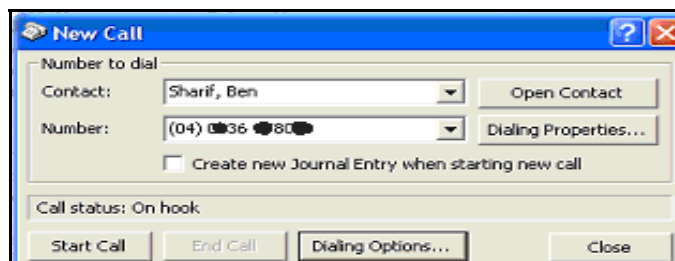
The Dialling Option box will be displayed.

In the "connect using line" drop down box, select Asterisk.

Then click on Line properties.
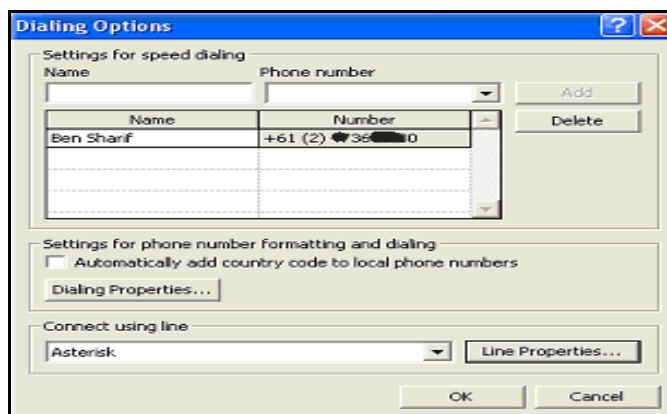
In the **Host** field, enter the address of your asterisk server: **e.g. 192.168.1.100**

In the Port Field: **5038**
User Field: **AstTapi**
Password: **AstTapi**
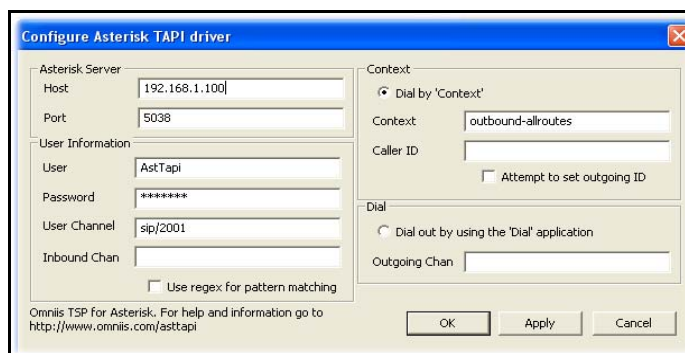
User Channel: (your extension – the extension you want Asterisk to ring before connecting you the number you are calling) e.g. `sip/2001`

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.
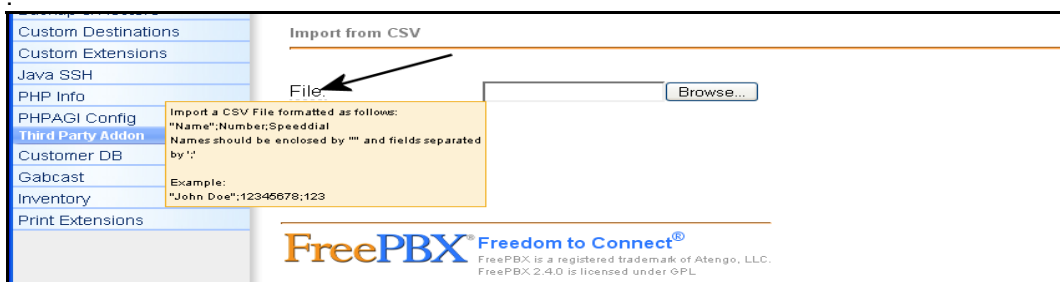
Here you simply enter the Phone number of the party you wish to call, the name and the Speed Dial Code in the appropriate fields as per the example below. Submit change when done.

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)
.

The "Import from CSV" is a misnomer because, if you have created a list using EXCEL and are 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 exists 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
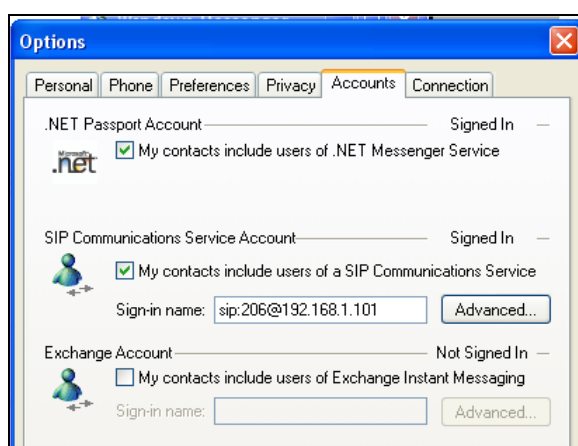
2. Click on the [Advanced] button next to the Sign-in name.
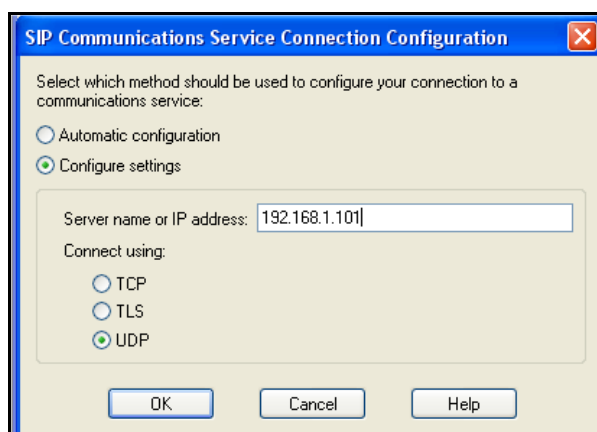
    - Check the [Configure settings] button
    - Enter Asterisk IP address (e.g.192.168.1.101) in the [Server name of IP address] field
    - Check the [UDP] button
    - Click [OK] and [OK] again.

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 DUNDi for a more complete solution). A little tutorial on DUNDi can be found here.

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.

	System 1 *def.com.au*	System 2 *xyz.com.au*
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	xxxyyy	xxxyyy
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 *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=peer	host=def.com.au *(or System 1 IP)* qualify=yes type=peer
User Context	InterOffice-In	InterOffice-In
User Details	context=from-internal host=xyz.com.au *(or IP)* type=user	context=from-internal host=def.com.au *(or IP)* type=user

	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

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. ***This method will not work for calling external numbers through the remote server.***

	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)*	XX. *(To handle all Local and remote office extensions)*
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.

This is important – Please read

If two or more boxes are connected together and you only need to call extensions that belong to those boxes and vice versa, all the 3 methods above will work fine. However if you wish to be able to call external number via any of the other boxes, you must adopt method 1 and method 2. ***Method 3 – Friend/Friend arrangement will not work.***

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
http://www.voip-info.org/wiki/view/Asterisk+-+dual+servers

If you require a complete solution tailored to your exact requirement, my advice 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, let's 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.

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 Elastix/freePBX as covered in the chapter referring to *System Recording* 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 *http://voip-info.org/wiki/view/Asterisk+sound+files*

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
```

- 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 -c1 "`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 wav 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 FreeRip 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

```
;
; Music 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
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
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)

For complete instruction, please refer to:

STREAMING MUSIC ON HOLD FOR FREEPBX 2.4 & EARLIER

First of all lets download the latest mpg123 application. via the Linux Console or via SSH, at he Linux prompt type the following commands

```
cd /usr/src
wget ftp://ftp.sunet.se/pub/os/Linux/RPMForge/dag/redhat/el3/en/i386/RPMS.dag
mpg123-1.6.2-1.el3.rf.i386.rpm
rpm -ivh mpg123-1.6.2-1.el3.rf.i386.rpm
```

(the line in italic red is one line with no break in the line)

There is a slightly newer version (1.6.2-8) but the only rpms that are available are from ATRPMS, and either they have built them incorrectly, or mpg123 in that version requires further dependencies, which are not readily available without loading further rpms with further dependencies. So stick with this version for now.

add the following lines to /etc/asterisk/musiconhold_custom.conf (if other lines are in this file, just add those lines)

```
[uptemposmoothjazz]
mode=custom
application = /usr/bin/mpg123 -q -s --mono -r 8000 -f 8192 -b 0
http://208.122.59.30:7716
```

```
[bestofeighties]
mode=custom
application = /usr/bin/mpg123 -q -s --mono -r 8000 -f 8192 -b 0
http://160.79.128.62:6618

[Classical]
mode=custom
application = /usr/bin/mpg123 -q -s --mono -r 8000 -f 8192 -b 0
http://208.122.59.30:7064

[ClassicalGuitar]
mode=custom
application = /usr/bin/mpg123 -q -s --mono -r 8000 -f 8192 -b 0
http://208.122.59.30:7070
```

(the line in italic red is one line with no break in the line)

When you have selected your list of classes you need to create directories, otherwise Freepbx will not automatically add the classes to the list of Music On Hold. So for the above classes we perform the following:

```
mkdir /var/lib/asterisk/mohmp3/uptemposmoothjazz
mkdir /var/lib/asterisk/mohmp3/bestofeighties
mkdir /var/lib/asterisk/mohmp3/Classical
mkdir /var/lib/asterisk/mohmp3/ClassicalGuitar
```

Now we make those directories accessible to asterisk.

```
chown asterisk:asterisk /var/lib/asterisk/mohmp3/uptemposmoothjazz
chown asterisk:asterisk /var/lib/asterisk/mohmp3/bestofeighties
chown asterisk:asterisk /var/lib/asterisk/mohmp3/Classical
chown asterisk:asterisk /var/lib/asterisk/mohmp3/ClassicalGuitar
```

Now that is completed, go back into Freepbx to your Music On Hold Menu and you should now see the new music categories that you have

Thats it, you will now find under your inbound routes, ringgroups, queues, the MusicOnHold options now include your new music classes.

STREAMING MUSIC ON HOLD FOR FREEPBX 2.5 AND FURTHER

As of Freepbx 2.5, they have now made it even easier to using Internet Streams by including a Streaming Category option in the Music on Hold Menu.

However we still need to add mpg123 to the system to allow it to play streams, so at the Linux console or via SSH, perform the following commands
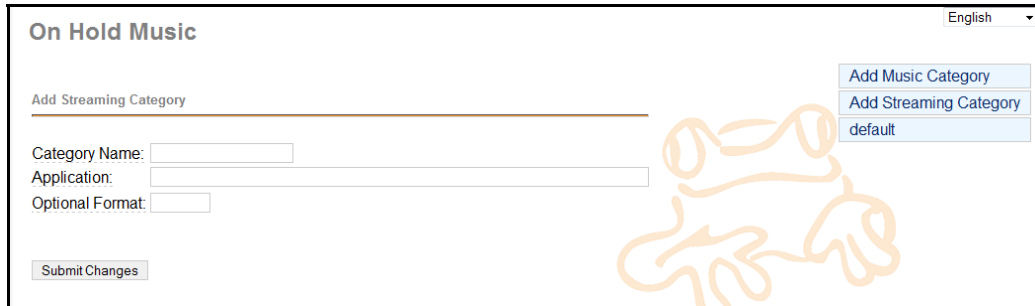
```
cd /usr/src
wget
ftp://ftp.sunet.se/pub/os/Linux/RPMForge/dag/redhat/el3/en/i386/RPMS.dag/mpg123
```

```
-1.6.2-1.el3.rf.i386.rpm
rpm -ivh mpg123-1.6.2-1.el3.rf.i386.rpm
```

(the line in italic red is one line with no break in the line)

Now let's go to the Music On Hold section in the Unembedded Freepbx
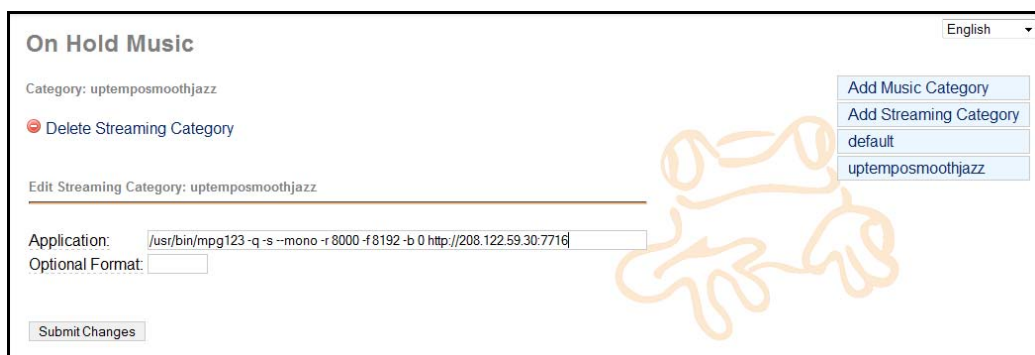
All you do is select Add Streaming Category and you will see a screen like the above.

Fill in the details similar to the example below

The Category name is like the one in the Freepbx 2.4 version, where we selected a category name of uptemposmoothjazz, and the line is /usr/bin/mpg123 -q -s --mono -r 8000 -f 8192 -b 0 http://208.122.59.30:7716, which is the Internet feed that we used in the Freepbx 2.4 version.

Submit your changes and that's it. You will end up with a new Music Category to choose from similar to below. You can set up many more as you please.

WHERE CAN I GET INTERNET STREAMS

This is where a lot of people are coming undone, or becoming confused. They read a

few tutorials about how to use Internet Streams, they go to their favourite Radio Station, find a feed and try to run it on Elastix. Many of the links now provided by the Internet radio stations download a file called something.pls. The PLS extension being most common which is a play list. This is not suitable for the command line players that are used with Linux, so you hit a stop right there....

Quite often I start off with www.shoutcast.com which has heaps of Internet feeds all at a click of a mouse. Generally I do not click on "click here to listen" but use the actual link to the radio station. Once I reach the radio stations site, I look for their links.

As a guide, I am going to pick one as they have a good selection of music types, as well as a range of feeds based on Bandwidth. For your interest, these feeds were used in the actual code for Elastix/Freepbx that were listed previously in this article.

The site I visited was http://www.sky.fm/

So on the site, I found the "Listen Now" menu, then found a Genre of music that I wanted and then selected MP3 Streams (to make life simple, the MP3 are the feeds we want), and then I found a further choice of Bandwidth. Remember we going to use it with Telephony here, not a fantastic sound system, so actually smaller is better as long as you are happy with the sound quality.

So in this case I selected 24k Modems which I find are ok on Asterisk based systems such as Elastix

If you select this, you will find that it will try and download a file with an extension called PLS. (Note: if you already have player on your Workstation system that accepts the PLS such as Winamp, you will need to save it as a file and use a file view to look at the contents. Basically the file contents will look like the following:

```
[playlist]
NumberOfEntries=3

File1=http://212.23.5.99:8002
Title1=S K Y . F M - Best of the 80s - hear your classic favorites and relive
those retro 80s!
Length1=-1

File2=http://shoutcast01.edpnet.net:8766
Title2=S K Y . F M - Best of the 80s - hear your classic favorites and relive
those retro 80s!
```

```
Length2=-1

File3=http://160.79.128.62:6618
```
Title3=S K Y . F M – Best of the 80s – hear your classic favorites and relive
those retro 80s!
```
Length3=-1
Version=2
```

(the line in italic red is one line with no break in the line)

All we are after out of this file is one of the entries. I recommend that you try each one in the list in your favourite Media Player (e.g. Windows you might use the Windows Media Player). As you might find a server that has gone offline, or reacts very slow which is not what you want. In fact this one was a very good example, and on testing found the first one didn't seem to respond at all (possibly offline) and the second was slower to start than the third. So for this example I chose the third one.

So after testing, if we take the third one in the list, it would mean our feed address for an MP3 stream for this type of music is http://160.79.128.62:6618

The above are just examples, you can add more, or change the ones above. You may want to find some low bandwidth streams that are closer to your Country to eliminate any dropouts etc.

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 developers of Elastix have 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*
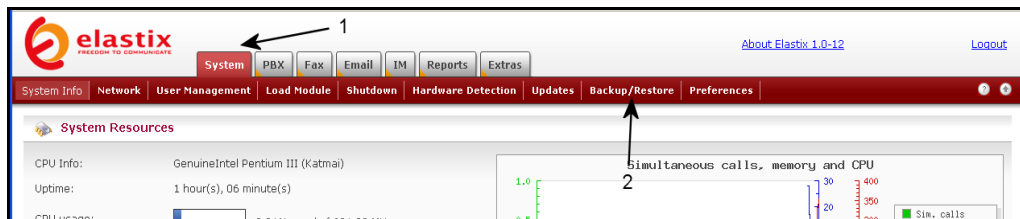
You will then be presented with the following screen:

Next select *Backup* as per the illustration above.

The screen that follows is where you define your backup requirement.
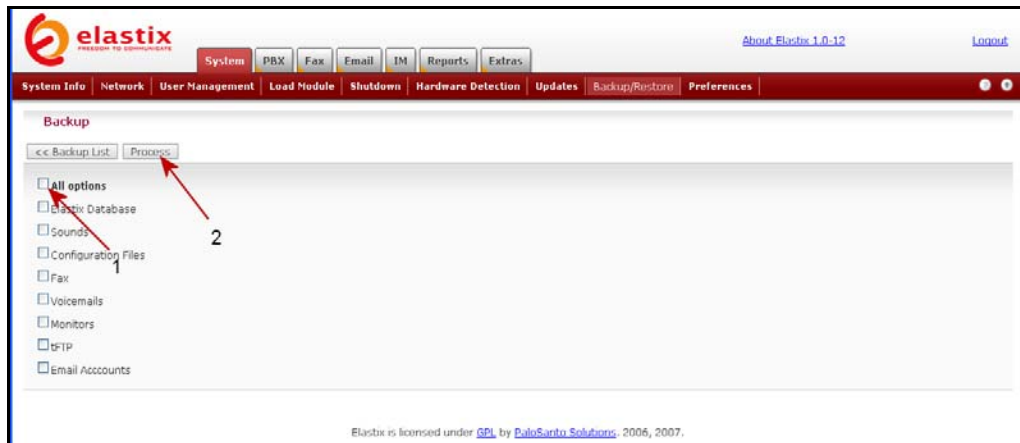
I have chosen to select ALL option as per the illustration above:

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.

