

TRIXBOX without Tears

The ICT serial following
Asterisk@Home withoutTears



trixbox Without Tears

<http://creativecommons.org/licenses/by-nc-sa/2.1/au/>

The Other Dumb-Me Guide (Australianised for AU&NZ users)

Versions 1.x with freePBX

(Version 1.0.2a)

by

Ben Sharif

(The Author of Asterisk@Home without Tears)

[About the Author](#)

Last updated on June 18, 2006

Please send comment or suggestions to me at the address below

<mailto:bensharif@gmail.com>

Please do not send questions directly to me. Instead post all questions in the Whirlpool Forum below so others may also benefit from it.

<http://forums.whirlpool.net.au/forum-threads.cfm?f=107>

First Published on 1st June 2006

This document is released under the [Creative Commons By-Attrib Non-Commercial Share-Alike 2.1](#) licence.

This work is dedicated to my wife, who has been very patient with me, Andrew Gillis and Rob Thomas whose devotions to TRIXBOX and freePBX made it all possible.



Stay tuned for our next episode – TRIXBOX version X

TABLE OF CONTENT

Forward	7
1 Introduction	8
1.1 The Components.....	8
1.1.1 The IP PBX.....	8
1.1.2 Phones	9
1.1.3 SIP Gateway	9
1.1.4 Home Network.....	9
1.1.5 VOIP Service Providers.....	9
2 Is VOIP for You?	10
2.1 What is it going to cost ?	10
2.2 What will the Quality of the phone calls be?	10
3 Installation	11
3.1 Change default Settings.....	14
3.1.1 To get Help.....	15
3.1.2 Change IP Address (set IP address to Static).....	15
3.2 Connect to TRIXBOX from Web Browser	17
3.2.1 Log in to System Administration.....	17
3.3 General Settings	21
3.3.1 Dial Command Options	23
3.4 Extensions.....	25
3.4.1 Extension numbers to avoid	25
3.4.2 Create Extensions.....	25
3.5 Follow Me.....	27
3.6 Ring Groups	28
3.6.1 Now it's a good time to set up your softphone.	29
4 Trunks to the outside world	30
4.1 What is a Dial Pattern?	30
4.2 What is a Trunk?	30
4.3 Let's Create Some Trunks	31
4.3.1 Pennytel	31
4.3.2 Oztell (SIP).....	33
4.3.3 Oztell (IAX).....	34
4.3.4 Astratel	35
5 Calls Handling	38
5.1 Inbound Routes (Incoming Calls).....	38
5.2 Time Conditionds	40
5.3 Outbound Routes	41
5.3.1 What is an Outbound Route?.....	41
5.3.2 How does it work?	41
5.3.3 International.....	42
5.3.4 Domestic	43
5.3.5 MobileAust.....	43
5.3.6 Oztellonly.....	44
5.3.7 Astratelonly.....	44
5.4 ENUM.....	45
5.4.1 Setting up ENUM Trunk	45
5.4.2 Setting up ENUM Outbound Route	46
5.5 How to restrict extensions activities	47
6 System Recordings	51
6.1 Direct recording.....	51
6.2 Upload Pre-recorded Material	52
7 Digital Receptionist	53
7.1 Setting Up Digital Receptionist.....	53

7.2 Customising Individual Extension	55
8 Get Under The Bonnet	56
8.1 Editing The .conf Files.....	57
8.1.1 sip.conf	57
8.1.2 iax.conf	57
8.1.3 Indications.conf	58
8.1.4 enum.conf.....	58
8.1.5 extensions_custom.conf.....	58
9 Check your System.....	59
9.1 System Process Status	59
9.2 Asterisk Info	60
10 Interfacing Asterisk to PSTN.....	62
10.1 Digium Wildcard X100P FXO PCI Card	62
10.1.1 zapata.conf.....	62
10.1.2 Zapata-auto.conf	63
10.1.3 zaptel.conf.....	63
10.1.4 modprobe.conf	63
10.2 Digium TDM400P FXO/FXS Card.....	63
10.2.1 zapata-auto.conf.....	64
10.2.2 modprobe.conf	64
10.3 Rebuild Zaptel Driver	65
10.3.1 IF ALL ELSE FAIL.....	67
10.3.2 How to overcome the Zaptel 1.2.5 dependency.....	67
10.4 Sipura SPA3000 as a PSTN Interface	68
10.4.1 Log in to SPA3000	68
10.4.2 Change the settings	68
10.4.3 Add SIP Trunk.....	70
10.4.4 SPA3000 as an outbound PSTN Trunk.....	70
11 PSTN to VOIP Gateway.....	71
12 DISA.....	72
12.1 DISA - Hand Coding Method.....	72
12.2 DISA – Using freePBX	73
13 Setting Up a BOL SIPPhone.....	74
Profile Tab.....	74
Audio & Video Tab	74
Network Tab.....	75
STUN Tab	75
Call Forward.....	75
14 Operator Panel	76
14.1 Flash Operator Panel (FOP)	76
14.1.1 Setting the Admin Password	77
14.1.2 Hang-up a Call	77
14.1.3 Transfer a Call.....	78
14.1.4 Initiate a Call.....	78
14.1.5 Barge in or Create a Conference	78
14.2 HUDLite.....	78
14.2.1 Setting up HUD Manager	78
14.2.2 Setting Up HUDLite	80
14.2.3 What if it does not work?	82
15 Call parking and transfer.....	83
15.1 Call Transfer - Managed	83
15.1.1 How is it done?.....	84
15.2 Call Transfer – Blind.....	84
15.3 Put a Call On Hold	84
15.4 Call Pickup	84

15.5 Filter Your Incoming Calls - Only Accept Known Calls.....	85
16 MeetMe – Teleconference.....	87
meetme.conf.....	87
meetme_additional.conf	87
16.1 Create Conference.....	87
16.1.1 /var/spool/asterisk/meetme	89
16.2 Invalid Conference number	89
17 Voicemail	90
17.1 Enabling Email Notification	90
17.1.1 Installing sendmail.....	90
17.1.2 /etc/hosts.....	91
17.2 How to Retrieve Voicemail from External Phone	93
17.2.1 Method 1	93
17.2.2 Method 2	93
17.3 How to Change Voice Mail Greetings	94
18 Fax to E-Mail.....	95
18.1 Install-pdf.....	95
18.2 Installing Fax for TRIXBOX	95
19 Weather Forecast.....	98
19.1 Configure Weather Report – On Demand.....	98
19.1.1 Create Extension codes.	99
19.2 Configure Weather Report – Background Method.	100
19.2.1 Create Extension codes.	100
19.3 Now put it in your Digital Receptionist.....	101
20 Remote Management.....	102
20.1 httpd.conf	102
21 How to set up a Remote Extension	103
21.1 Create a new extension	103
21.2 IAX.Conf.....	104
21.3 Sip_Nat.Conf.....	105
21.4 The correct softphone for IAX	105
21.4.1 First the general options.....	105
21.4.2 Accounts options.....	106
22 Tools.....	107
22.1 Webmin	107
22.1.1 To update WebMin	108
22.2 Putty	108
22.3 WinSCP.....	108
23 STUN Servers	109
24 Dialing through MS Outlook®.....	109
24.1 Download AstTapi.....	109
24.2 Install AstTapi.....	110
24.3 Configure Outlook Address Book.....	110
25 Speed Dial.....	112
25.1 The Dumb-Me method	112
25.2 TRIXBOX’s method.....	113
26 How to use Window Messenger® 5.x.....	114
27 How to interconnect 2 boxes	116
27.1 Method 1 - With the Peer Asterisk box as Extensions	116
27.2 Method 2 - In a Peer/User Arrangement	117
Thinking of more than 2 boxes?	118
28 Customised Voice.....	119
28.1 Customising Voice Prompts	119
28.2 Installing Other Languages	120
29 MOH (Music On Hold)	120

29.1 Streaming Music on Hold	121
30 Backup and restore.....	123
30.1 Schedule a Backup	123
30.2 To Restore a Backup	124
31 My TRIXBOX® PBX Network.....	125
32 Publications and References	126
32.1 DUNDi – a guide to a Simple Setup	126
32.2 The future of Telephony by O'Reilly Publishing	126
32.3 Other Asterisk Forums, Tutorials and Wikis	126
32.4 Voice over IP – Per call bandwidth consumption	126
33 USB Phone Support.....	127
34 Bugs Reports.....	128
34.1 Missing Zaptel support	128
34.2 Missing /var/spool/asterisk/meetme	128
34.3 Editing email notification message	128
34.4 WakeUp Calls	128
35 Trouble Shooting	129
35.1 Asterisk Feature Codes not working	129
35.2 Asterisk Drops Calls after a few seconds.....	129
35.3 Pennytel Asterisk Problem	129
35.4 Unable to receive Incoming Calls.....	129
36 Port Forwarding and QOS – Routers	130
36.1 Port Forwarding.....	130
36.2 QOS – Routers.....	130
37 Updating TRIXBOX.....	132
Appendix A	135
A.1 Asterisk Feature Codes	135
A.2 Asterisk CLI commands.....	135
A.2.1 General commands.....	135
A.2.2 AGI Commands	136
A.2.3 Database Handling	136
A.2.4 IAX Channel Commands	137
A.2.5 SIP Channel commands	137
A.2.6 Server management	137
A.3 Asterisk Special Extensions.....	138
A.4 Asterisk Common Variables	138
A.5 Indications for Various Countries	139
A.6 Codec (Coder Decoder).....	147
Appendix B	148
B.1 – More trunks	148
B.1.1 BBPglobal	148
B.1.2 Engin BYO	148
B.1.3 Faktortel (IAX).....	149
B.1.4 FWD aka Pulver(IAX).....	149
B.1.5 inet	149
B.1.6 iTalk (New Zealand).....	150
B.1.7 Koala.....	150
B.1.8 MyNetFone	151
B.1.9 Nehos (IAX)	151
B.1.10 Nodephone	151
B.1.11 SipBroker	152
B.1.12 SipMe.....	152
B.1.13 SipPhone	152
B.1.14 TelephoneGlobal.....	153
B.1.15 VOIP Buster	153

B.2 Related Links – (for vsp TRUNK settings)	154
B.2.1 Trunk and Route Settings	154
B.2.2 MyFone settings.....	154
B.2.3 Setting up FaktorTel for TRIXBOX.....	154
Appendix C	155
C.1 Voice for Asterisk Messages	155
C.2 Australian Voice and Language Set for Asterisk	156
C.3 More Australian Voice for Asterisk	156
Appendix D (ATAs and Routers)	157
D.1 Alcatel SpeedTouch 510/530 ADSL Router	157
D.2 Billion 7402 VL.....	157
D.3 Linksys PAP2 and Sipura SPA2000	158
Appendix E (Users' suggestions).....	160
E.1 Suggestions for Dial Plan and configuration	160
E.1.1 Astratel.....	160
E.1.2 Further .conf modifications by Colin.....	161
E.2 Multiple TDM400P Installation	162
E.3 Eliminating Echo Problems in SPA-3000.....	165
E.4 Immediate Answer of PSTN Calls	167
E.5 Rebuild Zaptel Driver (Users' Suggestions).....	168
E.5.1 As experienced by marner – a Whirlpool Forum participant.....	168
E.5.2 As suggested by Rob Thomas (the FreePBX guru).....	168
E.6 Zaptel card configuration for Australia	168
Notes	171
A2Billing	171
Conversation Recordings.....	171
Dial 0 for Outside Line.....	171
Using Zaptel Card Method	172
Using SPA3K Method.....	172
Delete Call Records from CDR	172
Extension Numbers to avoid using.....	173
Giving Priority to Emergency Numbers (E.g. 000)	173
sip_nat.conf.....	174
Sipura SPA3000 – Not hanging up after PSTN Calls	174
Trunks and Outbound Routes – a circus of confusion	176
Outgoing Route Dial Pattern	176
Trunk Outgoing Dial Rules	176
Adding the missing Maintenance Modules to freePBX 2.1.0	177
Bibliography	178
Changes Made since last Update	179

FORWARD

This guide is not meant to be a detailed Manual for TRIXBOX®; instead it is just a **quick and dirty** guide and a resource to assist you get started with minimum pain. Use this document only as a guide, not as a Gospel. It does not cover all minute details, variations and possibilities of TRIXBOX neither will it provide you with a complete solution.