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).
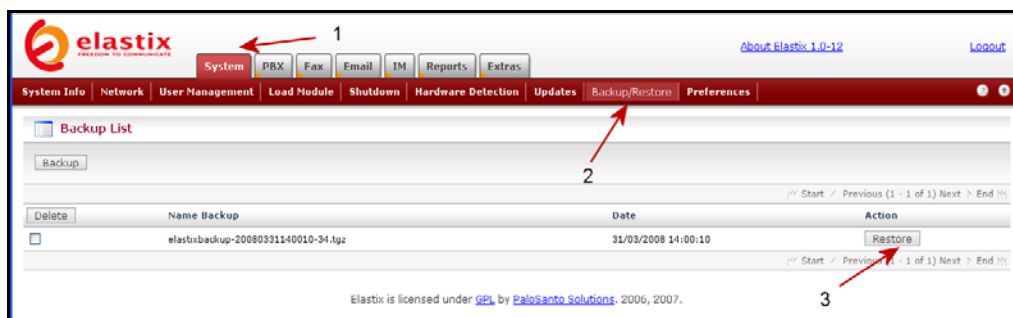
1. Select System Tab
2. Select Backup/Restore option
3. Select the backup file you wish to restore and click on the restore button
4. As per the illustration below:
    a. Select whatever components that you want to restore
    b. Click on the Process button

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*
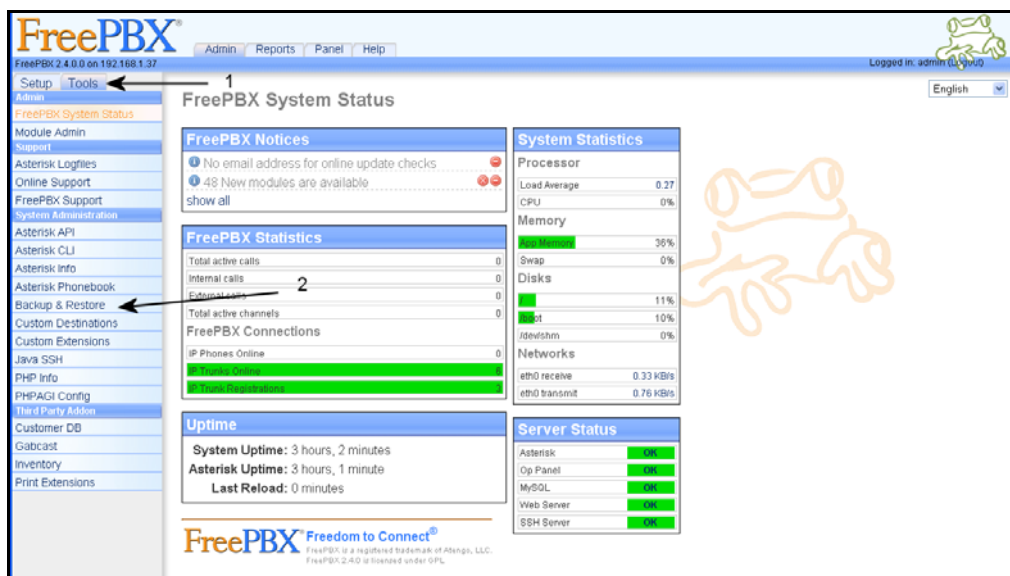
You will then be presented with the following screen:

Next select **Add Backup Schedule**

The screen that follows is where you define your backup requirement.
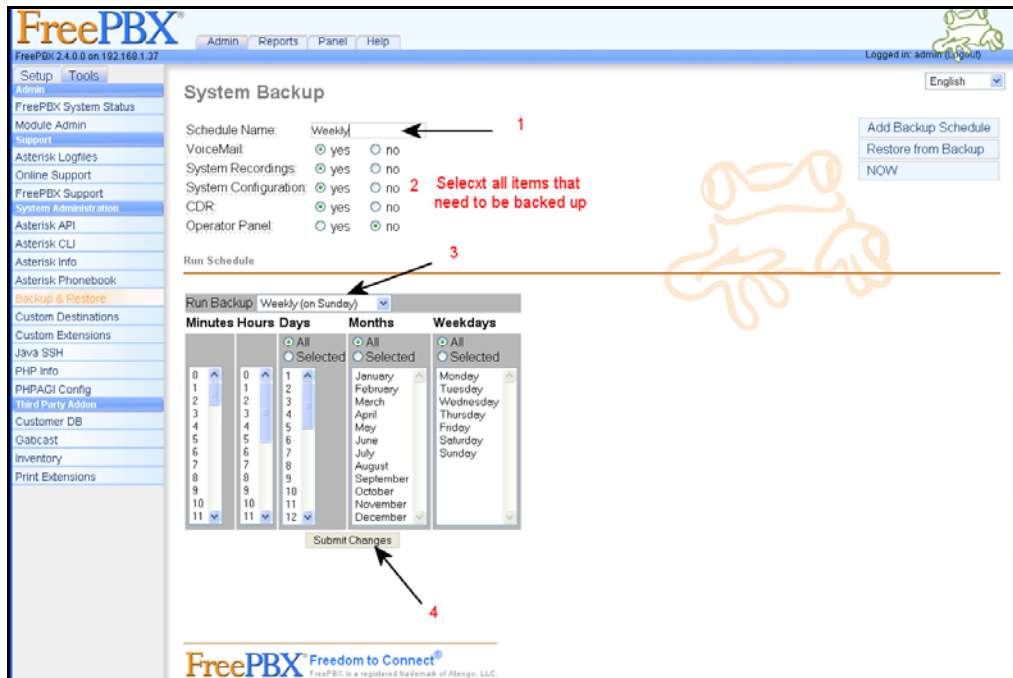
I have chosen to schedule a *Weekly (on Sundays^t)* backup as per the illustration above:

- 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.

Select the backup schedule you want to restore from.

Select the backup set.

Determine which of the various items in the list you want to restore and click on it and your restore will be executed.
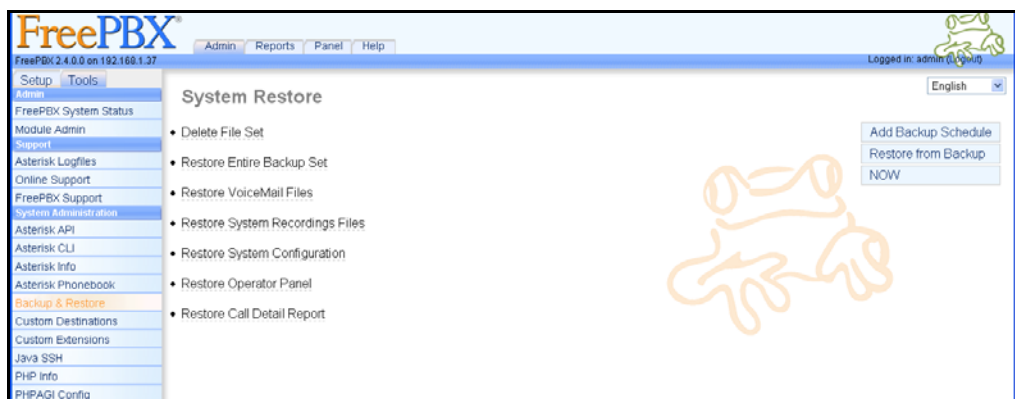
34 SUGARCRM

SugarCRM is a contact management software that comes bundled with Elastix and 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 **click here** link.

(see illustration below)

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 reasons 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:

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:

```
$number =  preg_replace ( "/[^\d\*]/", "", $number );
```

To

```
$number =  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.
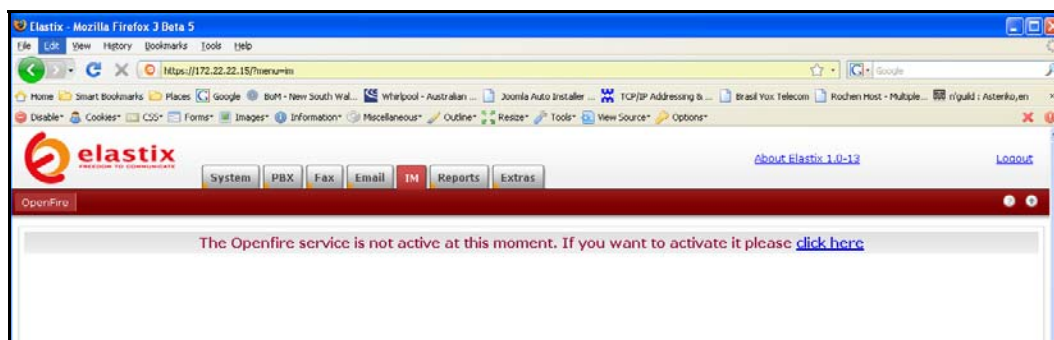
Click on the CLICK HERE hyperlink and the following screen will appear after a short wait (it is setting up the application for use).
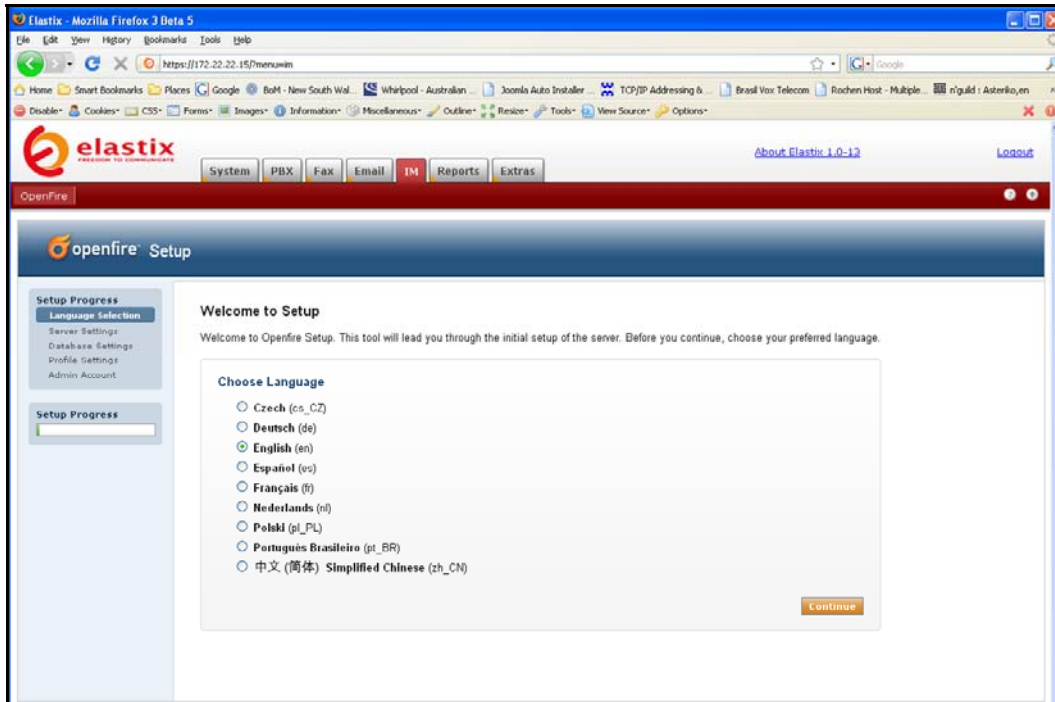
Select the language that you want to use and click on Continue and the next screen will appear
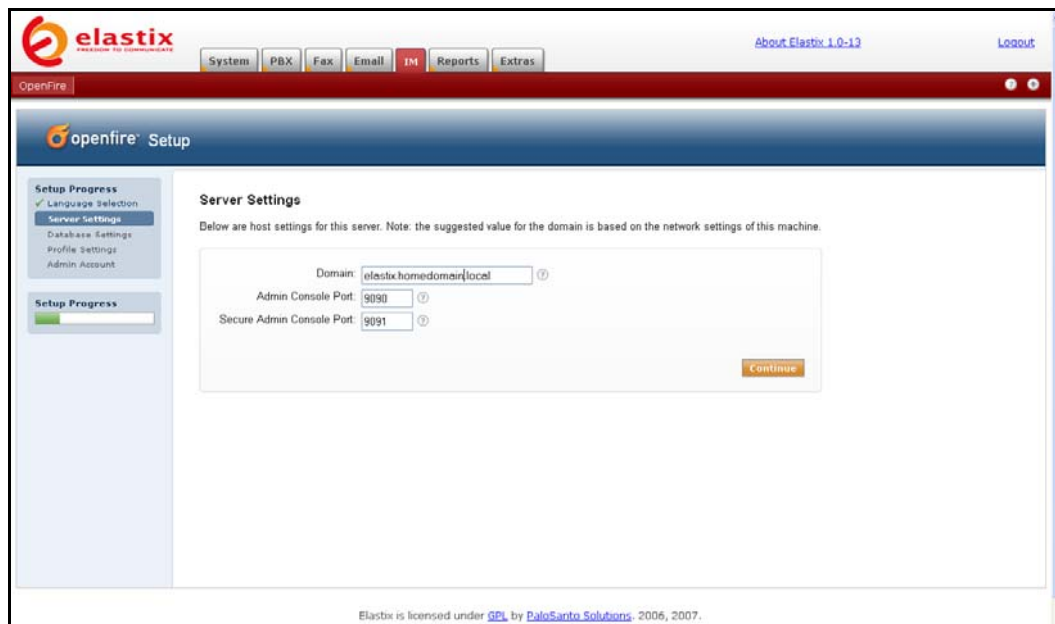
This screen lets you setup your domain name for your system and must match the domain name of your Elastix System. If you haven't setup a domain name correctly on your Elastix system, then drop out of this installation program and do it now. In this guide, you will find a Brief Guide to Networking with 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
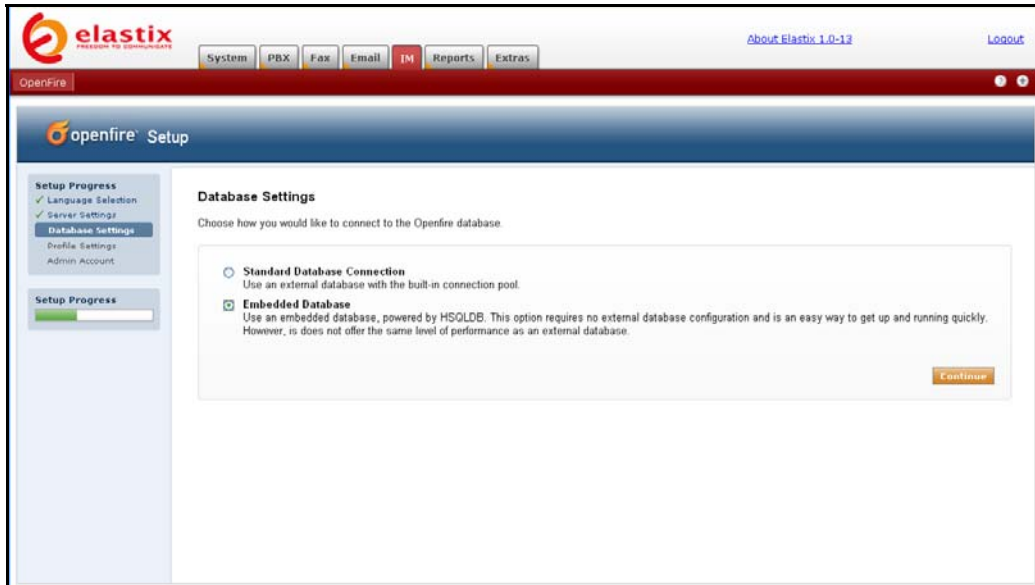
In this screen, you have two choices. The first one is a standard Database connection, which unless you have done a large amount of database set-ups, including Java nomenclature, then avoid this.

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.
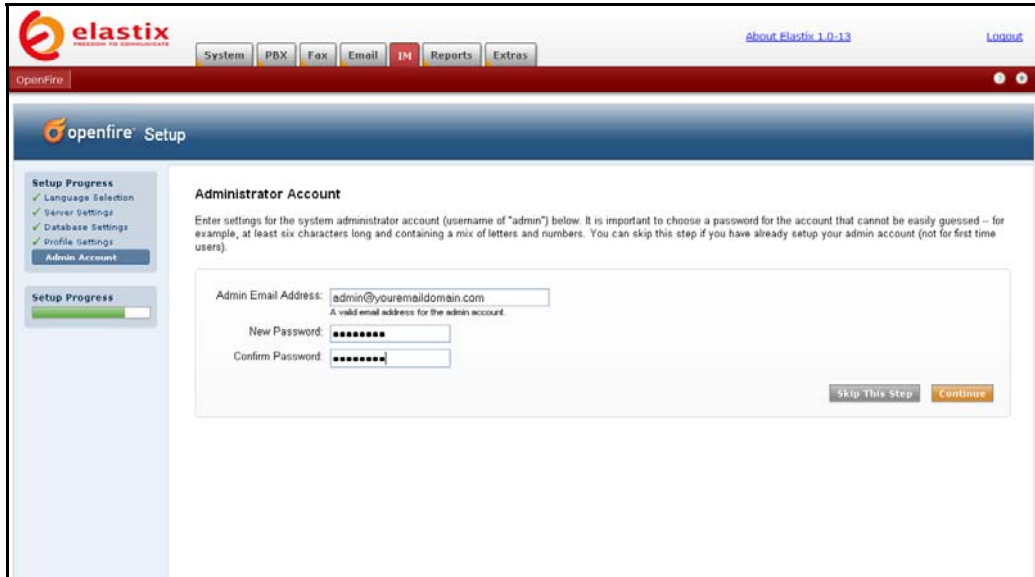
So the recommendation is select the Embedded Database, and click on Continue.
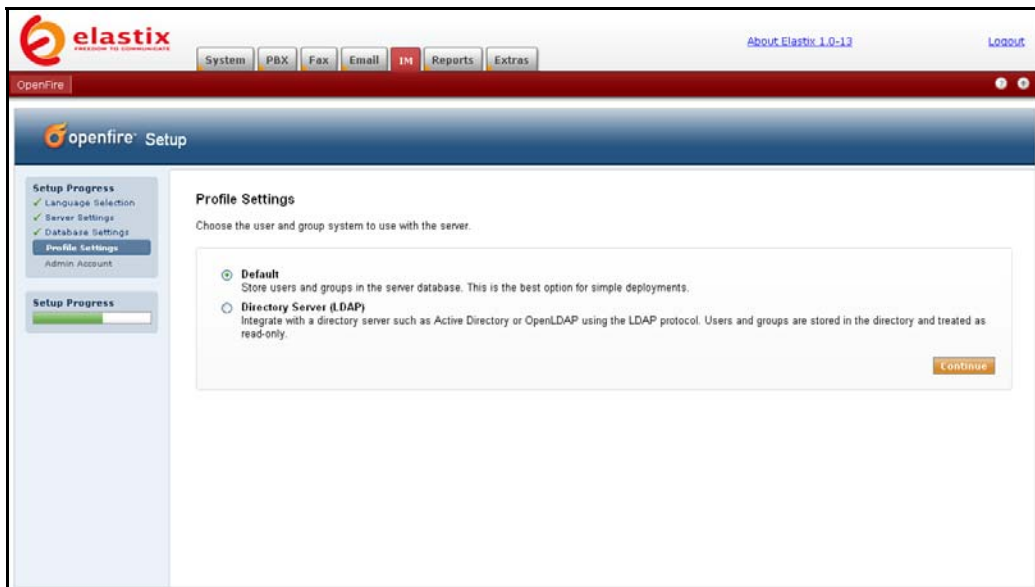
This screen allows you to choose where to store your users and groups. For most you will select default, using the database to store your users and groups, which is fine for most set-ups.

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

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.

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.

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.
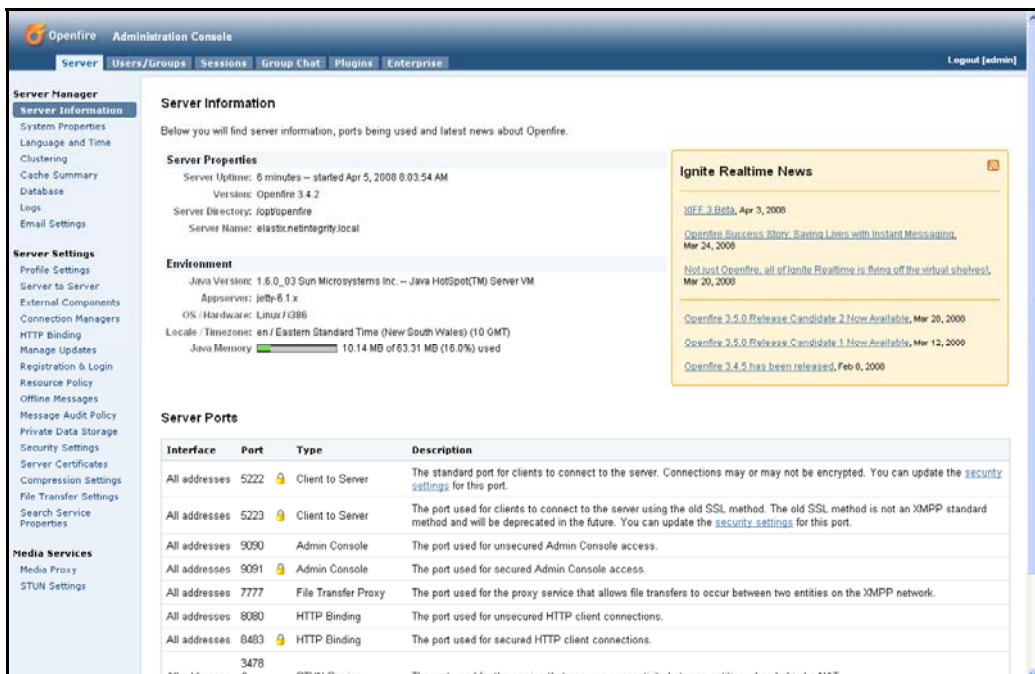
Don't panic, there is a lot on this screen, but we will take you through the few basics that you need. Don't upgrade Openfire yourself; these updates will come from Elastix. Remember that they have integrated it into the Elastix system, and there may be minor tweaks to allow this to happen. These minor 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
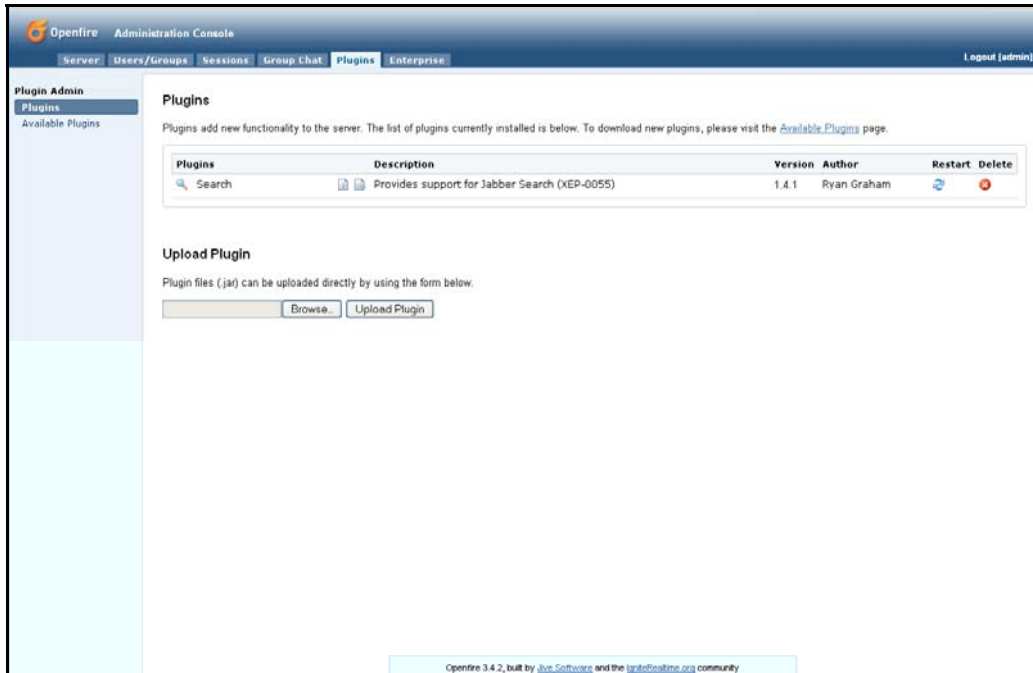
This screen shows that there is only one Plugin installed by default. We are now going to quickly add a few more.

So click on Available Plugins from the side menu and the following screen will appear.
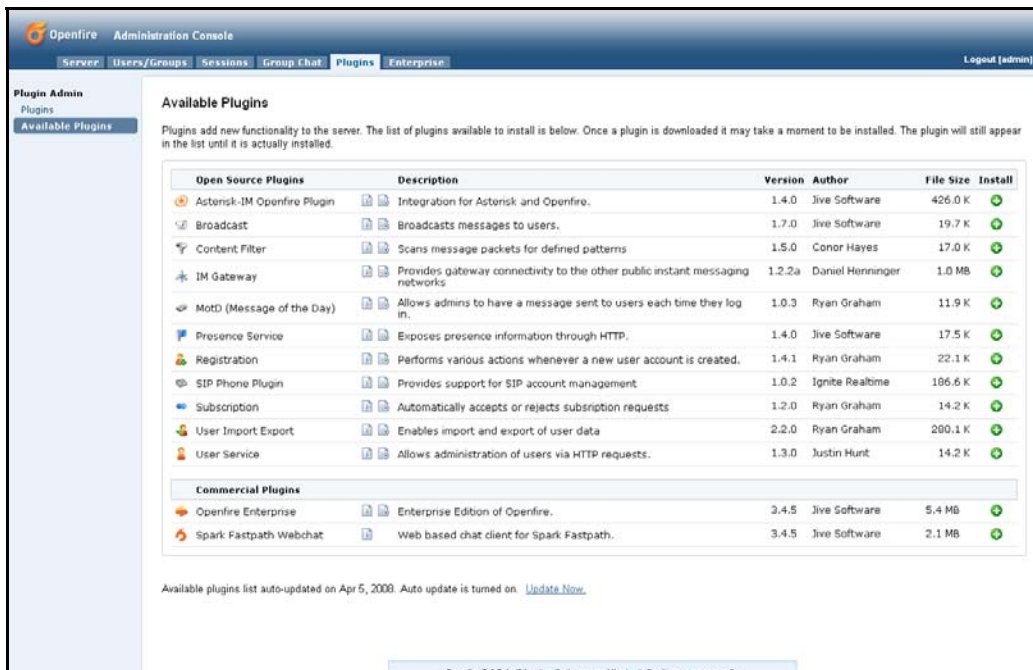
What you see in this list is a list of all Plug-ins that have been registered with Openfire. You may find this list increase, allowing you to add another new plug-in that might be

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.

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.
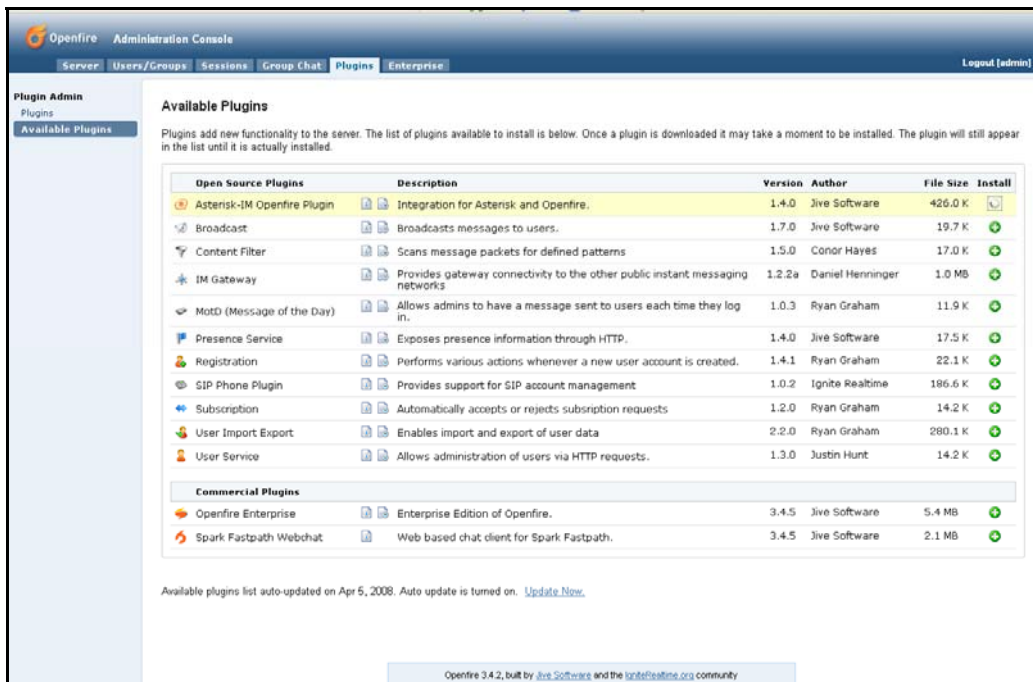
You will note that the Green Plus sign turns into a little "Loading Timer", and may take a short while to install. Once completed, it will automatically remove itself from the screen, as it has now ended up in the Plugins

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.

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.

Click on the Asterisk-IM Enablled 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

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

```
[root@elastix /]#
[root@elastix /]#
[root@elastix /]# cd /opt
[root@elastix opt]# cd openfire
[root@elastix openfire]# cd plugins
[root@elastix plugins]# cd asterisk-im
[root@elastix asterisk-im]# cd database
[root@elastix database]# nano asterisk-im_hsqldb.sql
```

You will get a screen very similar to the following

```
create table phoneServer (
    serverID bigint not null,
    serverName varchar(255) not null unique,
    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)
);

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);




                          [ Read 28 lines ]
^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
```

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                    Modified

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)
);

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

```
  GNU nano 1.3.12              File: asterisk-im_hsqldb.sql                    Modified

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)
);

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);




Save modified buffer (ANSWERING "No" WILL DESTROY CHANGES) ?
 Y Yes
 N No              ^C Cancel
```

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: asterisk-im_hsqldb.sql                Modified

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)
);

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);




File Name to Write: asterisk-im_hsqldb.sql
^G Get Help        ^T To Files        M-M Mac Format        M-P Prepend
^C Cancel          M-D DOS Format     M-A Append            M-B Backup File
```

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.
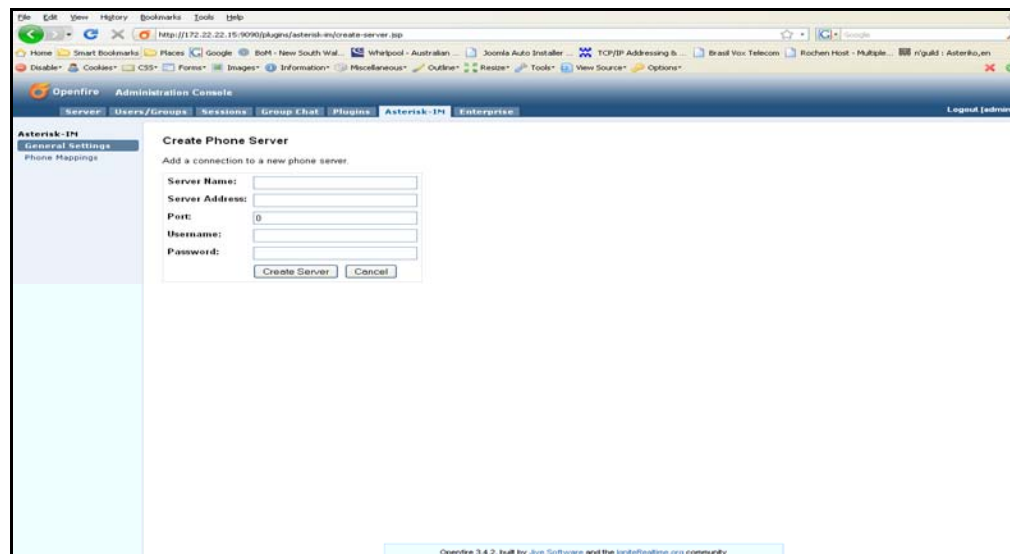
Add the following items into the input boxes

Server Name:	**elastix.yourdomain.local**	**(Same as your Elastix box domain)**
Server Address:	**127.0.0.1**	**(Always this for this install)**
Port:	**5038**	**(This is the Asterisk AMI Port number)**
Username:	**admin**	**(this is a user in manager.conf)**
Password:	**elastix456**	**(this is the default password)**

So you screen when filled in, should look this

Click on Create Server and if everything is successful, you should see the following screen

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

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.
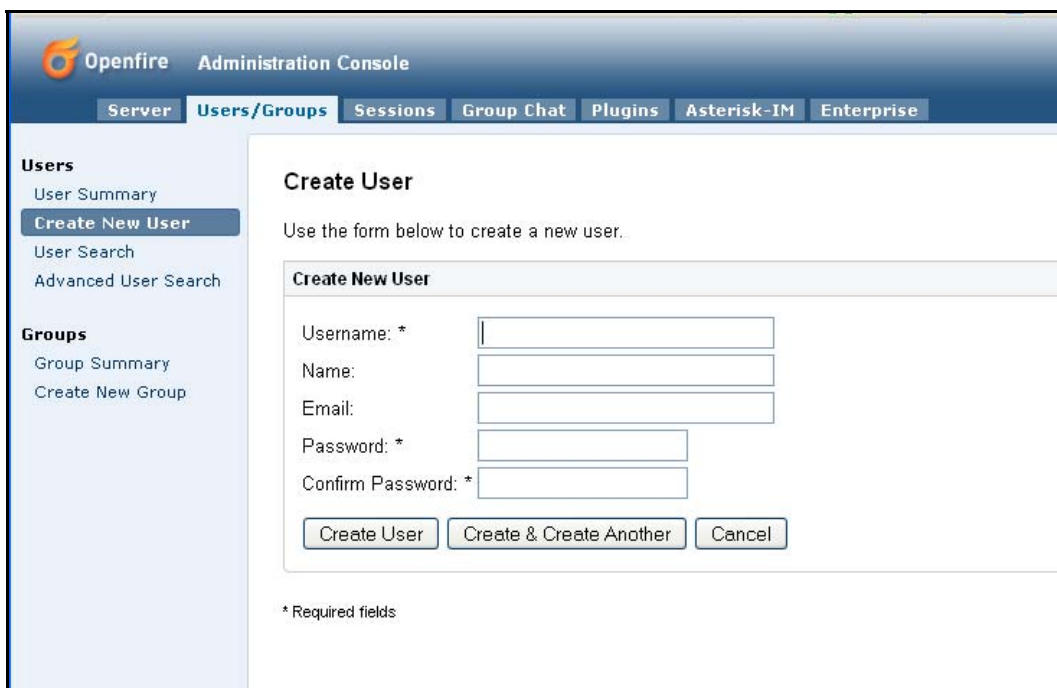
Fill in the details for each user you want to connect to Openfire. Keep the usernames in lower case (makes it easier), fill in their proper name, their current email address, and provide them with a password. This password does not have to match anything, it will be used by the client that resides on their desktop to connect to the Openfire server.

You should now have a screen like the illustration below:

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

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:
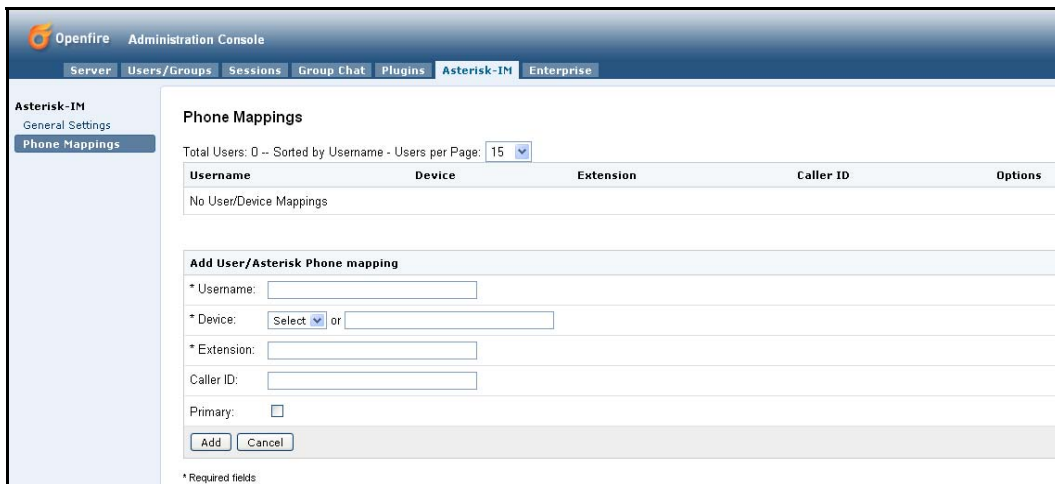
We are now going to setup a phone mapping for the user that we just setup under Users and Groups.

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.

After you have pressed the add button, you should see the following user setup
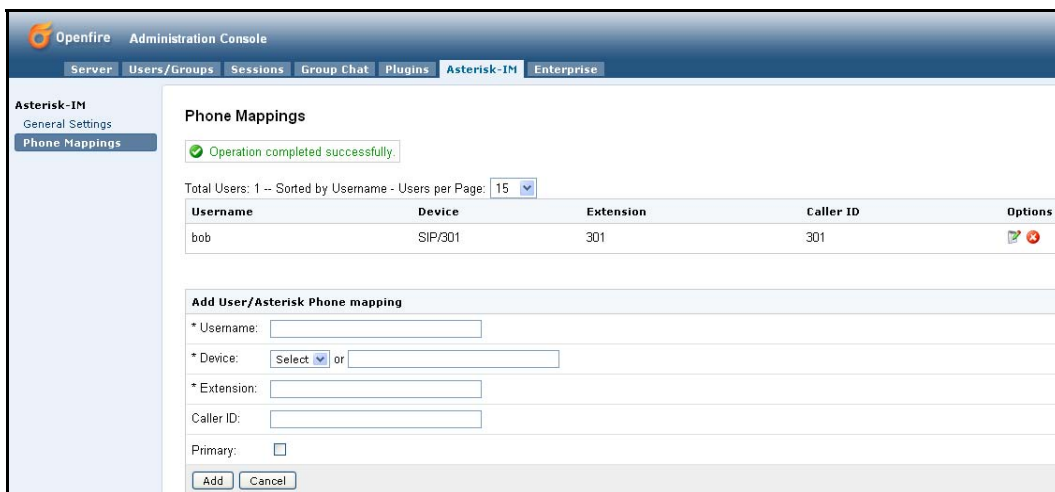
You have now successfully mapped a user and phone together.