If a complete solution is required for your specific requirement, I suggest you obtain the service of a VOIP consultant who will design your solution for a fee.

This document was compiled with input from dozens of participants in the Whirlpool - in particular, Peter Q, Chippy, Rob Thomas and Leslie (Un-Nefer), and other forums to whom I owe a great deal while learning how to configure TRIXBOX. These people have helped me whenever I get into some difficulties – and there are many difficulties and counting☺.

If this document is of some use to you, I take no credit for it. I invite others to participate in improving this document. Please send me your feedbacks; so together we can make this document a valuable resource for our peers, experienced and new alike.

- Ben Sharif June 1, 2006.

Caveat: (*Don't you just hate having to say this..*)

In its Basic form, all that I have outlined worked for me but it may contain many errors, omissions and typos that can do with a lot of extra corrections. While I have taken due care in compiling this document, however, it may not necessarily be 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 method or advice found in this document.

Footnote:

I DO NOT provide TRIXBOX support and NEITHER do DIGIUM. If assistance is required, please post in the forum. Participants in the forum 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.

<http://forums.whirlpool.net.au/forum-threads.cfm?f=107>

I have received numerous emails asking for assistance. The volume of email has become **unmanageable**. I am not an expert on Asterisk. While I will try to assist within my limited knowledge, I hope you will understand if I don't answer each and every email. I ask those requiring assistance to post their questions in the Whirlpool Forum (**forum participants may not answer any question if the solution is already in this guide**) where there are many knowledgeable Asterisk and Linux savvy participants who will be able to assist with TRIXBOX.

1 INTRODUCTION

Asterisk © is a Linux based IPBX application developed by Mark Spencer of Digium™, the company behind Asterisk. TRIXBOX© evolved from the core Asterisk. 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, TRIXBOX and Asterisk will be referred to frequently and they are interchangeable as TRIXBOX is in essence a superset of the raw Asterisk.

The major components that make up TRIXBOX are

- Asterisk, the core PBX
- Sugar, a CRM system
- [A2Billing](#) – Calling Card platform
- Flash Operator Panel, a screen-based operator's console
- Web Meet Me Control, a meet me conferencing control application
- freePBX a web-based provisioning tool for Asterisk.
- A report system – the part of freePBX which provides CDR reporting tools
- A Maintenance system, also part of TRIXBOX, which provides low level interfaces to some components and real time system information
- CentOS, a version of Linux related to a very well known Enterprise Linux (but without the branding and support).

1.1 THE COMPONENTS

Four main components need to be set up:

- The Asterisk powered IP PBX
- The phones (or softphones) and
- The VOIP gateway service that lets you call other VOIP users and people on the PSTN.
- Have a home network and broadband access with a router and hubs/switches if needed.

I will attempt to describe the set up of each of these. It does not matter if you are behind a firewall. 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.

1.1.1 The IP PBX

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

- 250Mhz Pentium II PC or better.
- 128MB RAM – the more the better
- 4 GB Hard disk space - minimum.
- 10/100 NIC
- CD-ROM Drive

- 10/100 4 or 8 ports Ethernet hub/switch (not required if your router has spare ports)

Note* *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>

Install TRIXBOX on this “old” computer. It will take it over – it starts by formatting the hard disk, so make sure there is nothing on the machine that you want to keep. Detailed instructions are given in the installation section.

1.1.2 Phones

You can buy SIP phones from Grandstream, SNOM, Cisco etc. Or you can buy an ATA (Analogue Telephone Adapter) produced by Linksys, Netcomm or Sipura, which enables you to use an analog phone. The Netcomm and Linksys ATAs can be bought from Harris Technology and a number of VSPs.

Register with a VSP such as Faktortel, Oztell and several other VOIP Service Providers (VSPs) or you may buy Engin locked ATAs from Harvey Norman and Dick Smith Electronic. 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.

1.1.3 SIP Gateway

It is 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.

1.1.4 Home Network

The fact that you are reading this and contemplating on installing an Asterisk PBX, I assume that you have a home network, a broadband service, set up behind a Gateway Router of some sort (otherwise known as a NAT firewall).

You will need to pick a static IP address for your IP PBX that is on your home network e.g. 192.168.0.101. To keep it simple, please ensure that the Asterisk box is in the same network segment of your existing LAN.

1.1.5 VOIP Service Providers

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>

2 IS VOIP FOR YOU?

Whether VOIP is for you or not rely on a number of or combination of factors. Some economic and quality considerations should be examined.

2.1 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 Dick Smith) for the softphone. If you do not have a spare PC with the above specification, then you may be able to buy one from your local swap meets for under \$200.00, which may include a monitor. Ensure that the PC has an Ethernet NIC for connecting to your home network.

Your only other initial cost will be the \$20.00 or so activation fee to Oztel (or other VSP of your choice), if you want the ability to make PSTN calls. If you want to restrict all your calls to VOIP only, it may not cost you anything at all.

Some VSPs like Pennytel, Astratel, Spantalk etc will register you for SIP communication for free provided that you do not need to make PSTN calls.

All these "Major Expenses" will be recovered when you receive your monthly Telstra or Optus phone bills ☺

2.2 WHAT WILL THE QUALITY OF THE PHONE CALLS BE?

If you are expecting the quality to be as good as your existing PSTN calls, you will be somewhat disappointed, but if you will be happy with a quality that is not quite but close to your existing PSTN calls and prepared with the whining by the other half, the phone calls are not being as good while you are leaching off the P2P, you might be in luck.

VoIP 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

3 INSTALLATION

(The dumb-me way)

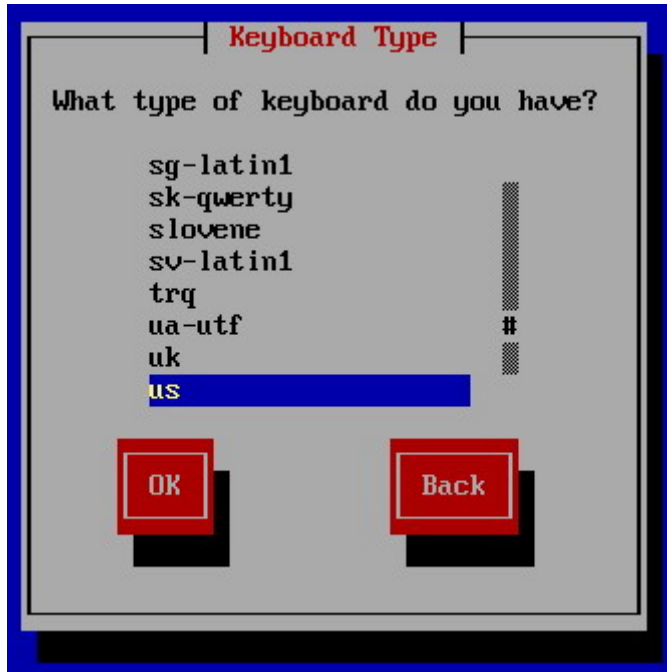
Download the ISO from here

<http://www.trixbox.org/modules/smartsection/item.php?itemid=2/>

- 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 reflect 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 TRIXBOX PC with the CD in the CD Drive and press enter. After a few seconds, the following screen will be displayed. You press F2 to see the various options. However, it is not really necessary. Just press [Enter] to start the installation



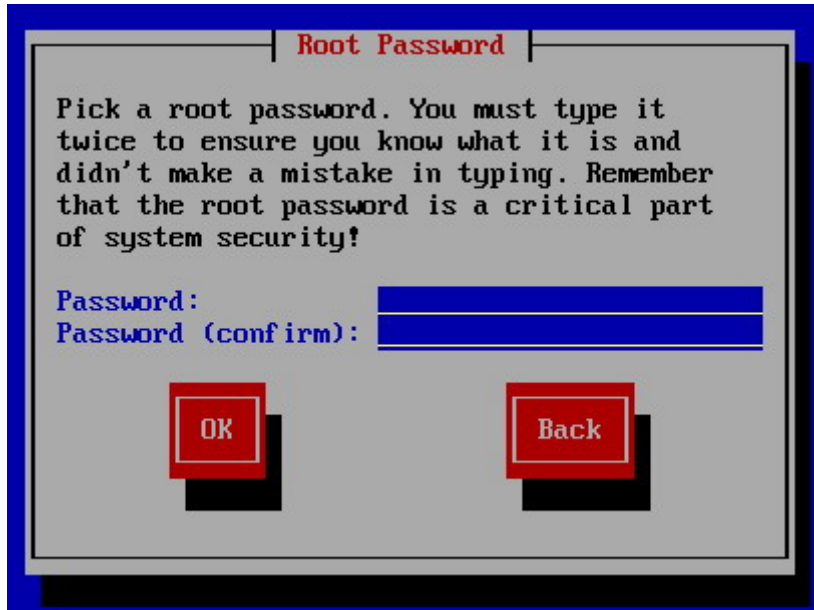
- Press the Enter key to start the installation.
- After initial system detection, you will be asked to select the type of keyboard that you will be using. Use your cursor keys to navigate the keyboard types and use the Tab key to move the focus to the **OK** or **Back** option. Select the appropriate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



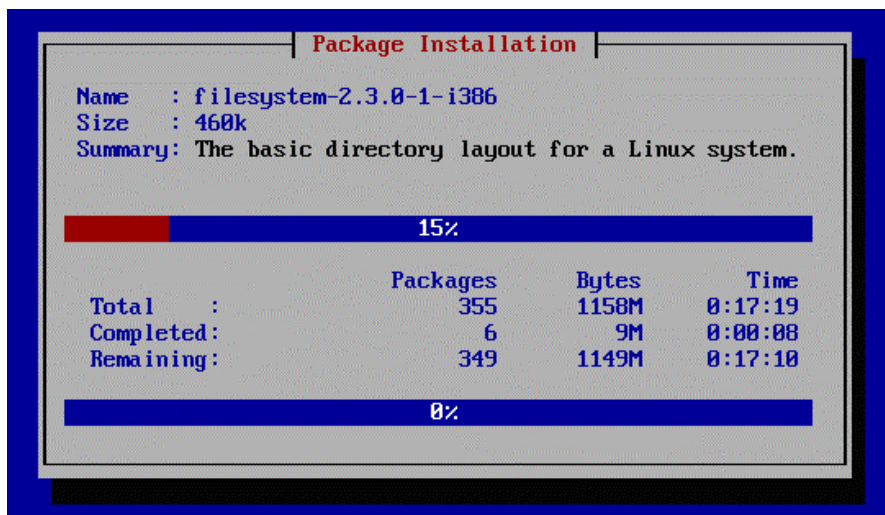
- After system hardware probing is completed, you will be asked to select the Time Zone you are in.



- Use your cursor arrow keys to navigate to the appropriate time zone. I selected **Australia/NSW** after which I use the TAB key to move the focus to **OK** and hit [Enter] to accept.
- Next you will be asked to enter your root password.



- After entering and confirming your password, installation will now commence by first formatting your Hard Disk.
- From this point it will take about 30-45 minutes for the installation to be complete 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 following screen.



- After a moment, it will continue and you will see lots of lines of codes. This process will take a while because it is building Asterisk.
- When Asterisk build is complete, it will reboot itself and once rebooted, TRIXBOX is ready for you to make changes to the system default and configure TRIXBOX the way you want it.

3.1 CHANGE DEFAULT SETTINGS

Once TRIXBOX has been installed, some default system changes need to be made to TRIXBOX.

Log in to your new TRIXBOX box (**user:** root, **password:** password)

```
login as: root
root@192.168.1.29's password:
Last login: Thu Jun  1 22:49:08 2006 from 192.168.1.3

Welcome to trixbox
-----

For access to the trixbox web GUI use this URL
http://192.168.1.29

For help on trixbox commands you can use from this
command shell type help-trixbox.

[root@asterisk1 ~]# █
```

Before you proceed any further, update TRIXBOX – go to section UPGRADING TRIXBOX - it will make you a happier digger 😊

3.1.1 To get Help

At the command line, type

```
help-trixbox
```

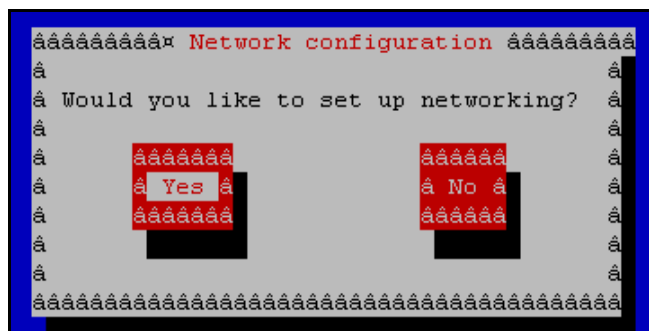
A list of help will be displayed – see illustration below;

```
[root@asterisk1 ~]# help-trixbox
trixbox - HELP
-----
Commands          Descriptions
-----
config            set the local time zone and keyboard type
netconfig        configure ethernet interface
genzaptelconf     autoconfig Zaptel cards
trixbox-update.sh upgrade trixbox to the latest version
install-ZAPHFC    install HFC ISDN support using BRISTUFF
install-AVMB1ISDN install support for AVB B1 ISDN card
install-EiconDiva install support for Eicon Diva ISDN card
install-pdf       installs support for emailing PDFs of faxes
install-netmrg    installs the netmrg package
passwd-maint      set master password for web GUI
passwd-amp        set password for amp only
passwd-meetme     set password for Web MeetMe only
passwd           set root password for console login
passwd admin     set admin password for checking system mail
setup-cisco      create a SIPDefault.cnf in /tftpboot
setup-dhcp       set up a dhcp server
setup-samba      set up a Samba server (Microsoft file sharing)
setup-mail       configure sendmail
rebuild_zaptel   rebuild zaptel driver after kernel update
asterisk -r      Asterisk CLI
yum -y update    Get latest patches for CentOS
```

3.1.2 Change IP Address (set IP address to Static)

Change Asterisk IP address from DHCP to Static. At the command prompt enter:

```
Netconfig
```

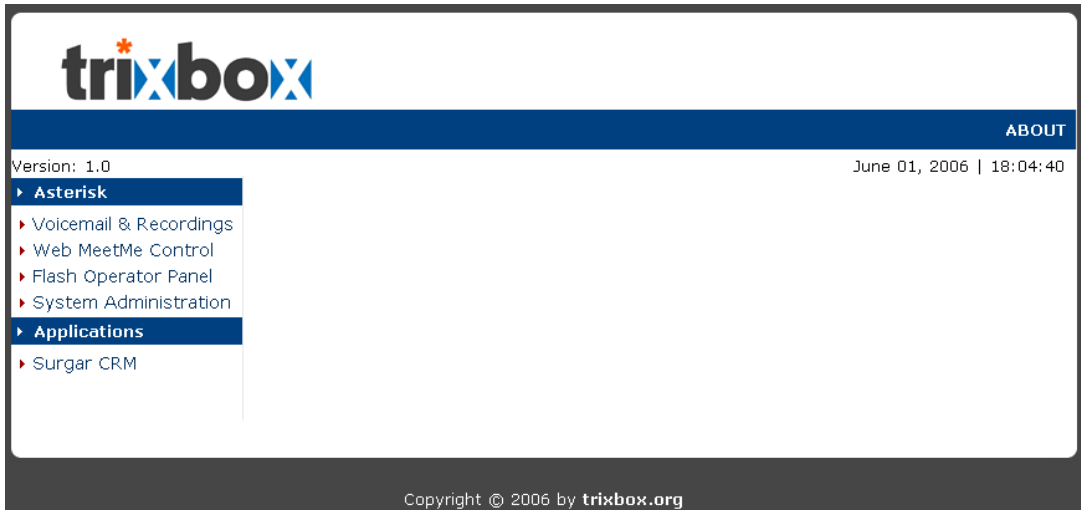


Select **[Yes]** to set up networking and hit enter.

You will then see the following screen.

3.2 CONNECT TO TRIXBOX FROM WEB BROWSER

Now you can connect to ***http://ipaddress/*** (e.g. `http:192.168.1.100`) to configure TRIXBOX. You will be presented with the Main Menu screen as illustrated below.

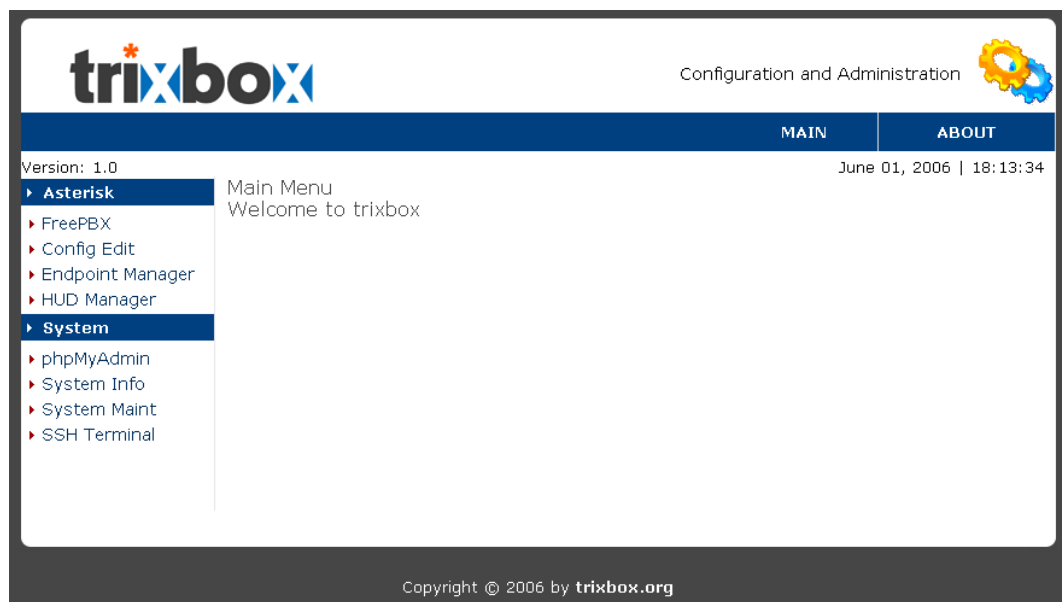


Main Welcome Screen

3.2.1 Log in to System Administration

***To log in to System Administration, use **user: maint**, **password:** password unless you have changed the password during initial set up in 3.1.2*

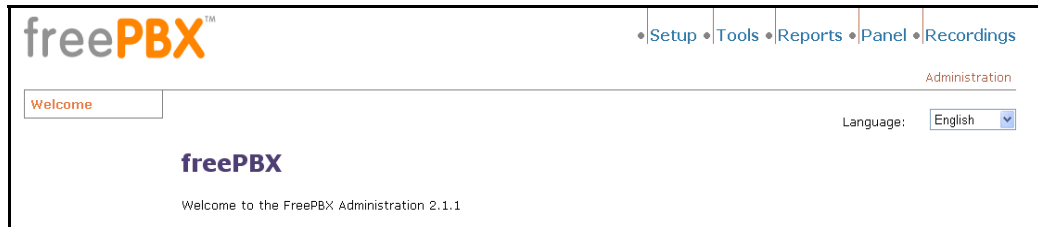
Once you logged in, you will be presented with the following screen,



Main Configuration Screen

At this stage select the **freePBX** option and you will be presented with the screen.

You are now in freePBX and this is where most of your configurations will start.



Select Tools

The red bar will appear.. click on it to begin



Click Tools again

The same screen will appear without the red bar on top.



You will need to activate all the Administration/Management modules that you require eg: Extensions, Queue, Digital Receptionists, Trunks etc.

To do this, select Module Admin option on the left right under the FreePBX decal and you will be presented with the Module Administration screen. You will be required to enable and installs all the modules that you are likely to need. It does not matter if you enabled modules that you do not require as all it will do is making FreePBX refresh a little slower.

Click on Module Admin option on the left.