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
http://www.igniterealtime.org/projects/spark/index.jsp

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

To enable remote client, do the following:

Username: **bob**
Password: (the password you setup for that user)
Server: Your server DNS (or DynDNS)

In my case, my DynDNS name is:
pcnovation.homelinux.org

**You must open your Router port 5222 and direct
it to your Elastix IP address.**

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
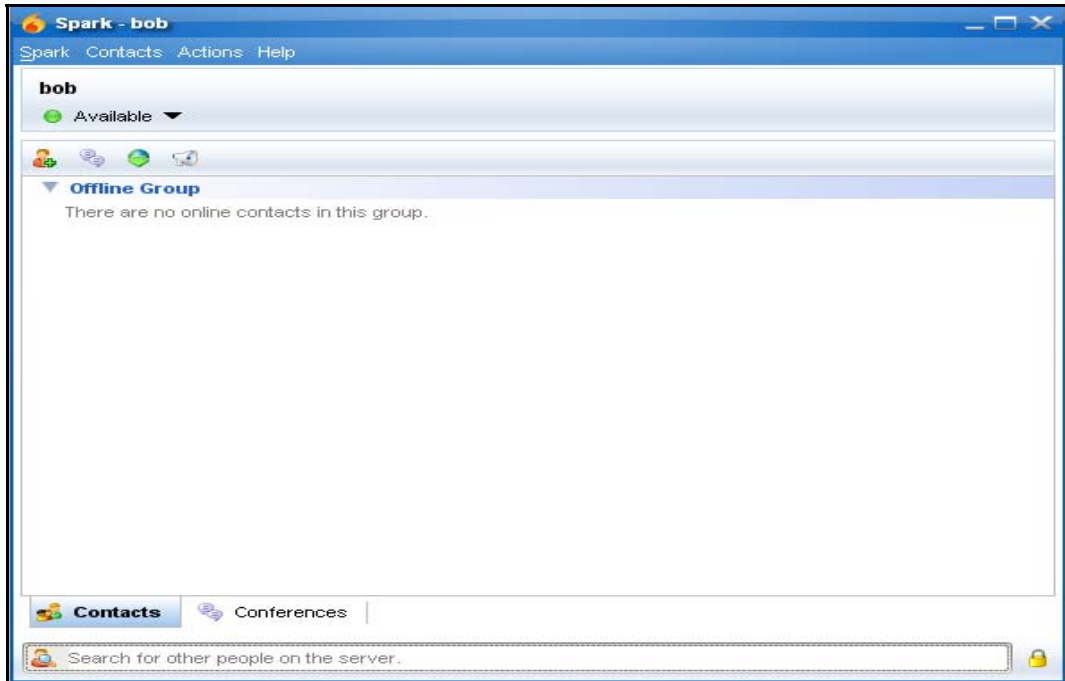
Our client has now successfully connected to the Openfire server. If we had more users installed, we would see the users listed, showing their presence status, whether they are offline, on the phone, away from keyboard etc.

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.
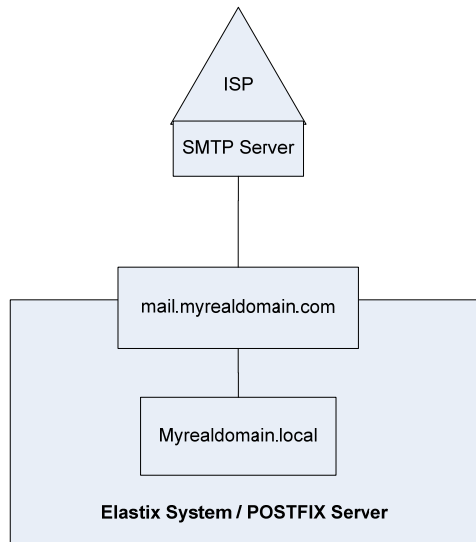
Not the best of diagrams I must admit, but hopefully it will give you a slightly more conceptual idea of what we are discussing.

It is important that we get that piece understood, as I have seen so many incorrect or very flawed installations, based on the misunderstanding of this concept, which eventually lead to kludges and fixes, that finally breakdown and leave the person feeling like the system is unreliable. Getting your wires crossed on the domain name issue can cause you untold internal strife.

37.2 EMAIL DIRECTLY VIA POSTFIX

By default, this is the first one everyone tries, because the minute that you configure faxes to go to an email address or you setup Voicemail to Email, it tries to use the Postfix Server to send out the email.

For many, this is also where it fails. It can fail for the following reasons:

1. The ISP Blocks port 25 for sending email directly in an effort to force you send the email to route through their SMTP Server. What they do is block port 25 to all addresses except their own mail server.

2. You ISP mail host rejects your mail, because it is leaving your system with your local domain name as the sender. It rejects it as it is not a valid domain.

3. Even after you make some configuration changes, your ISP rejects the email because they want you to authenticate, in other words use SMTP-Auth.

4. Another peculiar one, which people get confused with, as only some of their mail is rejected and other emails get through is the ISP/Webhost use of the POP before SMTP. What this means is that instead of putting on SMTP Auth, they work on the basis that your normal Mail client that you run on your workstation, normally performs a receive and then performs a send straight after. Basically they use the POP3 receive mail (which usually involves your mail client providing your login and password to receive email), and as long as an SMTP send is performed with 30 seconds of the POP3 receive, from the same IP address, they will let the SMTP mail out, without Authentication. The issue arises in that with your Postfix server, you are not performing a receive at anytime using POP3, so this method will normally fail. What becomes confusing is that sometimes it will go

through, and this is usually, especially in the case of a network that uses external 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.

```
################################
#Ingresado por yb-webadmin
mydomain =myrealdomain.com.au
myhostname = elastix.myrealdomain.local
```

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 there 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

```
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
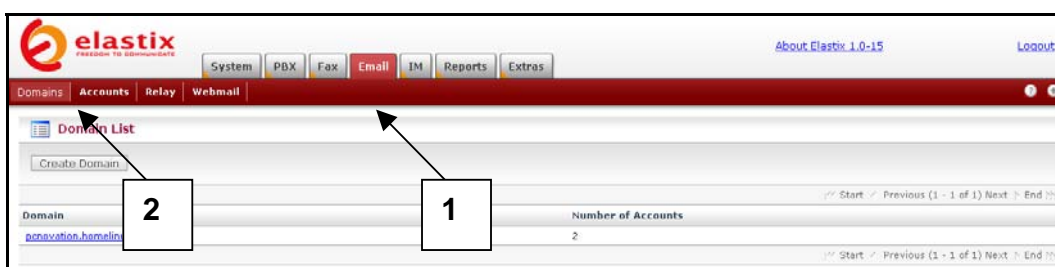
You should see an empty domain list, which is correct. Click on the Create Domain button.

You will now see the next screen asking for the domain name.
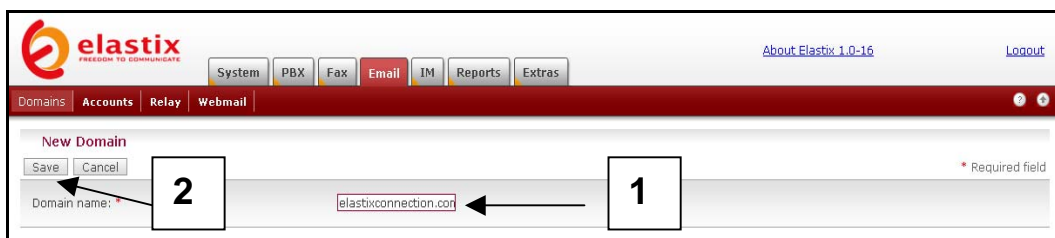
Now, you should have a domain name registered by now, and for this example, we are going to use elastixconnection.com as the domain name that we have just registered. Naturally you replace this with the one that you hopefully have registered.

So in the box, we are going to type **elastixconnection.com** and click on SAVE and the following screen will appear

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.
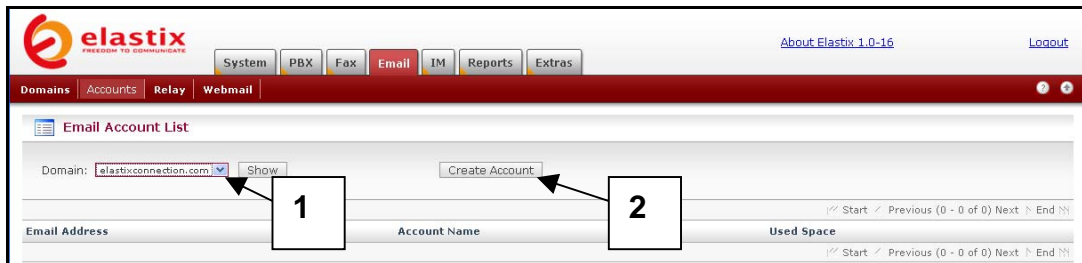
Now click on Create Account button, which will provide you with the following screen.

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.

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.
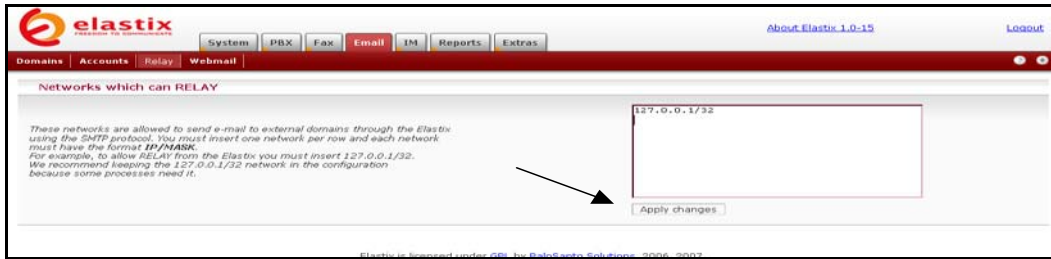
Now click on the Webmail Tab and the following screen will appear.
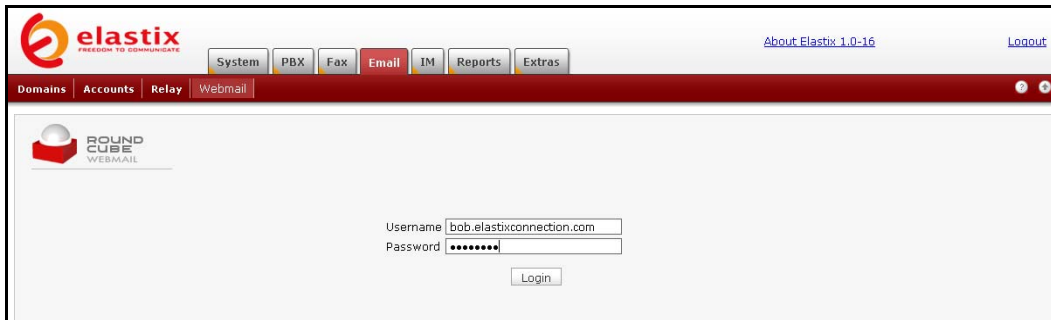
Enter in the login name that you previously setup in the email account list and the password and click on the Login button. The following screen will appear.
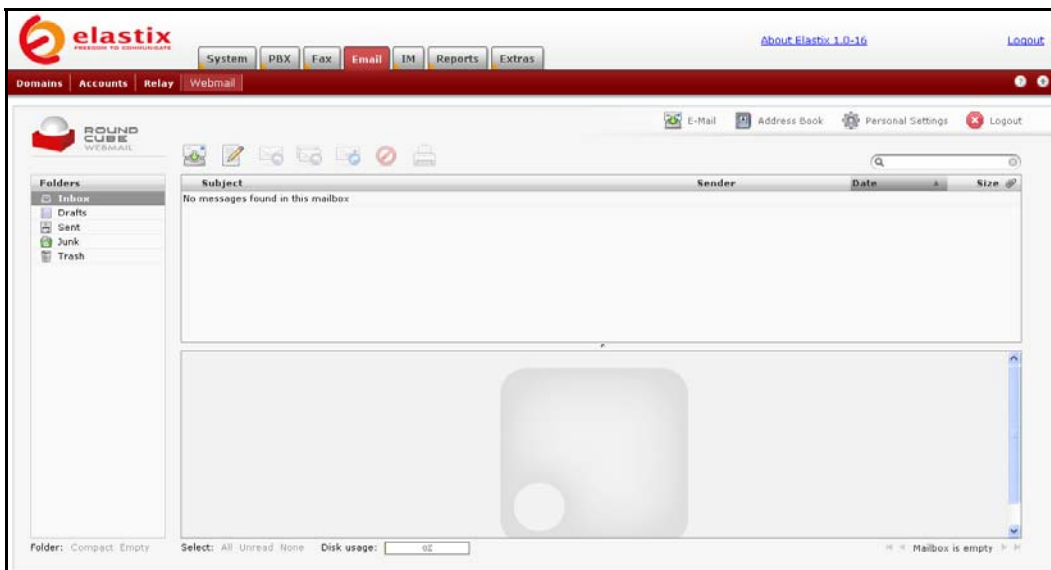
You are now looking at your Webmail screen. Before you get too excited and start sending emails, we still have a little work to do.

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.
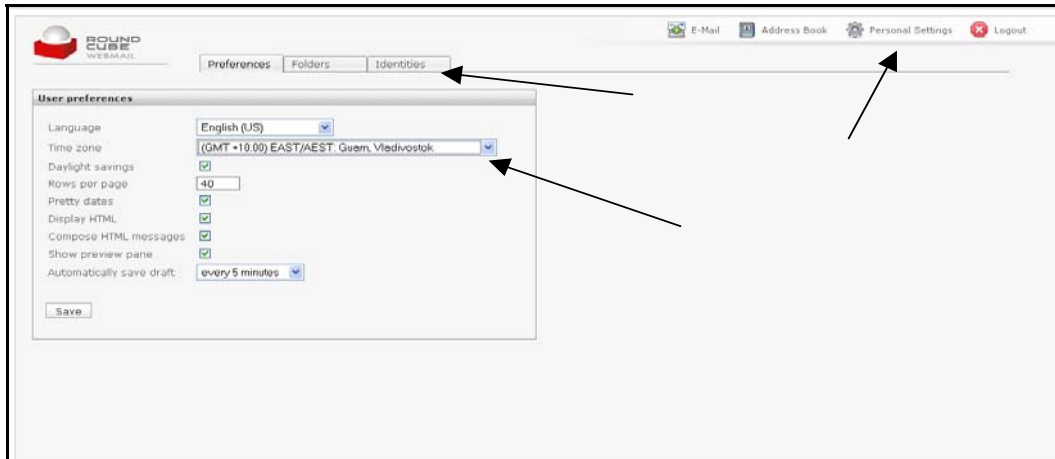
Correct the Time Zone information for area and click save. Now click on the Identities Tab and the following screen will appear.

You will note that an email account is setup, bob.elastixconnection.com@localhost instead of bob@elastixconnection.com, - 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.
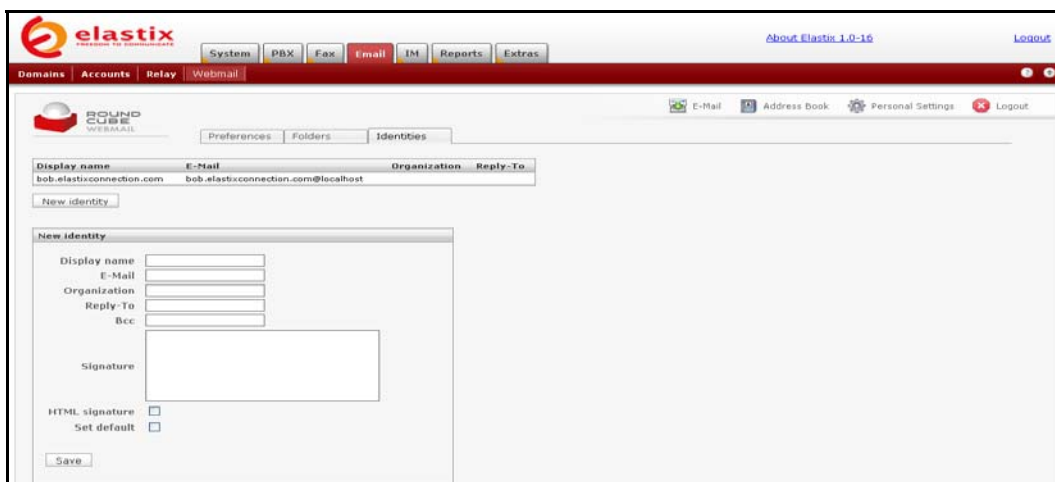
Now the details we will fill in here are:

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.

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.

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 o f 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 will not 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:       6
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 sse
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 *http://kvin.lv/pub/Linux/Asterisk/* 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 *http://kvin.lv/pub/Linux/Asterisk/README*

Now that we know what file we need you can now download the appropriate file.

Change folders to /usr/lib/asterisk/modules

```
# 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

```
;############################################
; To get outside line through ZAP - Ben 22-12-2005
; after dialling 0 will get a second dial tone.
; Then dial the number required.
; No need to press the send button after the second number dialled
;############################################

exten => 0,1,Playback(start-dial-after-dialtone)
;exten => 0,2,Dial(ZAP/1/{EXTEN}) ; < -- if using zaptel
exten => 0,2,Dial(DAHDI/1/{EXTEN}) ; < -- if using DAHDI
exten => 0,3,Hangup
```

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*

```
;############################################
; To get outside line through SPA3K - Ben 25-12-2005
; after dialling 0 will get a second dial tone. Then dial the
```

```
; number required. No need to press the send button again
;###########################################
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 hang-up"

```
;########################################################
; Give priority to emergency number by disconnecting existing user
; if using the PSTN line first, otherwise simply call the emergency number.
;########################################################
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
```

**Change Zap to DAHDI if you are using DAHDI instead of Zaptel

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

1) 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
02XXXXXXXX

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: 04XXXXXXXX
Trunk Sequence: Local/$OUTNUM$@outbound-mobile-calls/n

Local Route
Dial prefix: NXXXXXXX
             02XXXXXXXX
Trunk Sequence: Local/$OUTNUM$@outbound-local-calls/n

National route
Dial prefix: 0[356789]XXXXXXXX

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 http://presstheinter.com/part1.conf.

```
[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}-
${MATH(${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-
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.conf 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(__route2.000000=OUTBOUNDROUTE_2_CONTEXT_NAME_HERE)
exten => _.,n,Set(__call-limit-route2.000000=100)

exten => _.,n,Set(__route3.000000=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 (*http://presstheinter.com/part3.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=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-route3.000000=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)

            [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-route2.000000=300)

            exten => _.,n,Set(__route3.000000=outrt-006-gotalk3)
            exten => _.,n,Set(__call-limit-route3.000000=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)
            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-route2.000000=300)

            exten => _.,n,Set(__route3.000000=outrt-006-gotalk3)
            exten => _.,n,Set(__call-limit-route3.000000=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. If you do you must also edit rtp.conf to reflect the same value.