You will notice all the modules that are available as illustrated below.

freePBX™

• Setup • Tools • Reports • Panel • Recordings

Tools

Module Admin

Module Administration

[Connect to Online Module Repository](#)

Not Installed Local Modules

Module	Version	Type	Category
<input type="checkbox"/> Asterisk API (manager)	1.0.2	tool	Basic
<input type="checkbox"/> Core (core)	1.1	setup	Basic
<input type="checkbox"/> Feature Code Admin (featurecodeadmin)	1.0	setup	Basic
<input type="checkbox"/> Follow Me (findmefollow)	1.2.5	setup	Basic
<input type="checkbox"/> Ring Groups (ringgroups)	1.2.2	setup	Basic
<input type="checkbox"/> Time Conditions (timeconditions)	2.1	setup	Basic
<input type="checkbox"/> Call Forward (callforward)	1.0.2	setup	Call Management
<input type="checkbox"/> Call Waiting (callwaiting)	1.0.1	setup	Call Management
<input type="checkbox"/> Do-Not-Disturb (DND) (donotdisturb)	1.0.1	setup	Call Management
<input type="checkbox"/> VoicePulse (voicepulse)	0.1	setup	ITSP
<input type="checkbox"/> Conferences (conferences)	1.0.1	setup	Module
<input type="checkbox"/> IVR (ivr)	2.2.6	setup	Module
<input type="checkbox"/> On Hold Music (music)	1.0.5	setup	Module
<input type="checkbox"/> PIN Sets (pinsets)	1.0.5	setup	Module
<input type="checkbox"/> Paging and Intercom (paging)	1.2.1	setup	Module
<input type="checkbox"/> Queues (queues)	1.0.4	setup	Module
<input type="checkbox"/> Recordings (recordings)	2.0.7	setup	Module
<input type="checkbox"/> DISA (disa)	2.0	setup	Remote Access
<input type="checkbox"/> Backup & Restore (backup)	1.0	tool	System Admin

Enable Selected

freePBX

Version 2.1.1 on 192.168.1.29

Note: *You don't have to select all the modules. Only select the modules that you need.*

To make it simple, I selected all modules by ticking all the tick boxes.


Click on the Submit button

However you may, at a later stage, choose to disable some of the modules that you will not need by ticking the box next to the module and from the dropdown box next to the submit button, select "Disable Selected" and hit Submit.

Connect to online module repository to update any of the modules that have been updated since the installation, or include any other extra modules that have been released.

freePBX™ Setup • Tools • Reports • Panel • Recordings Tools

Module Admin **Module Administration**

[Connect to Online Module Repository](#) 

Not Installed Local Modules

Module	Version	Type	Category
<input type="checkbox"/> Asterisk API (manager)	1.0.2	tool	Basic
<input type="checkbox"/> Core (core)	1.1	setup	Basic
<input type="checkbox"/> Feature Code Admin (featurecodeadmin)	1.0	setup	Basic
<input type="checkbox"/> Follow Me (findmefollow)	1.2.5	setup	Basic
<input type="checkbox"/> Ring Groups (ringgroups)	1.2.2	setup	Basic
<input type="checkbox"/> Time Conditions (timeconditions)	2.1	setup	Basic
<input type="checkbox"/> Call Forward (callforward)	1.0.2	setup	Call Management
<input type="checkbox"/> Call Waiting (callwaiting)	1.0.1	setup	Call Management
<input type="checkbox"/> Do-Not-Disturb (DND) (donotdisturb)	1.0.1	setup	Call Management
<input type="checkbox"/> VoicePulse (voicepulse)	0.1	setup	ITSP
<input type="checkbox"/> Conferences (conferences)	1.0.1	setup	Module
<input type="checkbox"/> IVR (ivr)	2.2.6	setup	Module
<input type="checkbox"/> On Hold Music (music)	1.0.5	setup	Module
<input type="checkbox"/> PIN Sets (pinsets)	1.0.5	setup	Module
<input type="checkbox"/> Paging and Intercom (paging)	1.2.1	setup	Module
<input type="checkbox"/> Queues (queues)	1.0.4	setup	Module
<input type="checkbox"/> Recordings (recordings)	2.0.7	setup	Module
<input type="checkbox"/> DISA (disa)	2.0	setup	Remote Access
<input type="checkbox"/> Backup & Restore (backup)	1.0	tool	System Admin

Enable Selected

freePBX™ Version 2.1.1 on 192.168.1.29

Click on the link pointed by the red arrow in the illustration above.

When connected, you will be presented with the following screen:

freePBX™ Setup • Tools • Reports • Panel • Recordings Tools

Asterisk API **Module Administration (online)**

Backup & Restore [Terminate Connection to Online Module Repository](#)

Module Admin

Available Module Updates (online)

Module	Version	Type	Category
<input type="checkbox"/> PIN Sets (pinsets)	1.0.6	setup	Module
<input type="checkbox"/> Recordings (recordings)	2.2	setup	Module

Download and Update selected

Modules Available (online)

Module	Version	Type	Category
<input type="checkbox"/> Gabcast (gabcast)	1.0	setup	Addon
<input type="checkbox"/> Misc Destinations (miscdests)	1.1.1	setup	Basic
<input type="checkbox"/> PHP Info (phpinfo)	1.0	tool	Basic
<input type="checkbox"/> Online Support (irc)	1.0	tool	Help
<input type="checkbox"/> Info Services (infoservices)	1.2.1	setup	Information
<input type="checkbox"/> Asterisk Logfiles (logfiles)	1.0.1	tool	Maintenance
<input type="checkbox"/> Voicemail (voicemail)	1.0	setup	Messaging
<input type="checkbox"/> Callback (callback)	1.1.1	setup	Module
<input type="checkbox"/> Java SSH (javash)	1.0.1	tool	Module
<input type="checkbox"/> Asterisk CLI (asterisk-cli)	0.001	tool	System Admin

Download and Install selected

Enabled Modules

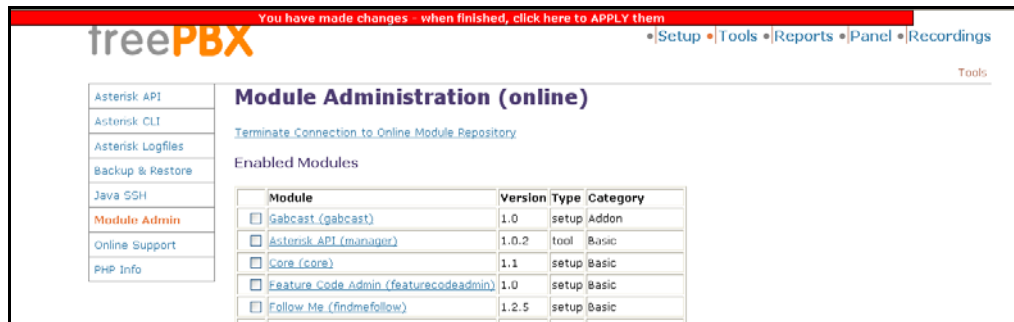
Module	Version	Type	Category
<input type="checkbox"/> Asterisk API (manager)	1.0.2	tool	Basic

Apart from the modules that are already enabled, you will also see more modules that you can install and also modules that have been updated (highlighted in amber) since your installation.

Again to simplify matter, I will select all the modules.

First I will select all the Yellow modules and hit the submit button. When the screen refresh, I will select all the other modules beneath it and hit the submit button.

I now have all modules installed and enabled.



Click on the red bar to commit the updates. When the screen refreshed, you are done with the modules management section.

You will now notice more options have appeared on the left side of the screen.

Once you have enabled all the required modules, you are ready to start. From this point onward, the setup process are similar the difference being the presentation of the management screen and some extra options in FreePBX.

Now you can start configuring TRIXBOX. Notice the selection options on the left. Selecting each option will display configuration screen for that particular function e.g. creating new extensions, creating new trunks etc.

This is where most of the action begins.

3.3 GENERAL SETTINGS

The first thing I do is select General Setting and set it up as illustrated below.

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

Notice that the extra information that will define the way Asterisk behave, are also required in freePBX. Set the fields to the following (these are vital settings):

Voicemail:

Change the Direct Dial to Voicemail message type to something other than default - this will save you a lot of head scratching later on when you start using follow me etc. as your customs recorded message will not be played, instead the default Asterisk message will be played back if you don't change the setting.

I have set mine to **Unavailable**.

Hovering your mouse on the corresponding field description with a yellow/amber underline will display the purpose of the fields.

Asterisk Outbound Dial command option: “r” which generate the ring when you dial out, or “m” if you want music instead. There are other options of course – refer to the chapter on [Dial Command Options](#).

Country Indications: Australia

Allow Anonymous Inbound SIP Calls?: Yes (if this is not set you ‘Yes’, all inbound unidentified SIP calls will not be accepted)

After setting up the General Settings, click on Submit Changes button and the red bar on top of the screen for the change to take effect.

3.3.1 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”. 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:

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 <i>after</i> 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 <i>calling</i> 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: <ul style="list-style-type: none"> • 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])	<p>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</p> <ul style="list-style-type: none"> • 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 behavior 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 calling party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in features.conf.

3.4 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 my case I have 4 extensions to experiment with – 3 soft phones and one ATA.

There are a number of extension numbers you should avoid using unless you are prepared to edit and change some codes.

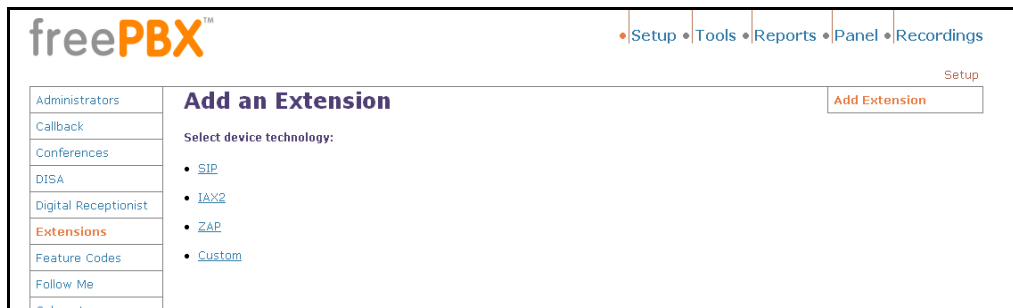
3.4.1 Extension numbers to avoid

It's best to avoid the following extension numbers:

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

3.4.2 Create Extensions

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



The illustration below is where you create the extension.

Submit when done.

freePBX™ Setup • Tools • Reports • Panel • Recordings

Setup

Administrators
 Callback
 Conferences
 DISA
 Digital Receptionist
Extensions
 Feature Codes
 Follow Me
 Gabcast
 General Settings
 Inbound Routes
 Misc Destinations
 On Hold Music
 Outbound Routes
 PIN Sets
 Paging and Intercom
 Queues
 Ring Groups
 System Recordings
 Time Conditions
 Trunks
 VoicePulse

Add SIP Extension

Add Extension

Extension Number:

Display Name:

Extension Options

Direct DID:

DID Alert Info:

Outbound CID:

Emergency CID:

Record Incoming:

Record Outgoing:

Device Options

secret:

dtmfmode:

Voicemail & Directory:

freePBX™ Version 2.1.1 on 192.168.1.29

Add Extension Screen

Click on the red bar on the top of the screen everytime you create a new extension.

You have made changes - when finished, click here to APPLY them

freePBX™ Setup • Tools • Reports • Panel • Recordings

Setup

Administrators
 Callback
 Conferences
 DISA
 Digital Receptionist
Extensions
 Feature Codes
 Follow Me
 Gabcast
 General Settings
 Inbound Routes

Add SIP Extension

Add Extension

Extension Number:

Display Name:

Extension Options

Direct DID:

DID Alert Info:

Outbound CID:

Ben Sharif <2001>
 Norsurya Sharif <2002>

Click on the Add Extension button to add more extensions.

My extensions are 2000, 2001, 2002 and 2003

For simplicity, I allocated passwords to be the same as the extension numbers. If you enabled Voicemail, you may 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. This is covered in more detail in the Chapter a little later in this document.

3.5 FOLLOW ME

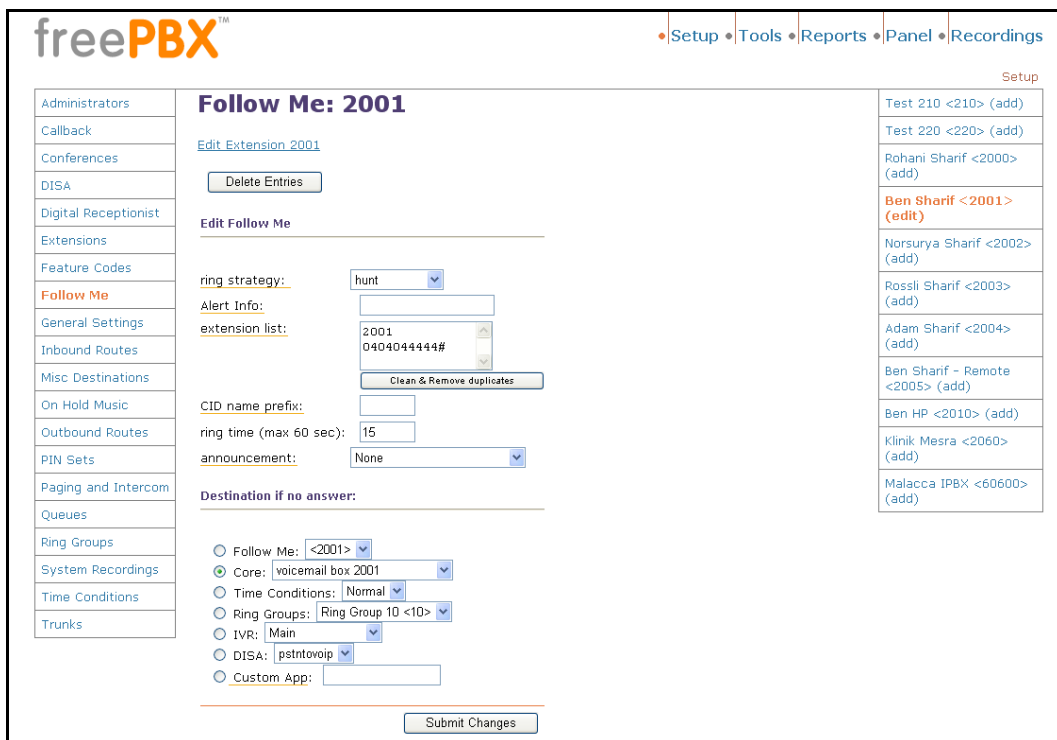
After setting up your extensions, you need to decide if you want Asterisk to call another pre-arranged extension, if the extensions called do not answer. This is where you will define it as per the illustrations below:

To do this, select the Follow me option; **Setup -> 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 my case, I picked my extension – **'Ben Sharif <2001> add'**



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

- **Ring strategy:** hunt (to call the numbers in sequence)
- **Extension List:** 2001 (my extension) followed by my Mobile phone number 04040444444# (notice the '#' after my mobile number – this is required)
- **Ring time:** 40 seconds
- **Destination if no answer:** Core – mailbox 2001

Every time my extension is called, Asterisk will try to connect to extension 2001 and if no answer, it will call my mobile. If still no answer, it will drop to my mailbox.

3.6 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 my mobile phone (see illustration below). *For mobile phone to work, you must have the appropriate route and trunk set up.*

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

When there is an incoming call, 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, I have only defined 1 ring group for all incoming calls from all trunks – at this stage, let's not get too fancy ☺

I created a ring group 10 for this purpose. I called it AllPhones as it rings all the phones in the group. A single digit ring group is not recommended.

The ring group screen is illustrated below:

freePBX™ • Setup • Tools • Reports • Panel • Recordings

Setup

- Administrators
- Callback
- Conferences
- DISA
- Digital Receptionist
- Extensions
- Feature Codes
- Follow Me
- Gabcast
- General Settings
- Inbound Routes
- Misc Destinations
- On Hold Music
- Outbound Routes
- PIN Sets
- Paging and Intercom
- Queues
- Ring Groups
- System Recordings
- Time Conditions
- Trunks
- VoicePulse

Ring Group: 10

[Add Ring Group](#)
AllPhones (10)

Edit Ring Group

group description:

ring strategy:

extension list:

CID name prefix:

ring time (max 60 sec):

announcement:

Alert Info:

Destination if no answer:

Ring Groups:

Core:

Custom App:

freePBX™ Version 2.1.1 on 192.168.1.29

3.6.1 Now it's a good time to set up your softphone.

To do this, go to the chapter – [Setting up Soft Phone](#) and come back to the next chapter after that is done.

If everything has been done as explained above, you should be able to make and receive calls between your 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.

4 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 or make calls to the outside world.

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 TRIKBOX 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 your 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 dialed to Asterisk – just delete your dialling rule from your ATA and replace it with (*x.|xx.)*

4.1 WHAT IS A DIAL PATTERN?

Dial patterns are part of Outbound Routing. They act like a filter for matching numbers dialed with trunks. A further explanation can be found under [Notes](#).

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**.

4.2 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 enable 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 call via trunk (b).

To create a new trunk using **freePBX**, select **Setup** tab and then select the **Trunks** option from the vertical menu on the left.

4.3 LET'S CREATE SOME TRUNKS

For this example, let's create 3 **SIP** trunks; Pennytel, Oztell and Astratel

[Pennytel](#) will be used for all International and mobile calls

[Oztell](#) will be used for all local and mobile calls.

[Astratel](#) will be used for international calls

To create a new Trunk, click on the Add Trunk option (first option on the right of the screen).

First create a Pennytel, Oztell and Astratel SIP Trunks that we will call **pennysip**, **oztellsip** and **astrasip** respectively and define their dialing patterns.

4.3.1 [Pennytel](#)

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

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

We will require the following Outgoing Dialing Rules for this trunk, which instruct the trunk to dial whatever is sent by the appropriate route. This example is for dialing 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.
60NXXXXXXXX	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 dialing 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
60NXXXXXXXX
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 sip**

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 TRIXBOX

```
allow=alaw&ulaw&gsm
canredirect=no
canreinvite=no
disallow=all
host=203.166.6.160
insecure=very
secret= Password
type=peer
username=8880XXXX
```

Incoming Settings

In the **User Context**, enter the Pennytel User ID: **888XXXXX**

In the **User Details** we will have the following:

```
canreinvite=no
context=from-trunk
fromuser=888XXXXX
qualify=no
secret= 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.

**** Note: It seems that Pennytel is rejecting Asterisk. To overcome this problem, it is necessary to add the following in the general section of your sip.conf.**

```
useragent = MyPABX
```

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

My **sip.conf** looks like this;

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



```
[general]

port = 5060 ; Port to bind to (SIP is 5060)
bindaddr = 0.0.0.0 ; Address to bind to (all addresses on machine)
disallow=all
allow=ulaw
allow=alaw
; If you need to answer unauthenticated calls, you should change this
; next line to 'from-trunk', rather than 'from-sip-external'.
; You'll know this is happening if when you call in you get a message
; saying "The number you have dialed is not in service. Please check the
; number and try again."
context = from-sip-external ; Send unknown SIP callers to this context
callerid = Unknown
tos=0x68
useragent = MyPABX

; #, in this configuration file, is NOT A COMMENT. This is exactly
; how it should be.
#include sip_nat.conf
#include sip_custom.conf
#include sip_additional.conf
```

4.3.2 [Oztell](#) (SIP)

Oztell requires the numbers dialed 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+NXXXXXXXXX	Using the Oztell 06 rate to dial any NSW number.
0661+NXXXXXXXXX	Using Oztell 06 rate to dial any Australian number.
61+1300XXXXXXXX	Dials 1300 numbers
61+13ZXXX	Dials 13 numbers
61+1800XXXXXXXX	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+NXXXXXXXXX
0661+NXXXXXXXXX
61+1300XXXXXXXX
61+13ZXXX
61+1800XXXXXXXX
```

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

Outgoing Settings

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

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

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

```
allow=ulaw&alaw
canredirect=no
disallow=all
fromdomain=oztralia
fromuser=899060xxxxx
host=sip.oztralia.com
insecure=very
secret=Password
type=peer
username=899060XXXXX
```

Incoming Settings

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

In the **User Details** enter the following:

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

Registration

In the registration field, enter the following:

```
899060XXXXX:Password@sip.oztralia.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.

4.3.3 [Oztell](#) (IAX)

Since I only have IAX trunk with Oztell, I might as well include my Oztell IAX trunk setting here for general information.

*Note** Don't forget to Forward UDP Port 4569 to your Asterisk IP address and edit your [iax.conf](#) file.

Outgoing Settings

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

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

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

```
allow=ulaw&alaw
disallow=all
host=iax.oztralia.com
secret=Password
type=peer
username=899060XXXXX
```

Incoming Settings

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

In the **User Details** enter the following:

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

Registration

In the registration field, enter the following:

```
899060XXXXX:Password@iax.oztralia.com
```

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

4.3.4 [Astratel](#)

To make an international call, Astratel requires the numbers dialed to follow the international format **Access Code (0011) + Country code + area code + phone number** e.g. 0011 60 6123456. To call an Australian number e.g. Sydney 67541234, Astratel requires it in the following Australian format, **Area Code + Phone number** e.g. 02 67541234. Therefore the dialing rule will be in this form:

We need several patterns here:

Pattern	Explanation
8888XXXX	This pattern tells Astratel to dial the inter Astratel's numbers
0011+ZXX.	This pattern dials all international numbers by prefixing it with 0011 since the Outgoing routing stripped the 0011 before the number is sent to the trunk.
0+4XXXXXXXXX	Using Astratel to dial any Australian Mobile number

Your **Outgoing Dial Rules** box will look like this;

```
8888XXXX
0011+ZXX.
0+4XXXXXXXXX
```

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

Outgoing Settings

In the **Trunk Name** field we will enter the name of this trunk: **astrasip** (you can call it whatever you like)

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

Note: Astratel requires **fromdomain=sip03.astrasip.com.au without which it may not work. Entries in bold are mandatory, while the rest can be left out.*

```
allow=alaw&ulaw
disallow=all
fromdomain=sip03.astrasip.com.au
fromuser=8888XXXX
host=sip03.astrasip.com.au
insecure=very
secret= Password
type=peer
username=8888XXXX
```

Incoming Settings

In the **User Context**, we enter the Astratel issued User ID: 8888XXXX

In the **User Details** we will have the following (entries in bold are mandatory. Others can be left out).