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

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™ WRT54G®. 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.

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.

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
```
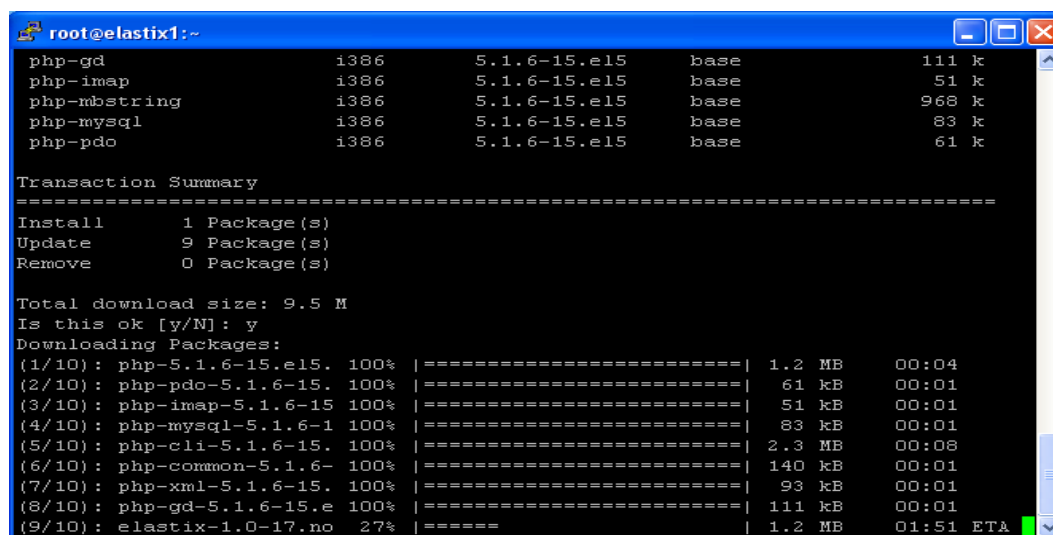
This will only take a few minutes and it gives the advantage of the process looking for the fastest mirror which will reduce your online update time by a factor of up to 80%.

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       0 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    00:04
(2/10): php-pdo-5.1.6-15. 100% |=========================|  61 kB    00:01
(3/10): php-imap-5.1.6-15 100% |=========================|  51 kB    00:01
(4/10): php-mysql-5.1.6-1 100% |=========================|  83 kB    00:01
(5/10): php-cli-5.1.6-15. 100% |=========================| 2.3 MB    00:08
(6/10): php-common-5.1.6- 100% |=========================| 140 kB    00:01
(7/10): php-xml-5.1.6-15. 100% |=========================|  93 kB    00:01
(8/10): php-gd-5.1.6-15.e 100% |=========================| 111 kB    00:01
(9/10): elastix-1.0-17.no  27% |======                   | 1.2 MB 01:51 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-2.7.0.tar.gz (or whatever the new
version is)
tar zxvf freepbx-2.7.0.tar.gz
cd freepbx-2.7.0
./start_asterisk start # for upgrade use "amportal start" without the quote
./install_amp
```

Once this is done, freePBX is now upgraded to version 2.7.0 and you can start managing and/or upgrading the freePBX modules.

42.3 UPGRADING ASTERISK

You may also update Asterisk using the yum command.
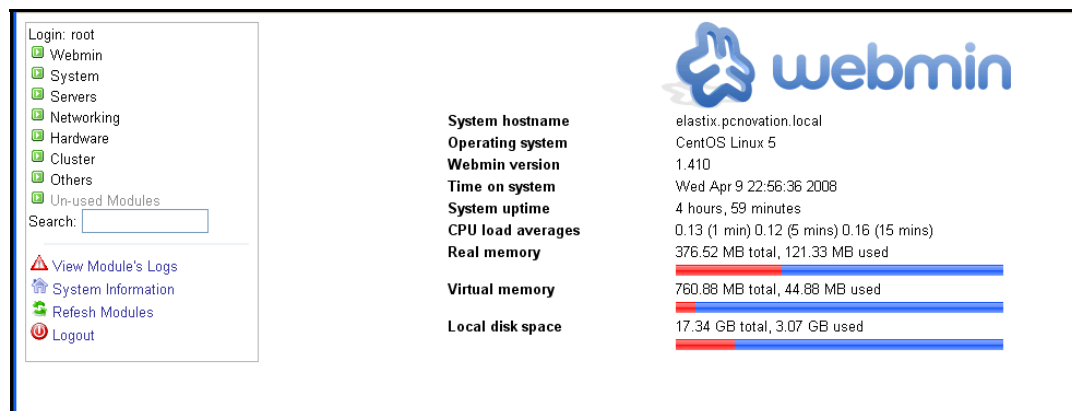
```
yum update asterisk
```

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.
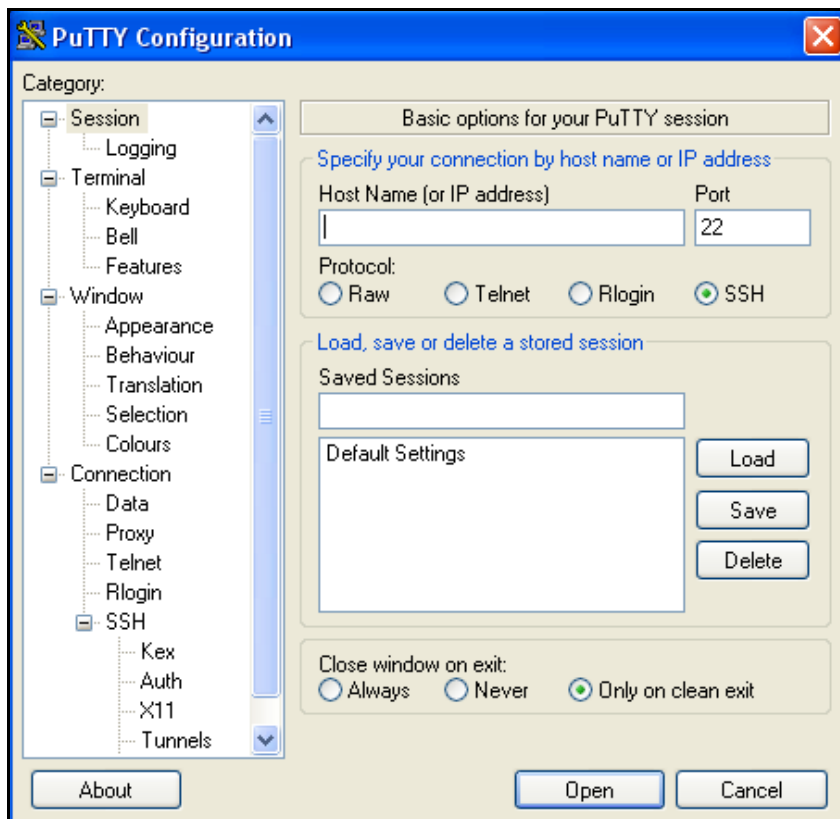
It is written and maintained primarily by Simon Tatham and can be downloaded from the following link.*http://www.putty.nl/download.html*

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.

It can be downloaded from the following link.

http://winscp.net/eng/index.php
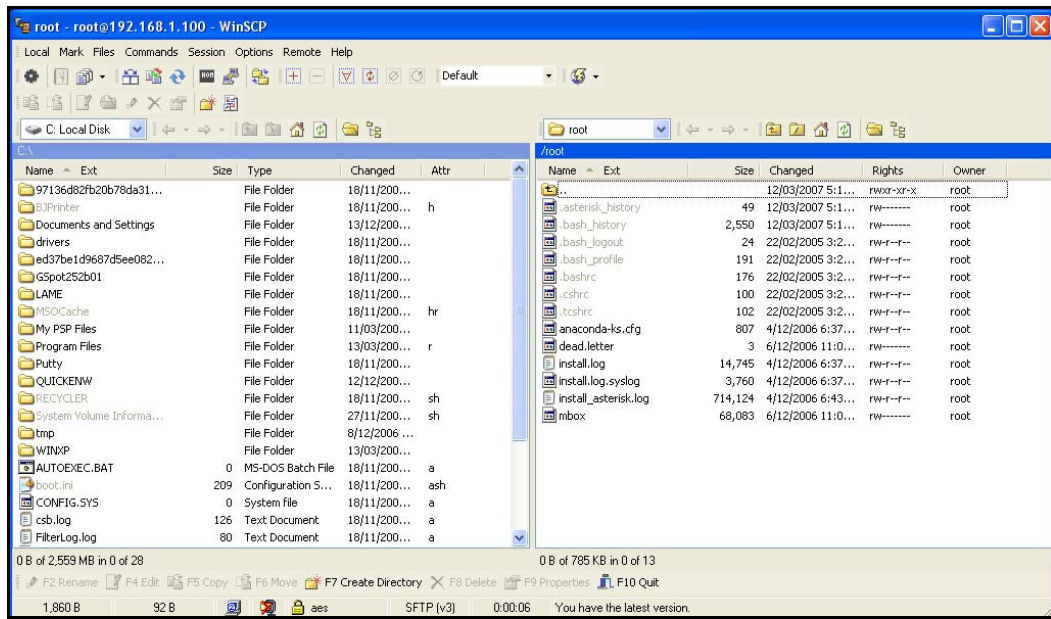
43.4 NANO

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 The phpMyAdmin 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
   <?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.2oo7* as the password and you can now start using the phpMyAdmin.

44 GET UNDER THE BONNET

> *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.
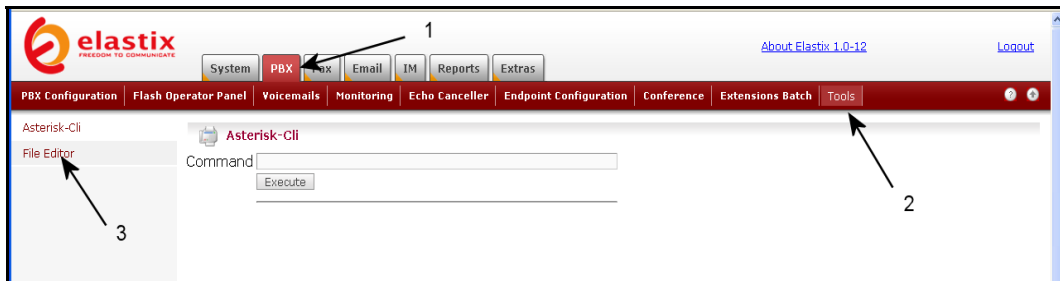
You may need to navigate through the pages to find the file you wish to edit from the resulting screen.

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.
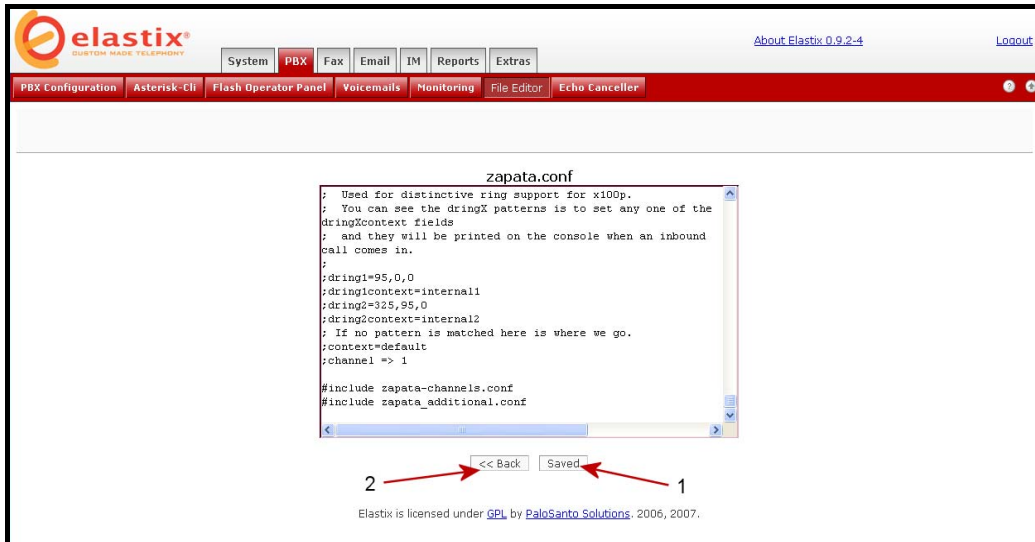
Once done with your editing, click Save and then click Back as illustrated above.

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_general_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_general_custom.conf***

44.1.1 sip.conf

There is a slight change in the structure of sip.conf from older versions of Elastix and as such there are a number of areas that we should pay attention to and changes were made to overcome previous issues. You will notice that *sip_custom.conf* is missing. If necessary you can add this conf file manually.

In the later versions of Elastix, the sip.conf looks like this:

```
;-------------------------------------------------------------------------;
;Do NOT edit this file as it is auto-generated by FreePBX. All modifications to
;this file must be done via the web gui. There are alternative files to make
;custom modifications, details at: http://freepbx.org/configuration_files
;-------------------------------------------------------------------------;
;
[general]
; These files will all be included in the [general] context
;
#include sip_general_additional.conf

;sip_general_custom.conf is the proper file location for placing any sip
;general options that you might need set. For example: enable and force the
;sip jitterbuffer. If these settings are desired they should be set the
;sip_general_custom.conf file.
```

```
;
; jbenable=yes
; jbforce=yes
;
;It is also the proper place to add the lines needed for sip nat'ing when going
;through a firewall.  For nat'ing you'd need to add the following lines:
; nat=yes , externip= , localhost= , and optionally fromdomain= .
;
#include sip_general_custom.conf

;sip_nat.conf is here for legacy support reasons and for those that upgrade
;from previous versions.  If you have this file with lines in it please make
;sure they are not duplicated in sip_general_custom.conf, if so remove them
;from sip_nat.conf as sip_general_custom.conf will have precedence.
#include sip_nat.conf

;sip_registrations_custom.conf is for any customizations you might need to do
to
;the automatically generated registrations that FreePBX makes.
;
#include sip_registrations_custom.conf
#include sip_registrations.conf

; These files should all be expected to come after the [general] context
;
#include sip_custom.conf
#include sip_additional.conf

;sip_custom_post.conf If you have extra parameters that are needed for a
;extension to work to for example, those go here.  So you have extension
;1000 defined in your system you start by creating a line [1000](+) in this
;file.  Then on the next line add the extra parameter that is needed.
;When the sip.conf is loaded it will append your additions to the end of
;that extension.
;
#include sip_custom_post.conf
```

44.1.2 sip_general_additional.conf

```
;------------------------------------------------------------------------------;
;Do NOT edit this file as it is auto-generated by FreePBX. All modifications to
;
; this file must be done via the web gui. There are alternative files to make
;
; custom modifications, details at: http://freepbx.org/configuration_files
;------------------------------------------------------------------------------;
;
vmexten=*97
disallow=all
allow=ulaw
allow=alaw
context=from-sip-external
callerid=Unknown
notifyringing=yes
notifyhold=yes
limitonpeers=yes
tos_sip=cs3
tos_audio=ef
tos_video=af41
alwaysauthreject=yes   ;       Added
allowguest=no          ;       Added
```

44.1.3 sip_general_custom.conf

Any changes you need to apply under the general context should be applied in this file.

```
bindport = 5060        ; Port to bind to (SIP is 5060)
bindaddr = 0.0.0.0     ; Address to bind to (all addresses on machine)
```

```
nat=yes
;externip=<your fixed external IP> or
externhost=netvoice.selfip.com  ; < -- Do not use this for Pennytel – leave it out
localnet=192.168.1.0/255.255.255.0
context=from-trunk
externrefresh=10
allowguest=no
language=au
videosupport=yes

;defaultexpirey=600 ; **Engin users: include this if necessary
;maxexpirey=3600       ; **Engin & BBP Global users: include this if necessary
rtptimeout=60
rtpholdtimeout=120
progressinband=yes

useragent = Elastix
```

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**

44.1.4 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>  ; < -- Do not use this for Pennytel – leave it out
localnet = 192.168.1.0/255.255.255.0

language=au
delayreject = yes
jitterbuffer = yes
mailboxdetail = yes
```

44.1.5 indications.conf

In the general section ensure the following exist:

```
[general]
country = au  ; (The default is country = us, therefore replace "us" with "au")
```

44.1.6 enum.conf

```
[general]
;
; The search list for domains may be customized.
; Domains are searched in the order they are listed here.
;
search => e164.arpa
search => e164.org
```

44.1.7 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}) <-- use this if you are using Zaptel
exten => 0,1,Dial(DAHDI/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
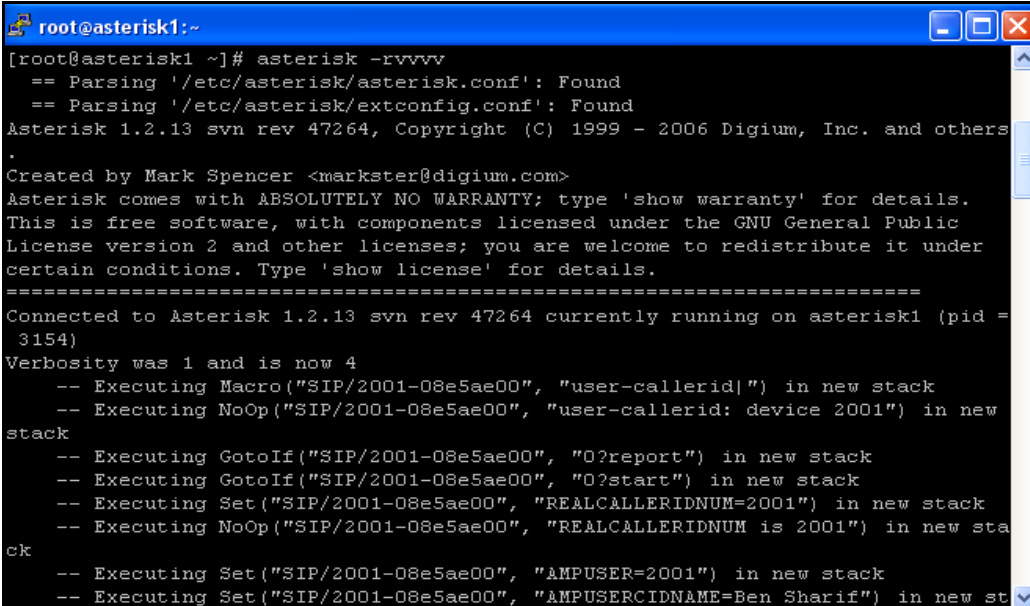
With any installation, you will find some unexpected problems. Some are inherent issues due to bugs or undocumented features, while others are unintentional mistakes.

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)

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 *'Ringtime Default'* *(Number of seconds to ring phones before sending callers to voicemail)* too short. This setting can be found in the **General Setting** under **Voicemail**. A setting of **20** would be appropriate.

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 Calls Handling 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:

```
Apr  6 04:02:49 elastix postfix/pickup[8777]: B89B7100FBF: uid=0 from=<root>
Apr  6 04:02:49 elastix postfix/cleanup[9515]: B89B7100FBF: message-id=<20080405180249.B89B7100FBF@elastix.pcnovation.local>
Apr  6 04:02:49 elastix postfix/qmgr[6758]: B89B7100FBF: from=<root@pcnovation.local>, size=1473, nrcpt=1 (queue active)
Apr  6 04:02:49 elastix postfix/smtp[9530]: B89B7100FBF: to=<root@pcnovation.local>, orig_to=<root>, relay=none, delay=31,
delays=31/0/0.01/0, dsn=5.4.4, status=bounced (Host or domain name not found. Name service error for name=pcnovation.local type=A: Host not found)
Apr  6 04:02:49 elastix postfix/cleanup[9515]: C1EC6100FC5: message-id=<20080405180249.C1EC6100FC5@elastix.pcnovation.local>
Apr  6 04:02:49 elastix postfix/qmgr[6758]: C1EC6100FC5: from=<>, size=3469, nrcpt=1 (queue active)
Apr  6 04:02:49 elastix postfix/bounce[9610]: B89B7100FBF: sender non-delivery notification: C1EC6100FC5
Apr  6 04:02:49 elastix postfix/qmgr[6758]: B89B7100FBF: removed
Apr  6 04:02:49 elastix postfix/smtp[9530]: C1EC6100FC5: to=<root@pcnovation.local>, relay=none, delay=0.03, delays=0.01/0/0.01/0,
dsn=5.4.4, status=bounced (Host or domain name not found. Name service error for name=pcnovation.local type=A: Host not found)
Apr  6 04:02:49 elastix postfix/qmgr[6758]: C1EC6100FC5: removed
Apr  6 04:07:06 elastix master[9802]: about to exec /usr/lib/cyrus-imapd/ctl_cyrusdb
Apr  6 04:07:06 elastix ctl_cyrusdb[9802]: checkpointing cyrus databases
Apr  6 04:07:06 elastix ctl_cyrusdb[9802]: archiving log file: /var/lib/imap/db/log.0000000001
Apr  6 04:07:06 elastix ctl_cyrusdb[9802]: archiving log file: /var/lib/imap/db/log.0000000001
Apr  6 04:07:06 elastix ctl_cyrusdb[9802]: archiving database file: /var/lib/imap/annotations.db
Apr  6 04:07:06 elastix ctl_cyrusdb[9802]: archiving database file: /var/lib/imap/mailboxes.db
```

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	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:	high	Min CPC Duration:	0.2
Detect Disconnect Tone:	yes		
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 LOCKING DOWN YOUR INSTALLATION

46.1 INSTALLING FAIL2BAN

The following is an Extract reproduced for those who do not have ready access to the internet (which is very few if you need this), I have reproduced the procedure as blogged by Sunshine Networks.

Your VoIP server, just like any other server that is accessible through the internet, is a potential target to hackers. Today's blog is about securing your Elastix 1.6 installation so that hackers don't get to play with your VoIP server. I won't state the obvious like setting strong passwords or changing default passwords, this is purely aimed at countering the most common way to hack SIP servers at the moment: Brute-forcing SIP passwords.
Disclaimer: This technique involves the use of the Fail2Ban script. The Fail2Ban script is not created or maintained by Sunshine Networks. The use of any blog item on their website is purely at your own risk. Like always, think before you do.

Securing Elastix 1.6 using Fail2Ban scripts:

SSH to your VoIP server and login as root, then type the following commands:

```
# cd /usr/src/
# wget http://sourceforge.net/projects/fail2ban/files/fail2ban-
stable/fail2ban-0.8.4/fail2ban-0.8.4.tar.bz2/download
# tar -jxf fail2ban-0.8.4.tar.bz2
# cd fail2ban-0.8.4
# python setup.py install
# cp /usr/src/fail2ban-0.8.4/files/redhat-initd /etc/init.d/fail2ban
# chmod 755 /etc/init.d/fail2ban
# cd /etc/fail2ban/filter.d
# touch asterisk.conf
```

Copy these contents into /etc/fail2ban/filter.d/asterisk.conf :

```
# ===================
# /etc/fail2ban/filter.d/asterisk.conf
# Fail2Ban configuration file
#
#
# $Revision: 250 $
#

[INCLUDES]

# Read common prefixes. If any customizations available -- read them from
# common.local
#before = common.conf

[Definition]

#_daemon = asterisk

# Option:  failregex
# Notes.:  regex to match the password failures messages in the logfile.
The
#          host must be matched by a group named "host". The tag "" can
#          be used for standard IP/hostname matching and is only an alias
for
#          (?:::f{4,6}:)?(?PS+)
# Values:  TEXT
#

failregex = NOTICE.* .*: Registration from '.*' failed for '' - Wrong
password
            NOTICE.* .*: Registration from '.*' failed for '' - No
matching peer found
            NOTICE.* .*: Registration from '.*' failed for '' -
Username/auth name mismatch
            NOTICE.* .*: Registration from '.*' failed for '' - Device
does not match ACL
            NOTICE.* failed to authenticate as '.*'$
            NOTICE.* .*: No registration for peer '.*' (from )
            NOTICE.* .*: Host failed MD5 authentication for '.*' (.*)
            NOTICE.* .*: Failed to authenticate user .*@.*

# Option:  ignoreregex
# Notes.:  regex to ignore. If this regex matches, the line is ignored.
# Values:  TEXT
#
ignoreregex =
# ===================
```

Add the [asterisk-iptables] section to your /etc/fail2ban/jail.conf file :

```
# /etc/fail2ban/jail.conf
#===================
```

```
[asterisk-iptables]

enabled  = true
filter   = asterisk
action   = iptables-allports[name=ASTERISK, protocol=all]
           sendmail-whois[name=ASTERISK,
dest=youremailaddress@somewhere.com, sender=fail2ban@somewhere.com]
logpath  = /var/log/asterisk/full
maxretry = 5
bantime  = 600
#====================
```

Also in /etc/fail2ban/jail.conf file you want to add your own IP address range (ours is 192.168.1.0) :

```
ignoreip = 127.0.0.1 192.168.1.0/24
```

Edit /etc/asterisk/logger.conf to add the [general] section with dateformat:

```
#====================
[general]
dateformat=%F %T
#====================
```

Reload logger module in Asterisk:

```
asterisk -rx "module reload logger"
```

Add Fail2ban to the list of startup services :

```
chkconfig fail2ban on
```

Start Fail2ban:

```
/etc/init.d/fail2ban start
```

Check if fail2ban is showing up in iptables:

```
iptables -L -v
```

You should see "fail2ban-ASTERISK" in your iptables output.

Any hackers that try to brute-force your SIP passwords will now be banned after 5 attempts for 600 seconds (see jail.conf if you want to change these values)

How to test if your security is working correctly.

Download a software SIP client and try to connect to your Elastix box using false credentials. Make sure you don't try this from an IP address that is on the "ignoreip" list (192.168.1.0/24 for instance). If your client gets blocked after 5 attempts and you receive an email saying your IP has been blocked, then you can safely assume that your configuration is working correctly.

46.2 ADVANCED FIREWALL AND BRUTE FORCE ELASTIX SETUP

In the following draft document by Dave Edwards, some references are made to Trixbox, however the methods apply equally to Elastix.

APF stands for **Advanced Policy Firewall**. This is used to control iptables on the system to allow or disallow ports to be open. APF has additional features that make it stand out above the rest. **Reactive Address Blocking** (**RAB**), QoS (TOS), direct integration with BFD, and much more—see its site for full details.

BFD stands for **Brute Force Detection**. This is used to monitor any failed logins and block IP addresses from getting in. This runs as a cron daemon and works perfectly with APF.

46.2.1 Installing APF and BFD

Installing both of these applications is very simple. You can download both of them from the R-fx Networks links, uncompress them, and then run the *install.sh* script. Tim has also created an installer script that can be downloaded to your machine and run. This will install the latest and greatest APF/BFD. To get this script, you will need to use *wget* or another method to pull it off a web server. You will want to be logged into your system as root to use these commands:

```
wget http://engineertim.com/install_apf_bfd.sh
chmod 755 install_apf_bfd.sh
./install_apf_bfd.sh
```

This will start the installation process for both APF and BFD. Once the scripts complete, you will be returned to a command prompt.

APF

Configuring APF is pretty easy, and we will look at few of the *config* file options in this section. Two of the options are covered in great detail on its web site and well-commented in the *conf.apf* file.

The *config* file for APF lives in */etc/apf* and is called *conf.apf*.

We will need to edit the *conf.apf* file. If you have multiple network interfaces on your trixbox setup, you will want to set the IFACE_IN and IFACE_OUT to your external interface. This is the untrusted network interface that is connected to the Internet. If you have a second card, eth1, that is used for internal, trusted network, you can set the IFACE_TRUSTED to this interface.

To begin editing the file, use the following command:

For those who use windows I recommend winscp http://winscp.net/eng/download.php to edit files

```
nano /etc/apf/conf.apf
```

Please see the comments in the *conf.apf* if you are uncertain.

The setup script will try to properly determine which interface is used for the untrusted network and place it in the appropriate field. It is recommended to set the value of *SET_TRIM* to *0*. This value sets the total number of rules allowed inside of the *deny trust* system. It is designed to save memory and start time. With the default value of

50, the system will start to purge old rules once this number is met. With the inclusion of BFD, this number will generally climb past 50.

Setting this value to *0* will disable this feature.

```
SET_TRIM="0"
```

APF has the ability to do QoS on packets; this is defined with the TOS values in the *conf.apf* file. For SIP and IAX, you can set the following:

```
TOS_8="21,20,80,4569,5060,10000_20000"
```

This also requires a small tweak to one of the *config* files, which we will discuss later in this article, in order to tag UDP packets.

If you changed the SSH port to a different number, we have to edit the *conf.apf* file to match this new port.

```
HELPER_SSH_PORT="2222"
```

Make sure to replace 2222 with the correct port number on which you decided to run SSH. Ingress filtering is used to open inbound ports for access; both TCP and UDP have separate settings. For a trixbox setup, the following ports should be open; both TCP and UDP are listed. If you are not using TFTP, then do not have port 69 open. Do not forget to change the SSH port from 22, to the port you choose to run SSH on, otherwise you will be locked out; here we are using port 2222 from our last example. We have not included IAX ports in this setup. There is an easy way to ensure that only specific hosts can use IAX, which we will cover later.

This is handy if you use IAX to do interoffice trunks, as I do, but don't want IAX ports open for the world to see.

```
IG_TCP_CPORTS="2222,69,80,5060,6600,10000_20000"
IG_UDP_CPORTS="69,5060,10000_20000"
```

Egress filtering is used to allow outbound filtering. I don't use egress filtering, and it will not be covered in this article. It is set to *EGF="0"*, or disabled by default. In the section of the *onf.apf* file called Imported Rules, there are settings for various feeds. Feeds are used so that many people can get information about malicious IP addresses as soon as one system reports them; this way if a script from a certain IP is attacking systems, often before the script gets a chance to get to you, your system has already blocked that IP address. Some of these feeds are very handy and I use them all. You can even set up your own custom feed that would allow you to adjust all of your servers with global deny rules. You can disable or enable this feature with the *USE_DS* setting—a *1* is enabled, a *0* is disabled.

We are now ready to start APF for the first time. If you start APF right now and something is wrong, it will disable itself in 5 minutes. This is called *DEVEL_MODE* and is the first setting in the *conf.apf* file. Leave this set to *1* until you are certain you can get in via SSH and things are working.

To save the configuration file, hit *Ctrl+O* to save and *Ctrl+X* to exit.

To see a list of command-line options, run *apf* without any flags.

```
[trixbox1.localdomain apf]# apf
apf(3402): {glob} status log not found, created
APF version 9.6 <apf@r-fx.org>
Copyright (C) 1999-2007, R-fx Networks <proj@r-fx.org>
Copyright (C) 2007, Ryan MacDonald <ryan@r-fx.org>
This program may be freely redistributed under the terms of the GNU
GPL

usage /usr/local/sbin/apf [OPTION]
```

```
-s|--start ........................ load all firewall rules
-r|--restart ...................... stop (flush) & reload firewall rules
-f|--stop........ ................. stop (flush) all firewall rules
-l|--list ......................... list all firewall rules
-t|--status ....................... output firewall status log
-e|--refresh ...................... refresh & resolve dns names in trust

Rules

-a HOST CMT|--allow HOST COMMENT ... add host (IP/FQDN) to allow_hosts.rules
and immediately load new rule into firewall

-d HOST CMT|--deny HOST COMMENT .... add host (IP/FQDN) to deny_hosts.rules and
immediately load new rule into firewall

-u|--remove HOST .................. remove host from [glob]*_hosts.rules and
immediately remove rule from firewall

-o|--ovars ........................ output all configuration options
```

To start APF, we issue the following command:

```
[trixbox1.localdomain apf]# apf –s

apf(3445): {glob} activating firewall
apf(3489): {glob} determined (IFACE_IN) eth0 has address 192.168.1.31
apf(3489): {glob} determined (IFACE_OUT) eth0 has address 192.168.1.31
apf(3489): {glob} loading preroute.rules
apf(3489): {resnet} downloading http://r-fx.ca/downloads/reserved.

Networks

apf(3489): {resnet} parsing reserved.networks into
etc/apf/internals/reserved.networks
apf(3489): {glob} loading reserved.networks
apf(3489): {glob} SET_REFRESH is set to 10 minutes
apf(3489): {glob} loading bt.rules
apf(3489): {dshield} downloading http://feeds.dshield.org/top10-2.txt
apf(3489): {dshield} parsing top10-2.txt into /etc/apf/ds_hosts.rules
apf(3489): {dshield} loading ds_hosts.rules
apf(3489): {sdrop} downloading http://www.spamhaus.org/drop/drop.lasso
apf(3489): {sdrop} parsing drop.lasso into /etc/apf/sdrop_hosts.rules
apf(3489): {sdrop} loading sdrop_hosts.rules
apf(3489): {glob} loading common drop ports

...........trimmed for this document........

apf(3489): {glob} default (ingress) input drop
apf(3445): {glob} firewall initalized
apf(3445): {glob} !!DEVELOPMENT MODE ENABLED!! - firewall will flush every 5
minutes.
```

We can see that APF has started, downloaded some rules from *dshield.org* and *spamhaus.org*, and then told us it is in *DEVELOPMENT MODE*. Now, test connecting to your server over SSH to ensure that you have set up the correct port number ingress. If you can't connect, you will have to wait 5 minutes and then APF will shutdown.

Once you are sure you can get in with SSH, we can change the *conf.apf* file from *DEVEL_MODE="1"* to *DEVEL_MODE="0"* and restart/start APF. APF will start and not warn you about being in *DEVELOPMENT MODE*; your firewall should be good to go.

APF additional tweaks

This setup might not be ideal for everyone. If you connect to your provider over IAX, then you will definitely want to add the IAX ports to the *conf.apf*. However, if you have two or more systems that you connect to each other over IAX for interoffice connections, this is the way to go. This will work with static IP addresses and DynDNS setups alike. You can use a fully qualified DNS hostname or IP address. One of the flags for the *apf* command is *-a*, which is allow. This will globally allow a host to connect to this system, bypassing the firewall rules. It can't be stressed that how handy this is. Some examples are allowing an SNMP query, IAX connections, or other ports that you do not want open, but need to allow specific hosts to connect to.

To do this, just issue the following command and substitute your remote system IP address for the one we have here.

```
apf -a 192.168.1.216
```

This will allow the system *192.168.1.216* to connect to any port on the firewalled server, thereby bypassing the firewall rules. If you are running APF on both systems, be sure to do the same thing on the other host using the correct IP address.

APF also allows a system admin to block a host or a complete subnet. This is handy if you see someone attempting to connect to your machine over FTP, Telnet, SSH, and so on. To block a specific host, use the following; be sure to use the IP address you want to block.

```
apf -d 192.168.1.216
```

To block a complete subnet (CIDR), the command is very similar:

```
apf -d 202.86.128.0/24
```

This will block the entire subnet. You can sometimes get the subnet (CIDR) listing using a WHOIS on the IP address. You can also look up a CIDR by IP on Google or ripe.net. Be sure that the subnet that you are blocking is not the one you are using or you could lock yourself out.

TOS for UDP packets are not defined for APF. Only TCP packets have the TOS bit set. There is an easy way to fix this. In the */etc/apf/internals* folder, there is a file called *functions.apf*. We need to edit this file manually. It is pretty straightforward as to what we need to change, so don't worry. There are several places where we have to add a single line. Look for the *TOS_* section in the *functions.apf* file. It will look like this:

```
if [ ! "$TOS_0" == "" ]; then
for i in `echo $TOS_0 | tr ',' ' '`; do
i=`echo $i | tr '_' ':'`
$IPT -t mangle -A PREROUTING -p tcp --sport $i -j TOS --set-tos 0
done
fi
```

We have to add the settings for UDP. We copy one line and change *tcp* to *udp*. A sample is below, highlighted.