```
canreinvite=no
context=from-trunk
fromuser=8888XXXX
insecure=very
qualify=no
secret=Password
type=user
username=8888XXXX
```

Registration

In the registration field, we will enter the following:

```
8888XXXX:Password@sip03.astrasip.com.au/8888XXXX
```

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

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

freePBX™ • Setup • Tools • Reports • Panel • Recordings

Setup
Add Trunk
Trunk: ZAP/gd

- Administrators
- Callback
- Conferences
- DISA
- Digital Receptionist
- Extensions
- Feature Codes
- Follow Me
- Gabcast
- General Settings
- Inbound Routes
- Misc Destinations
- On Hold Music
- Outbound Routes
- PIN Sets
- Paging and Intercom
- Queues
- Ring Groups
- System Recordings
- Time Conditions
- Trunks**
- VoicePulse

Add SIP Trunk

General Settings

Outbound Caller ID: <8882593347>

Maximum channels:

Outgoing Dial Rules

Dial Rules:

Dial rules wizards:

Outbound Dial Prefix:

Outgoing Settings

Trunk Name:

PEER Details:

```
canredirect=no
canreinvite=no
disallow=all
dtmfmode=rfc2833
host=sip.pennytel.com
insecure=very
secret=xxxxxx
type=peer
username=8882593347
```

Incoming Settings

USER Context:

USER Details:

```
canreinvite=no
context=from-trunk
dtmfmode=rfc2833
fromuser=8882593347
qualify=no
secret=xxxxxx
type=user
username=8882593347
```

Registration

Register String:

freePBX™ Version 2.1.1 on 192.168.1.29

** For more trunk settings, please refer to [APPENDIX B](#)*

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

5 CALLS HANDLING

5.1 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 or Queue. For this purpose, Inbound Route needs to be set up.

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

Initially, I have set all my incoming calls to be directed to Ring Group 10 as illustrated below (later on in this guide I will create a digital receptionist for this purpose).

The screenshot shows the FreePBX web interface for configuring an inbound route. The left sidebar lists various system settings, with 'Inbound Routes' highlighted. The main content area is titled 'Add Incoming Route' and contains the following configuration options:

- Administrators:** Administrators, Callback, Conferences, DISA, Digital Receptionist, Extensions, Feature Codes, Follow Me, Gabcast, General Settings, **Inbound Routes**, Misc Destinations, On Hold Music, Outbound Routes, PIN Sets, Paging and Intercom, Queues, Ring Groups, System Recordings, Time Conditions, Trunks, VoicePulse
- Navigation:** Setup, Tools, Reports, Panel, Recordings
- Buttons:** Setup, Add Incoming Route
- Add Incoming Route Form:**
 - DID Number:** 8882593347
 - Caller ID Number:** [Empty]
 - OR**
 - Zaptel Channel:** [Empty]
 - Fax Handling:**
 - Fax Extension:** disabled
 - Fax Email:** [Empty]
 - Fax Detection Type:** None
 - Pause after answer:** [Empty]
 - Privacy:**
 - Privacy Manager:** No
 - Options:**
 - Alert Info:** [Empty]
 - Set Destination:**
 - Ring Groups: AllPhones <10>
 - Core: Rohani Sharif <2000>
 - Custom App: [Empty]
- Submit** button

I only put in the following data. The rest are optional.

- **DID Number:** 8882593347 (This is my Pennytel SIP number)
- **Fax Extension:** Disabled (I do not require fax facility here)
- **Privacy Manager:** No
- **In the radio button, I enabled Ring Group 10 (AllPhones).** I want all my incoming calls from Pennytel trunk to ring all the phones.

For each extra trunk, a corresponding inbound route may be created. 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).

At least one Inbound Route must be created for TRIXBOX to answer incoming calls. This includes calls from PSTN trunk as well.

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.

The screenshot shows the FreePBX web interface. At the top, a red notification bar reads: "You have made changes - when finished, click here to APPLY them". Below this is the "treePBX" logo and a navigation menu with links for "Setup", "Tools", "Reports", "Panel", and "Recordings". On the left is a sidebar menu with items like "Administrators", "Callback", "Conferences", "DISA", "Digital Receptionist", "Extensions", "Feature Codes", "Follow Me", and "Gabcast". The main content area is titled "Add Incoming Route" and contains a form with the following fields: "Add Incoming Route" (a header), "DID Number:" with the value "8882593347", "Caller ID Number:" (empty), "OR", "Zaptel Channel:" (empty), and "Fax Handling". In the top right corner of the form area, there is a "Setup" link and a button labeled "Add Incoming Route" with the text "8882593347 / any CID" below it.

To handle the calls that come in without Trunk ID, you may create a Catch-All route. In my case, I simply create an **Inbound Route with the DID Number and Calling ID Number fields left blank**. This will have the effect of handling all calls from trunks that does not have an Inbound Route created (which includes all calls from PSTN and incorrectly created incoming route). 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 call may not work.

5.2 TIME CONDITIONDS

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

The screenshot shows the 'Add Time Condition' configuration page in the freePBX web interface. The page has a navigation menu on the left with options like Administrators, Callback, Conferences, DTS, Digital Receptionist, Extensions, Feature Codes, Follow Me, General Settings, Inbound Routes, Misc Destinations, On Hold Music, Outbound Routes, PIN Sets, Paging and Intercom, Queues, Ring Groups, System Recordings, Time Conditions (highlighted), and Trunks. The main content area is titled 'Add Time Condition' and includes a 'Server time: 17:42:31' indicator and an 'Add Time Condition' button. Below this, there are two sections: 'Destination if time matches:' and 'Destination if time does not match:'. Each section contains radio buttons for 'Core', 'Time Conditions', 'Ring Groups', 'IVR', 'DISA', and 'Custom App', with corresponding dropdown menus and input fields. A 'Submit Changes' button is located at the bottom of the form. The footer of the page displays the freePBX logo and the version information 'Version 2.1beta1 on 192.168.1.100'.

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

5.3 OUTBOUND ROUTES

5.3.1 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.

To create a new route using **freePBX**, select **Setup** tab and then select the **Outbound Route** option from the vertical menu on the left.

The screenshot shows the 'Edit Route' configuration page in freePBX. The route name is 'Pennytel'. The dial patterns list includes: 777, 84848810, 88601575, 88836804, 88897446, 17361X., 60386572X., and 8882XX. The trunk sequence is configured with two trunks: 0 SIP/SipPennytel and 1 SIP/SipPennytel-2. The 'Submit Changes' button is visible at the bottom.

5.3.2 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 and if matches, it will initiate the call using that trunk. If it does not match, it will compare the number with the pattern you have defined with route 2 and so on.
- Pass the number to the appropriate trunk to make the call.

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

In my case I created 5 routes:

- International
- Domestic
- MobileAust
- OztellOnly

- AstratelOnly

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 three will do as the names suggest.

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

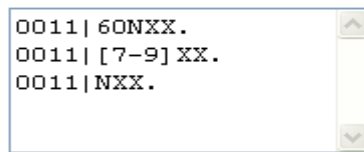
5.3.3 International

You will need the following pattern to match international dialing.

Pattern	Explanation
0011 60NXX.	Most of my 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 dialing including International Mobile (excluding USA which is a 1).

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

Dial Patterns



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

Trunk Sequence



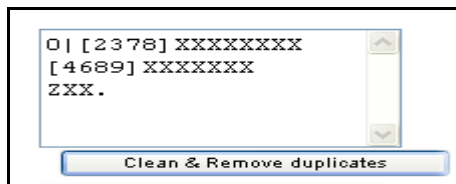
Any number dialed with the 0011 prefix will be routed through the Astratel trunk and should it fail, it will be routed to Pennytel.

5.3.4 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]XXXXXXXX	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 dialing 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.



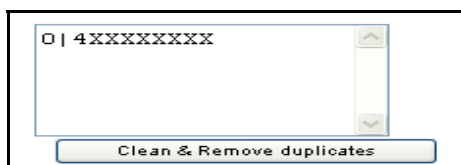
You will then need to pick the Trunk Sequent. In this case I need Oztell as the trunk for this route and if it fails, I want Astratel to take over as a second choice/backup



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

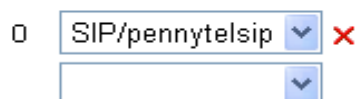
5.3.5 MobileAust

I have only a single pattern for calling Australian mobile – illustrated below.



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

Trunk Sequence



5.3.6 Oztellonly

I have the following pattern in my Oztellonly route. I 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]NXXXXXXXX	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 dialing 1300, 1800 and 13 numbers.
899060XX.	For dialing Oztell inter Oztell users
NXXXXXXXX	Catchall for dialing a local number that doesn't start with 1

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

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.

5.3.7 Astratelonly

There is only one pattern for Astratel only calls. It is the Astratel DID for calling inter-Astratel users.

I use 1278 as a unique number to identify that this number must be dialed through Astratel.

Correspondingly, there is only one trunk selected.

Trunk Sequence

5.4 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.*

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

The screenshot shows the 'freePBX' web interface. On the left is a sidebar with navigation links: Administrators, Callback, Conferences, DISA, Digital Receptionist, Extensions, Feature Codes, Follow Me, General Settings, Inbound Routes, Misc Destinations, On Hold Music, Outbound Routes, PIN Sets, Paging and Intercom, Queues, Ring Groups. The main content area is titled 'Edit ENUM Trunk'. It contains a 'Delete Trunk' link, a status 'In use by 1 route', and a 'General Settings' section with 'Outbound Caller ID' set to '"Ben Sharif" <612473641E' and 'Maximum channels' as an empty input field. Below is the 'Outgoing Dial Rules' section, which includes a 'Dial Rules' list with three entries: '612+NXXXXXXXX', '61+NXXXXXXXX', and 'ZXXXXXXXX.'. There is a 'Clean & Remove duplicates' button below the list. The 'Dial rules wizards' dropdown is set to '(pick one)'. The 'Outbound Dial Prefix' is an empty input field. At the bottom right is a 'Submit Changes' button.

6. Click the **Submit Changes** button at the bottom of the page
7. Click the **You have made changes - when finished, click here to APPLY them** at the top of the page.

5.4.2 Setting up ENUM Outbound Route

1. Click the **Outbound Routing** link on the left hand side.
2. In the routes window on the right-hand side, click the route you would like to enable with ENUM. I call my route **e164**.
3. I decided to use the prefix 164 to force dial through ENUM
4. In the Dial Pattern, on the first line, I had 1640|[378]NXX. This will allow me to dial an STD number such as 0732123456 where only the 732123456 will be sent to the ENUM trunk.
5. On the second line I have 1640011|NXX. This will allow me to call an International number eg. To call Malaysia I will dial 16400116062861234 and only the 6062861234 will be sent to the ENUM trunk.
6. On the third line I have 164|NXX. This enables me to dial NSW local number without having to dial 02.

7. Make the ENUM trunk the first route in your **Trunk Sequence**. You may also add other trunks in the sequence in case ENUM failed.

8. Click the **Submit Changes** button at the bottom of the page.
9. Click the **You have made changes - when finished, click here to APPLY them** at the top of the page.
10. 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.](#)

5.5 HOW TO RESTRICT EXTENSIONS ACTIVITIES

This requirement often manifests itself in an office environment where management restricts staff from making International, Long Distance or even local calls. In many organization, staff are restricted to making internal calls only.

There is no easy way of imposing this restriction through the freePBX GUI. The only way that that can be done currently is to introduce a special script in the `extensions_custom.conf` file or create a special `.conf` file. For now, we will stick with the simplest method.

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 **Config Edit** of TRIKBOX, 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-TeleGlobal
include => outrt-006-Astratel
include => outrt-007-MyNetFone
include => outrt-008-SpanTalk
include => outrt-009-VoIPBuster
include => outrt-010-Domestic
include => outrt-011-MobileAust
include => outrt-012-International
include => outrt-013-12Number
include => outrt-014-13N1800Numbers
include => outrt-015-USFree
include => outrt-016-e164
include => outrt-017-LA-Link
include => outrt-018-UK-Link
include => outrt-019-SingaporeLink
include => outrt-020-KualaLumpurLink
include => outrt-021-MalaccaLink
include => outrt-022-PenangLink
exten => foo,1,Noop(bar)

; end of [outbound-allroutes]
```