```
if [ ! "$TOS_0" == "" ]; then
for i in `echo $TOS_0 | tr ',' ' '`; do
i=`echo $i | tr '_' ':'`
$IPT -t mangle -A PREROUTING -p tcp --sport $i -j TOS --set-tos 0
$IPT -t mangle -A PREROUTING -p udp --sport $i -j TOS --set-tos 0
done
fi
```

This additional line has to be done for all the TOS bits you are using. If you are using only TOS_8, then only worry about doing it for those. Make sure you do the *tospostroute* and *tospreroute* sections.

BFD

Brute Force Detection is used to capture illegitimate login attempts for services on the system. We see quite often a large number of SSH attempts into servers that haven't had the SSH port changed. These attempts are often an outside attempt to gain access by running dictionary attacks against common user names. These can now easily be stopped by using BFD.

If you ran the *install_apf_bfd.sh*, then BFD should be installed. The configuration file for BFD is located in*/usr/local/bfd* and is called *conf.bfd*. This file, like the one for APF, is heavily commented and covered in great detail on the R-fx Networks web site. This section will just cover some of the settings. First, this must be stated that you can become locked out of your own server if you fail to type your own password correctly. This is another good reason to add a trusted system using the *apf -a* command. You can also add a host to specifically block by adding the IP address to the */usr/local/bfd/ignore.hosts* file.

The *ban* command that BFD uses is tied directly to APF. The command is *apf -d*, which is the same as we saw to manually ban addresses and subnets. The first configuration variable we will look at is TRIG; this is the number of failed attempts before becoming banned. The default is 15, and is pretty good. Keep in mind that this is per IP address connections, not account. So if 1 IP address fails 15 times using multiple accounts, it will be banned.

Feel free to change this value if you want; I recommend not setting this above 5 to reduce the number of attempts that are allowed.

BFD has the ability to send emails out to alert of brute force attempts. This is a good idea as it will give you notice when attempts to access your system are occurring. To enable email alerts, set the value of *EMAIL_ALERTS* to *1*; then set the address you want emails to be sent to using *EMAIL_ADRESS*. You can define the subject for the email as well. This makes for easy flagging/filtering in email applications.

BFD runs from cron and places a cron entry in */etc/cron.d* called *bfd*. This runs BFD every 3 minutes. This should be acceptable for almost anyone. You can get a list of offending IP addresses using *bfd* on the command line. This is useful for looking at specific IP subnets that you might want to start blocking, if you see a pattern starting. To get this list, use the following command:

```
bfd -a
```

To start BFD, use the following command:

```
bfd -s
```

46.2.2 Summary

While there are other ways to help ensure the security of your system, we have covered some of the most important in this article. Besides a good firewall, changing access to the SSH service and adding login attempt protection to your extensions is going to go a long way in keeping hackers out of your system. Do not underestimate the importance of security; these steps can mean the difference between being secured and having someone log in and start making thousands of phone calls around the country from your phone system.

Now we need to also add some extra BFD rules for SIP and IAX

This one is for SIP wrong passwords. Save this as asterisksip

```
# failed logins from a single address before ban
# uncomment to override conf.bfd trig value
TRIG="5"

# file must exist for rule to be active

REQ="/usr/sbin/asterisk"

if [ -f "$REQ" ]
then
LP="/var/log/asterisk/full"
TLOG_TF="asterisk.sip"
TMP="/usr/local/bfd/tmp"

## ASTERISK: SIP wrong password

ARG_VAL=`$TLOG_PATH $LP $TLOG_TF | grep -i "wrong password" | grep
chan_sip | awk '{NF=NF-3} {print $NF}'| tr -d '\'\' `
fi
```

This rule is for wrong or non existent extension. Save this as asterisksip2

```
# failed logins from a single address before ban
# uncomment to override conf.bfd trig value

TRIG="5"

# file must exist for rule to be active

REQ="/usr/sbin/asterisk"

if [ -f "$REQ" ]
then
LP="/var/log/asterisk/full"
TLOG_TF="asterisk.sip2"
TMP="/usr/local/bfd/tmp"

## ASTERISK: SIP no extension

ARG_VAL=`$TLOG_PATH $LP $TLOG_TF | grep -i "No matching peer found" |
grep
chan_sip | awk '{NF=NF-5} {print $NF}'| tr -d '\'\' `
fi
```

This rule is for IAX. Save this as asteriskiax.

```
# failed logins from a single address before ban
# uncomment to override conf.bfd trig value

TRIG="5"

# file must exist for rule to be active

REQ="/usr/sbin/asterisk"

if [ -f "$REQ" ]
then
LP="/var/log/asterisk/full"
TLOG_TF="asterisk.iax"
TMP="/usr/local/bfd/tmp"

## ASTERISK: IAX2 auth failed
```

```
    ARG_VAL=`$TLOG_PATH $LP $TLOG_TF | grep -i "failed MD5 authentication"
    |
    grep chan_iax2 | awk '{NF=NF-8} {print $NF}'`
    fi
```

Special thanks goes to Ioan Indreias and Gonzalo Gonzalez.

If it looks like the code is missing, try to highlight it all and copy/paste it into a text document. Hopefully this will show the entire rule without issue.

Make sure these rules are stored in /usr/local/bfd/rules/ Then rerun bfd with bfd -q and see if
the new rules are in place by checking for the files in /usr/local/bfd/tmp/ .

46.2.3 To setup Email Alerts

In **/usr/local/bfd**, we will need to edit conf.bfd

Look for

```
# local user or email address alerts are sent to (separate multiple with comma)
```

```
EMAIL_ADDRESS="your email addy"
```

Save and restart services

```
apf –a and bfd –s
```

Hopefully this keeps your systems locked down.

46.3 HOW TO BLOCK COUNTRY IP IN ELASTIX

Login to console or putty as root and install fail2ban.

```
yum -y install fail2ban
```

Fail2ban is needed to monitor attempts and refers to /etc/hosts.deny

Now the easy part, you have to do the following in **/etc**

Edit **/etc/hosts.allow** and add below for your local network

ALL: 192.168.1.0/255.255.255.0   < this is mine (if you don't add you will lock yourself out of ssh access)

Now we need to edit **/etc/hosts.deny** and add this below

```
#ALL:ALL
#ALL:ALL: .ac
#ALL:ALL: .ad
#ALL:ALL: .ae
#ALL:ALL: .af
#ALL:ALL: .ag
#ALL:ALL: .ai
#ALL:ALL: .al
#ALL:ALL: .am
#ALL:ALL: .an
#ALL:ALL: .ao
```

```
#ALL:ALL: .aq
#ALL:ALL: .ar
#ALL:ALL: .as
#ALL:ALL: .at
#ALL:ALL: .au
#ALL:ALL: .aw
#ALL:ALL: .az
#ALL:ALL: .ax
#ALL:ALL: .ba
#ALL:ALL: .bb
#ALL:ALL: .bd
#ALL:ALL: .be
#ALL:ALL: .bf
#ALL:ALL: .bg
#ALL:ALL: .bh
#ALL:ALL: .bi
#ALL:ALL: .bj
#ALL:ALL: .bm
#ALL:ALL: .bn
#ALL:ALL: .bo
#ALL:ALL: .br
#ALL:ALL: .bs
#ALL:ALL: .bt
#ALL:ALL: .bv
#ALL:ALL: .bw
#ALL:ALL: .by
#ALL:ALL: .bz
#ALL:ALL: .ca
#ALL:ALL: .cc
#ALL:ALL: .cd
#ALL:ALL: .cf
#ALL:ALL: .cg
#ALL:ALL: .ch
#ALL:ALL: .ci
#ALL:ALL: .ck
#ALL:ALL: .cl
#ALL:ALL: .cm
#ALL:ALL: .cn
#ALL:ALL: .co
#ALL:ALL: .cr
#ALL:ALL: .cs
#ALL:ALL: .cu
#ALL:ALL: .cv
#ALL:ALL: .cx
#ALL:ALL: .cy
#ALL:ALL: .cz
#ALL:ALL: .de
#ALL:ALL: .dj
#ALL:ALL: .dk
#ALL:ALL: .dm
#ALL:ALL: .do
#ALL:ALL: .dz
#ALL:ALL: .ec
#ALL:ALL: .ee
#ALL:ALL: .eg
#ALL:ALL: .eh
#ALL:ALL: .er
#ALL:ALL: .es
#ALL:ALL: .et
#ALL:ALL: .eu
#ALL:ALL: .fi
#ALL:ALL: .fj
#ALL:ALL: .fk
#ALL:ALL: .fm
#ALL:ALL: .fo
#ALL:ALL: .fr
```

```
#ALL:ALL: .ga
#ALL:ALL: .gb
#ALL:ALL: .gd
#ALL:ALL: .ge
#ALL:ALL: .gf
#ALL:ALL: .gg
#ALL:ALL: .gh
#ALL:ALL: .gi
#ALL:ALL: .gl
#ALL:ALL: .gm
#ALL:ALL: .gn
#ALL:ALL: .gp
#ALL:ALL: .gq
#ALL:ALL: .gr
#ALL:ALL: .gs
#ALL:ALL: .gt
#ALL:ALL: .gu
#ALL:ALL: .gw
#ALL:ALL: .gy
#ALL:ALL: .hk
#ALL:ALL: .hm
#ALL:ALL: .hn
#ALL:ALL: .hr
#ALL:ALL: .ht
#ALL:ALL: .hu
#ALL:ALL: .id
#ALL:ALL: .ie
#ALL:ALL: .il
#ALL:ALL: .im
#ALL:ALL: .in
#ALL:ALL: .io
#ALL:ALL: .iq
#ALL:ALL: .ir
#ALL:ALL: .is
#ALL:ALL: .it
#ALL:ALL: .je
#ALL:ALL: .jm
#ALL:ALL: .jo
#ALL:ALL: .jp
#ALL:ALL: .ke
#ALL:ALL: .kg
#ALL:ALL: .kh
#ALL:ALL: .ki
#ALL:ALL: .km
#ALL:ALL: .kn
#ALL:ALL: .kp
#ALL:ALL: .kr
#ALL:ALL: .kw
#ALL:ALL: .ky
#ALL:ALL: .kz
#ALL:ALL: .la
#ALL:ALL: .lb
#ALL:ALL: .lc
#ALL:ALL: .li
#ALL:ALL: .lk
#ALL:ALL: .lr
#ALL:ALL: .ls
#ALL:ALL: .lt
#ALL:ALL: .lu
#ALL:ALL: .lv
#ALL:ALL: .ly
#ALL:ALL: .ma
#ALL:ALL: .mc
#ALL:ALL: .md
#ALL:ALL: .mg
#ALL:ALL: .mh
```

```
#ALL:ALL: .mk
#ALL:ALL: .ml
#ALL:ALL: .mm
#ALL:ALL: .mn
#ALL:ALL: .mo
#ALL:ALL: .mp
#ALL:ALL: .mq
#ALL:ALL: .mr
#ALL:ALL: .ms
#ALL:ALL: .mt
#ALL:ALL: .mu
#ALL:ALL: .mv
#ALL:ALL: .mw
#ALL:ALL: .mx
#ALL:ALL: .my
#ALL:ALL: .mz
#ALL:ALL: .na
#ALL:ALL: .nc
#ALL:ALL: .ne
#ALL:ALL: .nf
#ALL:ALL: .ng
#ALL:ALL: .ni
#ALL:ALL: .nl
#ALL:ALL: .no
#ALL:ALL: .np
#ALL:ALL: .nr
#ALL:ALL: .nu
#ALL:ALL: .nz
#ALL:ALL: .om
#ALL:ALL: .pa
#ALL:ALL: .pe
#ALL:ALL: .pf
#ALL:ALL: .pg
#ALL:ALL: .ph
#ALL:ALL: .pk
#ALL:ALL: .pl
#ALL:ALL: .pm
#ALL:ALL: .pn
#ALL:ALL: .pr
#ALL:ALL: .ps
#ALL:ALL: .pt
#ALL:ALL: .pw
#ALL:ALL: .py
#ALL:ALL: .qa
#ALL:ALL: .re
#ALL:ALL: .ro
#ALL:ALL: .ru
#ALL:ALL: .rw
#ALL:ALL: .sa
#ALL:ALL: .sb
#ALL:ALL: .sc
#ALL:ALL: .sd
#ALL:ALL: .se
#ALL:ALL: .sg
#ALL:ALL: .sh
#ALL:ALL: .si
#ALL:ALL: .sj
#ALL:ALL: .sk
#ALL:ALL: .sl
#ALL:ALL: .sm
#ALL:ALL: .sn
#ALL:ALL: .so
#ALL:ALL: .sr
#ALL:ALL: .st
#ALL:ALL: .sv
#ALL:ALL: .sy
```

```
#ALL:ALL: .sz
#ALL:ALL: .tc
#ALL:ALL: .td
#ALL:ALL: .tf
#ALL:ALL: .tg
#ALL:ALL: .th
#ALL:ALL: .tj
#ALL:ALL: .tk
#ALL:ALL: .tl
#ALL:ALL: .tm
#ALL:ALL: .tn
#ALL:ALL: .to
#ALL:ALL: .tp
#ALL:ALL: .tr
#ALL:ALL: .tt
#ALL:ALL: .tv
#ALL:ALL: .tw
#ALL:ALL: .tz
#ALL:ALL: .ua
#ALL:ALL: .ug
#ALL:ALL: .uk
#ALL:ALL: .um
#.us  .  United States
#ALL:ALL: .uy
#ALL:ALL: .uz
#ALL:ALL: .va
#ALL:ALL: .vc
#ALL:ALL: .ve
#ALL:ALL: .vg
#ALL:ALL: .vi
#ALL:ALL: .vn
#ALL:ALL: .vu
#ALL:ALL: .wf
#ALL:ALL: .ws
#ALL:ALL: .ye
#ALL:ALL: .yt
#ALL:ALL: .yu
#ALL:ALL: .za
#ALL:ALL: .zm
#ALL:ALL: .zw
```

Now to ban a country just uncomment the country you wish to ban and start fail2ban.

```
service fail2ban restart
```

We are done.

47 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.

My VOIP Network
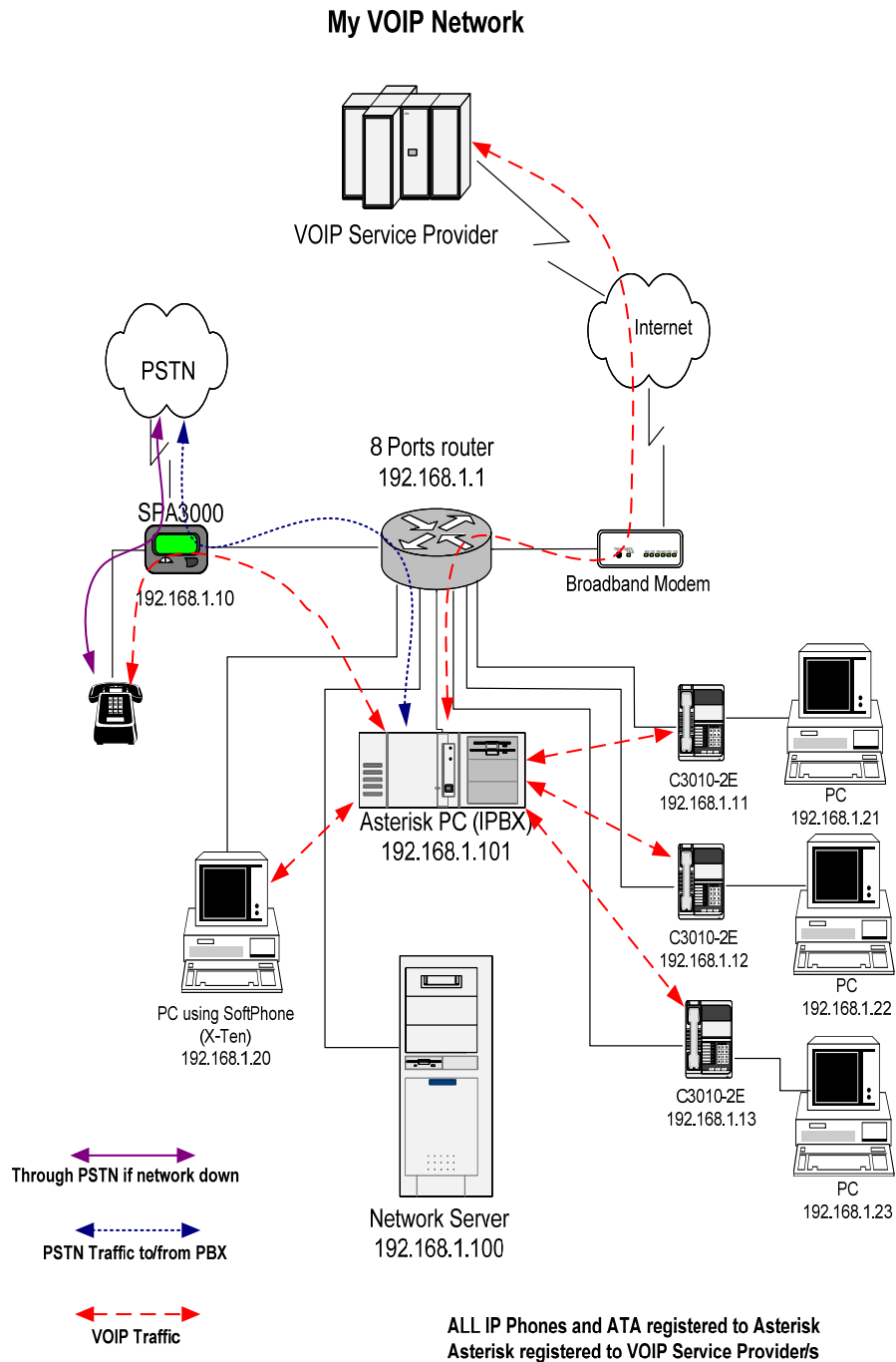
APPENDIX A

A.1 ASTERISK FEATURE CODES

Asterisk feature codes are defined in the feature codes selection of Elastix PBX Configuration screen, illustrated below:

#	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
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.

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)
`${SIPDOMAIN}`	SIP destination domain of an inbound call (if appropriate)
`${SIPUSERAGENT}`	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 = !425/1000,!0/15000,425/360,0/15000
info = 425/2500,0/500
std = !525/100,!0/100,!525/100,!0/100,!525/100,!0/100,!525/100,!0/100,!525/100
facility = 425
stutter = 413+438/100,0/40
ringmobile = 400+450/400,0/200,400+450/400,0/2000

[br]
ringcadence = 1000,4000
dial = 425
busy = 425/250,0/250
ring = 425/1000,0/4000
congestion = 425/250,0/250,425/750,0/250
callwaiting = 425/50,0/1000
dialrecall = 350+440
record = 425/250,0/250
info = 950/330,1400/330,1800/330
stutter = 350+440

[be]
ringcadence = 1000,3000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/3000
congestion = 425/167,0/167
callwaiting = 1400/175,0/175,1400/175,0/3500
dialrecall = !350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440
record = 1400/500,0/15000
info = 900/330,1400/330,1800/330,0/1000
stutter = 425/1000,0/250

[ch]
ringcadence = 1000,4000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = 425/200,0/200,425/200,0/4000
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+340/1100,0/1100

[cl]
ringcadence = 1000,3000
```

```
dial = 400
busy = 400/500,0/500
ring = 400/1000,0/3000
congestion = 400/200,0/200
callwaiting = 400/250,0/8750
dialrecall = !400/100,!0/100,!400/100,!0/100,!400/100,!0/100,400
record = 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,!0/100,!400/10
0,!0/100,400

[cn]
ringcadence = 1000,4000
dial = 450
busy = 450/350,0/350
ring = 450/1000,0/4000
congestion = 450/700,0/700
callwaiting = 450/400,0/4000
dialrecall = 450
record = 950/400,0/10000
info = 450/100,0/100,450/100,0/100,450/100,0/100,450/400,0/400
stutter = 450+425

[cz]
ringcadence = 1000,4000
dial = 425/330,0/330,425/660,0/660
busy = 425/330,0/330
ring = 425/1000,0/4000
congestion = 425/165,0/165
callwaiting = 425/330,0/9000
dialrecall =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,425/330,0/330,425/660,0/660
record = 1400/500,0/14000
info = 950/330,0/30,1400/330,0/30,1800/330,0/1000
stutter = 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,!0/5000,!425/200,!0/200,!425/200,!0/5000,!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+400

[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 = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = 950/650,0/325,950/325,0/30,1400/1300,0/2600
dialrecall = 425/650,0/25
record = 1400/500,0/15000
info = 950/650,0/325,950/325,0/30,1400/1300,0/2600
stutter =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/10
0,!0/100,425

[es]
ringcadence = 1500,3000
dial = 425
busy = 425/200,0/200
ring = 425/1500,0/3000
congestion = 425/200,0/200,425/200,0/200,425/200,0/600
callwaiting = 425/175,0/175,425/175,0/3500
dialrecall = !425/200,!0/200,!425/200,!0/200,!425/200,!0/200,425
record = 1400/500,0/15000
info = 950/330,0/1000
dialout = 500

[fi]
ringcadence = 1000,4000
dial = 425
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 = 1400/500,0/15000
info = 950/650,0/325,950/325,0/30,1400/1300,0/2600
stutter = 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,!0/100,!440/10
0,!0/100,440

[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 = 400,200,400,2000
dial = 350+440
busy = 480+620/500,0/500
ring = 440+480/400,0/200,440+480/400,0/2000
congestion = 480+620/250,0/250
callwaiting = 440/100,0/4000
dialrecall = 300/1500,0/500
record = 1400/500,0/15000
info = !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 =
!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 = 350+375+400

[il]
; Source: libtonezone.
description = Israel
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,!414/16
0,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,414

[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 = !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,!1800/330,0/1000
stutter =
!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!35
0+440/100,!0/100,!350+440/100,!0/100,350+440

[it]
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 =
!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

[jp]
; 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 =
!400/100,!0/100,!400/100,!0/100,!400/100,!0/100,!400/100,!0/100,!400/100,!0/100,!400/10
0,!0/100,400

[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,!425/10
0,!0/100,425

[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 =
!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!35
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,!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,
!0/100,400

[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 =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/10
0,!0/100,425

[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 = 425/1000,0/200
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,!0/100,!425/10
0,!0/100,425

[ru]
ringcadence = 800,3200
dial = 425
busy = 425/350,0/350
ring = 425/800,0/3200
congestion = 425/350,0/350
callwaiting = 425/200,0/5000
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,!1800/330,0

[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,!0/2024,!950/332,!0/24,!1400/332,!0/24,!1800/332,0
stutter =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/10
0,!0/100,425

[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]
ringcadence = 2000,4000
dial = 350+440
busy = 480+620/500,0/500
ring = 440+480/2000,0/4000
congestion = 480+620/250,0/250
callwaiting = 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,!1800/330,0
stutter =
!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!35
0+440/100,!0/100,!350+440/100,!0/100,350+440


[us-old]
ringcadence = 2000,4000
dial = 600*120
busy = 500*100/500,0/500
ring = 420*40/2000,0/4000
congestion = 500*100/250,0/250
callwaiting = 440/300,0/10000
dialrecall = !600*120/100,!0/100,!600*120/100,!0/100,!600*120/100,!0/100,600*120
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330,0
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
```

```
[tw]
ringcadence = 1000,4000
dial = 350+440
busy = 480+620/500,0/500
ring = 440+480/1000,0/2000
congestion = 480+620/250,0/250
callwaiting = 350+440/250,0/250,350+440/250,0/3250
dialrecall = 300/1500,0/500
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330,0
stutter =
!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!35
0+440/100,!0/100,!350+440/100,!0/100,350+440
```

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

```
disallow=all
allow=alaw&ulaw&gsm
authuser=<user number>
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

```
disallow=all
allow=ulaw&alaw
auth=md5
canreinvite=yes
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 Details
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`

```
disallow=all
allow=g729&gsm&ulaw&alaw
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 Faktortel on ELASTIX.
http://www.faktortel.com.au/support-Elastix-settings.shtml

B.1.4 GoTalk

Peer Details
Trunk Name: `GoTalk`

```
disallow=all
allow=alaw
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_general_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: `iinettrunk`

User Details
User Context: *<phone no)*

```
disallow=all
allow=alaw&ulaw
canreinvite=no
username=<phone no>
fromdomain=iinetphone.iinet.net.au
fromuser=<phone no>
insecure=very
dtmfmode=auto
nat=yes
host=sip.nsw.iinet.net.au
registersip=yes
pedantic=no
qualify=yes
secret=<password>
type=friend
```

```
context=from-trunk
fromuser=<phone no>
host= sip.nsw.iinet.net.au
insecure=very
nat=no
secret=<password>
type=friend
username=<phone no>
canreinvite=no
qualify=yes
```

The following may also need to be set in sip_general_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 iPrimus

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 *http://forums.whirlpool.net.au/forum-replies.cfm?t=604015*

First of all, using a browser, we need to enter the following address:

*https://spconfig.voip.iprimus.net.au/SoftphoneConfigurationServer/Default.aspx?Action=LOGIN&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:other_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_fec:enabled="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:digit_map="#1\a\a.T;match=1;prestrip=2;"
proxies:proxy0:domain="asnjla2.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:proxy="sbc01a.syd.voip.iprimus.net.au:15060"
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********

```
disallow=all
allow=alaw&ulaw
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_XXXXXXXX@asnjla2.bw.iprimus.net:YYYYYYYY:AU_XXXXXXXX@iprimus/02********
```

B.1.7 MyNetFone

Peer Details
Trunk Name: MyFonesip

User Details
User Context: 091XXXXX

```
disallow=all
allow=g729&alaw&ulaw
authname=091xxxxx
canreinvite=no
dtmfmode=rfc2833
fromuser=091xxxxx
host=sip01.mynetfone.com.au
insecure=very
nat=yes
pedantic=no
qualify=yes
username=091xxxxx
secret=<password>
type=friend
```

```
canreinvite=no
context=from-trunk
fromuser=091xxxxx
insecure=very
qualify=no
secret=<password>
type=friend
username=091xxxxx
```

Register String:
```
091xxxxx@sip01.mynetfone.com.au:<password>:091xxxxx@sip01.mynetfone.com.au/091xxxxx
```

For further information on MyNetFone, you may refer to the following link:
http://www.myfone.com.au/downloads/sip.conf and *http://www.myfone.com.au/downloads/extensions.conf*

B.1.8 Nehos

Peer Details
Trunk Name: Nehos

```
disallow=all
allow=g729
host=sip.ifone.com.au
qualify=yes
username=661xxxx
secret=<password>
type=peer
```

User Details
User Context: 661XXXX

```
context=from-trunk
host=sip.ifone.com.au
username=661xxxx
secret=<password>
type=user
```

Register String:
```
661xxxx:<password>@sip.ifone.com.au
```

B.1.9 Nodephone

Peer Details
Trunk Name: Nodephone

```
disallow=all
allow=g729
canreinvite=no
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: <usernumber>

```
context=from-trunk
host= sip.internode.on.net
username=<usernumber>
secret=<password>
type=user
```

Register String:
```
<usernumber>:<password>@sip.internode.on.net/<usernumber>
```

B.1.10 SipMe

Peer Details
Trunk Name: sipme

```
authname=1777xxxxxx
dtmfmode=rfc2833
fromuser=1777xxxxxx
host=sip.sipme.com.au
insecure=very
secret=<password>
type=peer
username=1777xxxxxx
```

User Details
User Context: 1777xxxxxx

```
context=from-trunk
fromuser=1777xxxxxx
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`

```
disallow=all
allow=ulaw&g729
canreinvite=no
context=from-trunk
dtmfmode=rfc2833
fromuser=64997xxxxx
host=akl.italk.co.nz
insecure=very
username=64997xxxxx
secret=<your password>
type=friend
```

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`

```
disallow=all
allow=ulaw&g729
dtmfmode=rfc2833
host=voip.0064.co.nz
insecure=very
username=your_Customer_ID
secret=<your password>
type=peer
```

User Details
User Context:; `Your_Customer_ID`

```
context=from-pstn
host=voip.0064.co.nz
insecure=very
username=CUSTOMER_ID
secret=PASSWORD
type=user
```

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

```
disallow=all
allow=ulaw
canreinvite=no
dtmfmode=rfc2833
fromdomain=sip.broadvoice.com
fromuser=<usernumber>
host=sip.broadvoice.com
insecure=very
nat=yes
username=<usernumber>
secret=<password>
type=peer
```

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

```
disallow=all
allow=ulaw&alaw
auth=rsa
canredirect=no
host=iax2.fwdnet.net
inkeys=freeworlddialup
insecure=very
username=65xxxx
secret=<password>
type=peer
```

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

```
disallow=all
allow=g729&ulaw&alaw
canreinvite=no
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>
```

You will not require user details since you will not be receiving incoming calls from this trunk.

You do not need to register either.

```
type=peer
```

If you are using SipBroker, you must ensure that your sip_nat.conf is modified to the following (or add these lines in sip_general_custom.conf).

```
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
username=1747xxxxxxx
secret=<password>
type=peer
```

User Details
User Context: 1747xxxxxxx

```
canreinvite=no
context=from-trunk
fromuser=1747xxxxxxx
insecure=very
qualify=no
username=1747xxxxxxx
secret=<password>
type=user
```

Register String:
```
1747xxxxxxx:<password>@proxy01.sipphone.com/1747xxxxxxx
```

B.3.5 VOIP Buster

Peer Details
Trunk Name: VoipBuster
```
disallow=all
allow=alaw&ulaw
context=from-pstn
dtmfmode=inband
fromdomain=sip1.voipbuster.com
fromuser=<your username>
host=sip1.voipbuster.com
insecure=very
nat=yes (necessary in some
cases)
qualify=yes
username=<your username>
secret=<your password>
srvlookup=yes
type=friend
```

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:

http://www.voip-info.org/tiki-index.php?page=Asterisk+sound+files
http://www.voip-info.org/wiki/view/Asterisk+sound+files+additional
http://www.nathanpralle.com/software/ast_masterlist.html

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
http://www.nuance.com/realspeak/demo/default.asp

The "British Daniel" voice was deep and professional. Check this out
http://www.cbservers.com.au/whirlpool.wav

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.
http://www.acoustica.com/mp3%2Daudio%2Dmixer/

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) *http://sox.sourceforge.net/* 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;
http://lists.digium.com/pipermail/asterisk-users/2002-December/006635.html
http://www.voip-info.org/tiki-index.php?page=Convert+WAV+audio+files+for+use+in+Asterisk
http://users.pandora.be/Asterisk-PBX/IVR.htm
http://www.voip-info.org/tiki-index.php?page=Asterisk+sound+files

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™ Open Source PBX.

The files (male voice) can be downloaded here: *http://www.openvoice.com.au/free.php*

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 ☺ 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 10.0.0.138` [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:
http://www.sipgate.co.uk/faq/index.php?aktion=artikel&type=faq&rubrik=024&id=398&lang=de

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*

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 Whirlpool, Voxilla 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, 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.

F.3 TO RESOLVE THE PROBLEM ON EMBEDDED FREEPBX 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.

F.4 SOLVING MISDIALING USING ANALOG PHONES
(Leonid Fainshtein)

How to solve the following two problems that Elastix users who use analog handset suffer a lot from:

1. User didn't finish the dialing but Asterisk starts to handle the call.
2. Asterisk always add three seconds delay before it starts to handle the dialed number.

Very frequently, PBX users who use analog handsets do not complete dialing the phone number before the PBX starts to make call.

Please be aware of the Asterisk inter-digit timeout handling algorithm for the Zaptel devices:

1. Asterisk waits 16 seconds for the first digit.
2. If there is no ambiguous match, Asterisk waits 8 seconds for the next digit.
3. If there is an ambiguous match, Asterisk waits 3 seconds for the next digit.

Those time-outs are hard-coded and cannot be changed in the Asterisk configuration files.

Following is an example. You have defined two patterns each of which ends with a dot. The first pattern is _9X. — the "x" is followed by a dot. The second pattern is _90044X.

The Asterisk user dials 900-44-208-123-4567.

In this case, the Asterisk will discover the ambiguous match when the user has dialed 900. In other words, the Asterisk will wait for the fourth digit a maximum of three additional seconds. Therefore, we suggest that you define the dial patterns according to the following two principles:

*   The dialing patterns must be as long as possible even when you do not know the exact phone number length. This lowers the probability of a failed dialing attempt.

*   Avoid defining default patterns like _9. Patterns like these will overcome your better defined patterns.

Unfortunately FreePBX by default generates context [bad-number] that is included into the [from-internal] context that is the default for PBX extensions. The [bad-number] context looks like the following:

```
[bad-number]
```

```
include => bad-number-custom
exten => _X.,1,ResetCDR()
exten => _X.,n,NoCDR()
exten => _X.,n,Wait(1)
exten => _X.,n,Playback(silence/1&cannot-complete-as-dialed&check-number-
dial-again,noanswer)
exten => _X.,n,Wait(1)
exten => _X.,n,Congestion(20)
exten => _X.,n,Hangup
exten => _*.,1,ResetCDR()
exten => _*.,n,NoCDR()
exten => _*.,n,Wait(1)
exten => _*.,n,Playback(silence/1&feature-not-avail-
line&silence/1&cannot-complete-as-dialed&check-number-dial-
again,noanswer)
exten => _*.,n,Wait(1)
exten => _*.,n,Congestion(20)
exten => _*.,n,Hangup
```

As you can see, presence of "_X." pattern adds ambiguity to each dialing that a user will perform from their handset. As result of that (a) Asterisk will always make 3 seconds delay after a user has dialed the last digit and (b) maximum inter-digit delay will also 3 seconds only.

It is possible to disable the [bad-number] context generation by adding the following parameter to the */etc/amportal.conf*

```
AMPBADNUMBER=false
```

In order to apply the change you need to force FreePBX to rebuild the configuration. You can make some changes in the configuration and then click on "Apply Configuration Changes   Here".

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:
http://blog.thegoldfish.net/dundi-tutorial-for-asteriskhome/

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 *http://www.elastixconnection.com.au/*

Others that I refer to are:

http://www.voip-info.org/wiki/view/Asterisk%40home+Handbook+Wiki
http://sourceforge.net/forum/?group_id=123387
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.

http://mundy.org/blog/index.php?p=81
http://www.asteriskguru.com/tutorials/
http://www.freewebs.com/voipformula/VoIP-HOWTO.html

G.4 VOICE OVER IP – PER 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 while testing. These will need to be deleted before going live where you will need clean record.

There are two(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

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.2oo7
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:

70-79	-	Reserved for calls on hold
700-799	-	Reserved for calls on hold
7777	-	Reserved extension for incoming calls simulation
911	-	Emergency number
999	-	Emergency number in some countries

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.

2. Through **File Editor** of Elastix, 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 Elastix is active as it may cause some problem while people using it.

3. Through **File Editor** of Elastix 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.

BUG REPORTS

Software is not software if it is released without any bugs. It just goes against natures logic, not to mention our Mr. Murphy ☺

MISSING DIRECTORIES.

The following directories are missing from Elastix 1.0RC. This has been remedied in later versions.

/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 (FOR ZAPTEL USERS)

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

In 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. This may be remedied in future versions.

CALL 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.

If this occurs in your installation, the steps below will eliminate 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. Luckily, **those using IAX trunks only are spared 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 trunk 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 trunk, it will scan for all enabled SIP trunks until the verification times out. If you happen to have 10 SIP trunks, all 10 will be verified first before you can 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 dies 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 fails 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 numbers after a long delay. In many cases the input key will be very sluggish and enough for users to think that nothing is happening, until it actually times out and dies. Worse still 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.

Kludge:

Create 2 boxes. Link both boxes using IAX. You can either use method 1 or method 2 as outlined in the chapter about linking 2 boxes. Make one of the boxs handles all your SIP trunks (Box A). Make the other box (Box B) handles your entire PSTN trunk and make sure you comment out externhost from sip_general_custom.conf or sip_nat.conf in Box B.

Attach all your extensions to to box B

Direct all incoming SIP calls from Box A to Box B.

Direct all outgoing SIP calls from Box B to Box A.

Naturally you must create all the appropriate dial plans and routes to handles your requirements in each of the boxes.

If for any reason internet is down, it will only affect Box A while Box B, where all your extensions and PSTN trunks are will be spared. Of course if your router dies then all will be dead.

Considering a basic box with 1GB RAM, DVD writer and 500GB of Hard disk only costs $350.00, I have resorted to this method and the reward is peace of mind ☺.

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	This works in custom context e.g. • Voicemail(s2000@default) - will not play default message • Voicemail(su2000@default) - will not play default message and instead play your unavailable custom message.
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 to use the "date" command. Example: To set the current time and date to May 12, 2007:10:15.30 seconds pm, type ``date 051**22152**007.**30'** (The time,in **bold**, 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
sip show registry	This command when executed from the Asterisk CLI will display the SIP register status.
sip show peers	This command when executed from the Asterisk CLI will display the SIP peers status.
iax2 show registry	This command when executed from the Asterisk CLI will display the IAX2 register status.
iax2 show peers	This command when executed from the Asterisk CLI will display the IAX2 peers status.

SOME LINUX COMMANDS IN COMMON USE

chown <newowner> <filename>	To change file ownership to the new owner
touch <filename>	To create a blank file using the <filename>

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.
4	**Bob Fryer - Voiceintegrity**	*Bob has been helping me putting this document together and providing me with valuable material and content. Thanks heap Bob.*
5	Colin Swan	*Major assistance, advice, and basically a place to go when things go wrong.*
6	Dave Edwards	*Advance Firewall and Brute Force Elastix Setup*
7	J Todd	*Seven Steps to Better Security with Asterisk*
8	Jack Zimmermann	*Helping me with e164*
9	lusyn.com	*X100P Patch for UK Caller ID http://www.lusyn.com*
10	**Mark Brooker - MBIT**	*Another of my source for help when things go wrong. Mark is a very competent VoIP consultant. No job is too small or too big for Mark.*
11	Openvoice	*Source for Australian Voice*
12	Peter Quodling	*Testbed and resident cynic*
13	Rehan	*Help with Raid configuration.*
14	Rob Thomas	*freePBX Developer and Super Hero – without which there will be no Elastix*
15	Sean Mahon	*Solution for Billion 7402 BL*
16	Shaun (*Ewing*)	*Advise, Tips and tricks on codecs and dial plan etc*
17	Sofoklis Sflomos	*Changes and assistance with Raid configuration*
18	Stefan Keller-Tuberg	*Helping me out with Cron Jobs and Scripts*
19	*SteveM*	*Advise, Tips and tricks on Webmin*
20	*Sunshine Networks*	*Procedure in inplimenting Fail2Ban*
21	*Thunderbird1/Chippy*	*Giving me clues to some obscure processes.*
22	Tim Yardley	*How to secure Trixbox CE*
23	Tzafrir Cohen of XORCOM	*For correcting my various mistakes. I am still wading through all the errors. Thanks Tzafrir*
24	voip-info.org	*A site I often refer to when I require further clarification.*
25	Vladimir de la Cruz	*Assistance with Custom Contexts*
26	Voxilla.com	*http://voxilla.com*
27	Ward Mundy	*My hero http://mundy.org my place of salvation☺.*

I know there are numerous other people I have referenced that I have not mentioned above but their assistance and knowledge are and will always be highly appreciated.

CHANGES MADE SINCE LAST UPDATE

Replace old chapter 46 and added new Chapter 46 – Locking down system