Highlight them, copy and paste in a notepad text file somewhere. We will get back to this a little later.

3. Through **Config Edit** of TRIKBOX, open **`extensions_custom.conf`** and create a section like my example below. So will have to do some typing

because you cant cut and paste this from another .conf file (you can cut and paste from here though ☺)

```
[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 outbound routes #
; #####
;
```

Now copy your [outbound-allroutes] that was saved to the text file directly underneath the last line of the above context. 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 outbound routes #
; #####
;
[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-TeleGlobal
include => outrt-006-Astratel
include => outrt-007-MyNetFone
include => outrt-008-SpanTalk
include => outrt-009-VoIPBuster
include => outrt-010-Domestic
include => outrt-011-MobileAust
include => outrt-012-International
include => outrt-013-12Number
include => outrt-014-13N1800Numbers
include => outrt-015-USFree
include => outrt-016-e164
include => outrt-017-LA-Link
include => outrt-018-UK-Link

```

```
include => outrt-019-SingaporeLink
include => outrt-020-KualaLumpurLink
include => outrt-021-MalaccaLink
include => outrt-022-PenangLink
exten => foo,1,Noop(bar)
;
; end of [outbound-allroutes]
```

Now add the following exten 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.

6 SYSTEM RECORDINGS

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

By now you should be able to navigate yourselves through the freePBX menu. To get to the system recordings; **FreePBX >> Setup >> System Recordings** and you will get to the following screen.

The screenshot shows the 'freePBX' interface for 'System Recordings'. The top navigation bar includes 'Setup', 'Tools', 'Reports', 'Panel', and 'Recordings'. The left sidebar lists various system settings. The main content area is titled 'System Recordings' and 'Add Recording'. It features two steps: 'Step 1: Record or upload' and 'Step 2: Name'. Step 1 includes a text input for an extension number and a 'Go' button, and an alternative upload section with a 'Browse...' button and an 'Upload' button. Step 2 includes a 'Name this Recording:' text input. A 'Save' button is at the bottom right. A top navigation bar shows 'Setup', 'Tools', 'Reports', 'Panel', and 'Recordings'.

There are two methods that you can capture your recording;

- By recording directly using your telephone
- Uploading pre-recorded materials

6.1 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.

The screenshot shows the 'freePBX' interface for 'System Recordings'. The top navigation bar includes 'Setup', 'Tools', 'Reports', 'Panel', and 'Recordings'. The left sidebar lists various system settings. The main content area is titled 'System Recordings' and 'Add Recording'. It features three steps: 'Step 1: Record or upload', 'Step 2: Verify', and 'Step 3: Name'. Step 1 includes instructions to dial *77 and speak the message. Step 2 includes instructions to dial *99 to listen to the recording. Step 3 includes a 'Name this Recording:' text input. A 'Save' button is at the bottom right. A top navigation bar shows 'Setup', 'Tools', 'Reports', 'Panel', and 'Recordings'.

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

In my message, I have asked Rohani to record the following:

“Hi, you have reached the Sharif’s household. Press 1 for Ben, 2 for Rohani, 3 for Norsurya, or you may hold for someone to answer your call if anyone is available. 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 message.

If you wish to re-record your message, dial *77 again.

I then verify my recording by dialing *99. I can go to step 1 again if I am not happy with the recording. Keep doing this till you are satisfied.

Once I am happy with the recording, I gave the recording a descriptive name “**MainMenu-Rohani1**” so I know what the recording is for.

Click Continue.

6.2 UPLOAD PRE-RECORDED MATERIAL

If you have a recording that you have prepared in a .wav format you can simply upload that recording

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.

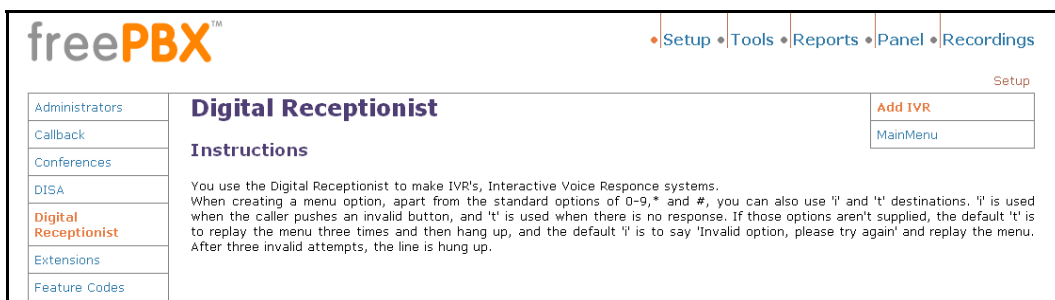
The above recorded files either recorded manually or uploaded using the system recording facility will be stored in the `/var/lib/asterisk/sounds/custom` directory.

7 DIGITAL RECEPTIONIST

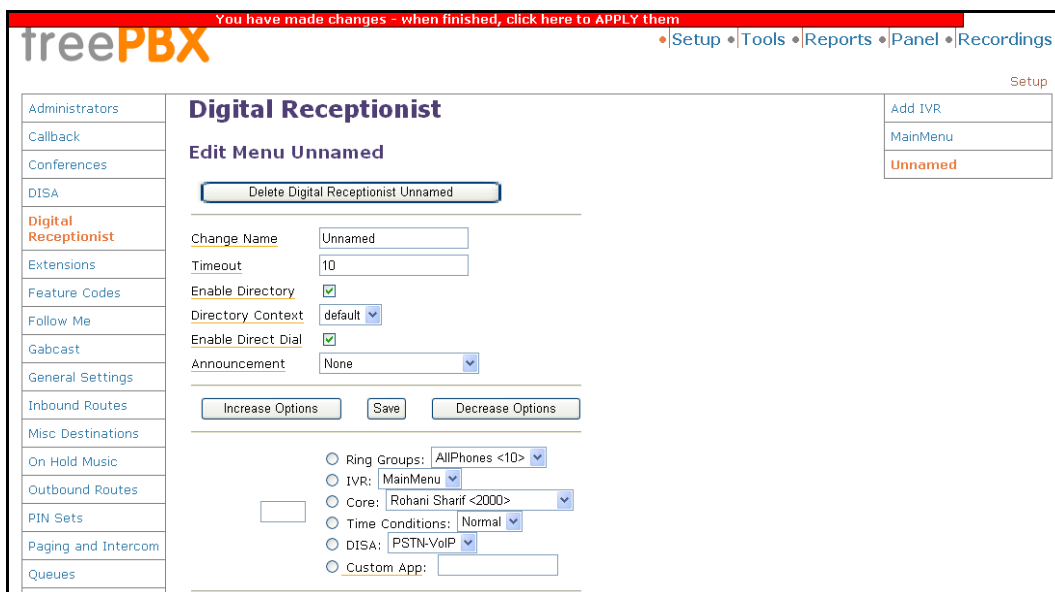
I found that the digital receptionist is very handy to filter some of the phone calls I have been receiving. As in most households, no one will pick up the phone when it rings. One is waiting for the other. Yours truly usually is the one who will pick it up and has to look for the recipient... usually my teenage daughter.

7.1 SETTING UP DIGITAL RECEPTIONIST

In the Setup tab, click on the Digital Receptionist option and get the following screen.



Select **Add IVR** option on the top right on the screen and the new Unnamed IVR configuration screen will be presented to you.



You then need to fill in the following fields:

Name (Change Name): MainMenu (*that's what I called m main menu*)

Timeout: 10 (*I left it at default*)

Enable Directory: Tick

Directory Context: Default

Enable Direct Dial: Tick

Announcement: MainMenu-Rohani1 (*That's what I called my IVR recording that I made for this purpose*)

(see illustration below)

The screenshot shows the 'Digital Receptionist' configuration interface in FreePBX. The main heading is 'Digital Receptionist' with a sub-heading 'Edit Menu MainMenu'. A 'Delete Digital Receptionist MainMenu' button is visible. The configuration fields include: 'Change Name' (MainMenu), 'Timeout' (10), 'Enable Directory' (checked), 'Directory Context' (default), 'Enable Direct Dial' (checked), and 'Announcement' (MainMenu-Rohani1). Below these are three buttons: 'Increase Options', 'Save', and 'Decrease Options'. At the bottom, there are radio button options for: Ring Groups (AllPhones <10>), Conferences (General Conference <8200>), IVR (MainMenu), Core (Ben Sharif <2001> - selected), Time Conditions (Normal), DSA (PSTN-VolP), Misc Destinations (ForVoice Mail), and Custom App.

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 my IVR, I have defined my requirement as outlined below:

For number 1 & 2, I selected **Extension**, for 3 & 4 I have selected **Custom App:** and for 5 which I have changes to the letter “t”, I selected **Ring Group**.

- For 1, I selected my extension from the drop down selection list.
- For 2, I selected Rohani’s extension.
- For 3, which is for Norsurya, I have selected Custom App: to call her mobile.
- For 4, I selected Custom App: for DISA (we will get to DISA a little later).
- For 5, (I changed it to read “t” for timeout), I selected Ring Group 1

To call Norsurya’s mobile I have included the following context in the **extensions_custom.conf** file.

```
[custom-callmobile]
; to direct calls to Norsurya’s mobile number.
exten => s,1,playback(pls-wait-connect-call)
exten => s,2,Dial(SIP/6140XXXXXXX@pennytelsip,60,r)
exten => s,3,Hangup
```

The last extension (which I 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 Ring Group 1.

Once finished, click **Save** and also click on the **red bar** on the top of the scree.

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

7.2 CUSTOMISING INDIVIDUAL EXTENSION

Note for home users

Most of the config files manual tweaks and extra configurations in this guide are meant for advanced users. Unless you have a compelling reason to do so, freePBX users are advised against it.

When the Digital Receptionist answers the call and directing the call to the appropriate extension, the caller will hear another ringing tone until the extension is answered. This is boring, as I prefer music instead, besides, it sounds more pleasant.

To achieve this, instead of nominating my extension number for the choice in Digital Receptionist, I use a custom extension. To do this, I create a custom extension in the `extensions_custom.conf` called `[custom-myextension]`.

See example below:

```
[custom-myextension]
;
exten => s,1,playback(pls-wait-connect-call)
exten => s,2,Dial(SIP/2001,60,m) ; Ben's extension
exten => s,3,Macro(vm,2001)
exten => s,4,Hangup
```

In the example above;

- Line 1 (priority 1) will tell the caller to wait while he/she is being connected.
- Line 2 (priority 2) will call extension 2001 (my extension) and will try calling the extension for 60 seconds. While waiting for me to answer the call, the caller will be listening to soothing music – hence the “m”, or, if “r” the caller will hear the ringing tone instead.
- Line 3 (priority 3) will direct it to my voicemail if the call is not answered within 60 seconds.

In digital receptionist, instead of selection my extension for my option, I will nominate Custom App:

Custom App `custom-myextension,s,1`

You can do this to all the individual extension or you can call a number of extensions simultaneously like a ring group – like so.

```
exten => s,1,playback(pls-wait-connect-call)
exten => s,2,Dial(SIP/2001&SIP/2002&SIP/2003,60,m)
exten => s,3,Macro(vm,2001); voice mail to my extension
exten => s,4,Hangup
```

8 GET UNDER THE BONNET

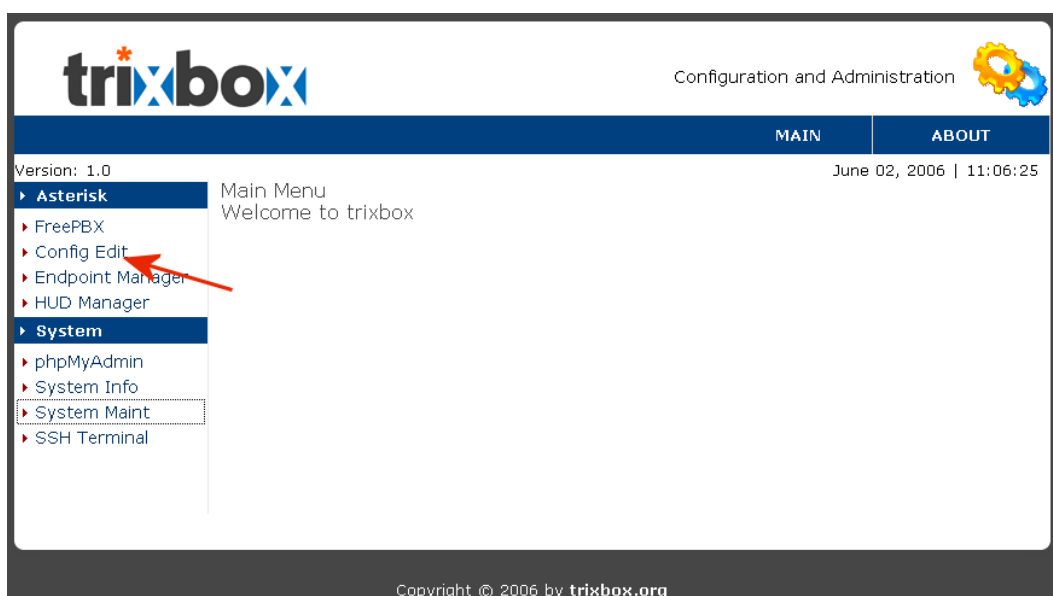
Note for home users

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 TRIKBOX as well.

To do this you 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 Config Edit.

To start editing the **.conf** files you need to log in to TRIKBOX Configuration and Administration and select the Config Edit selection.



Select **Config Edit** (marked with red arrow) and you will see a new screen with a list of all the **.conf** files, which can be edited manually.



You may scroll down the page to find the file that you wish to edit.

8.1 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**, **iax.conf** (just in case you want IAX, but not really necessary if you are just running SIP), **indications.conf**, **enum.conf**, **extensions_custom.conf**, **zaptel.conf** and **modprobe.conf** (the last 2 files live in the **/etc** directory and may need to be edited if you have a FXO card e.g. X100P installed).

8.1.1 sip.conf

```
[general]

port = 5060           ; Port to bind to (SIP is 5060)
bindaddr = 0.0.0.0   ; Address to bind to (all addresses on
machine)
disallow=all
allow=g729
allow=ulaw
allow=alaw
allow=gsm

defaultexpiry=600   ;**Engin users: include this if necessary
maxexpiry=3600      ;**Engin users: include this if necessary

qualify=yes
progressinband=yes

; If you need to answer unauthenticated calls, you should
; change this next line to 'from-trunk', rather than
; 'from-sip-external'.
; You'll know this is happening if when you call in you get
; a message saying "The number you have dialed is not in
; service. Please check the number and try again."

context = from-sip-external ; Send unknown SIP callers to this context
callerid = Unknown
tos=0x68

language=my

; #, in this configuration file, is NOT A COMMENT. This is exactly
; how it should be.
#include sip_nat.conf
#include sip_custom.conf
#include sip_additional.conf
;#include additional_a2billing_sip.conf
```

8.1.2 iax.conf

In the **general** section, add the following lines if it is not already there:

```
[general]
delayreject=yes
jitterbuffer=yes
mailboxdetail=yes
```

8.1.3 Indications.conf

In the general section ensure the following exist:

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

8.1.4 enum.conf

```
[general]
;
; The search list for domains may be customized.
; Domains are searched in the order they are listed here.
;
search => e164.org
search => e164.arpa
;h323driver => H323 ; this line exists in AAH v2.8
(There should only be the three entries above).
```

8.1.5 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.

Immediately after the line following the **include => from-internal-trixbox** context, insert the following codes:

```
exten => 0,1,Dial(ZAP/1/{EXTEN})
exten => 0,2,Hangup
```

This will allow you to dial 0 and you will get a second dial tone from PSTN. Following which you can dial a PSTN number that you wish to call.

Note: * If you're using a PRI line, you need to enable 'overlapdial' in /etc/asterisk/zapata.conf for this to work.

9 CHECK YOUR SYSTEM

After doing all of the above, you will no doubt feel frustrated if your VOIP does not work.

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.

There are two places where you can obtain a quick health report of your system – the System Process Status and the Asterisk Info screens.

Note: *If you don't have the Asterisk Info option, refer to the NOTES section on how to obtain this module.*

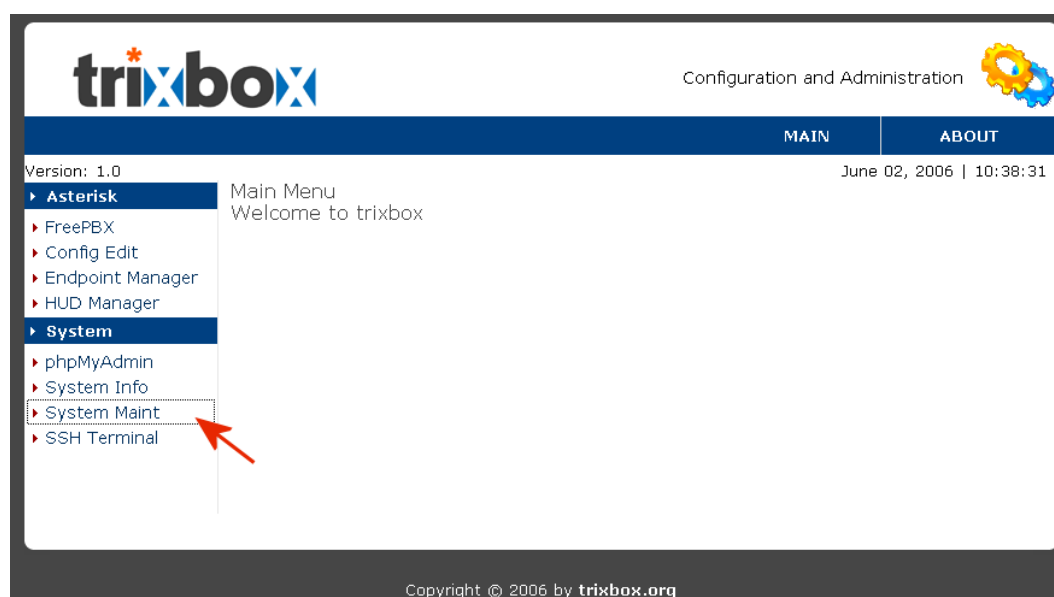
9.1 SYSTEM PROCESS STATUS

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

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

To do the initial checking, let us start with the System Maint option of TRIXBOX Configuration and Maintenance Menu.

Select the System Maint selection as highlighted by the red arrow below.



Clicking on the System Maint selection will open the Process Status screen.

When your maintenance screen is presented to you, all the processes should be running and highlighted in green as illustrated in the Process Status screen below.

If any of the four services aren't showing green and running, chances are something is wrong and your Asterisk 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.

This may be a good time to ensure that the minimum requirements have been met. If the minimum and the right type of hardware requirements have been met, and yet it is not working, maybe its time to start the installation again.

Next, select and click **Asterisk Info** to verify your status.

9.2 ASTERISK INFO

From freePBX select the Tools Tab and click on the Asterisk Information option as shown by the red arrow.

When presented with the info 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.

Sip Peers		
Host	Username	Refresh State
sip.spantalk.com:5060	98883457	105 Registered
sip.myfone.com.au:5060	09133432@sip	120 Request Sent
sip03.astrasip.com.au:5060	88881346	105 Registered
210.80.182.142:5060	222261282150	585 Registered
Xsip.pennytel.com:5060	60386572572	120 Unregistered
Xsip0.oztralia.com:5060	899060194748	120 Unregistered
sip.pennytel.com:5060	8882593347	280 Registered
proxy01.sipphone.com:5060	17476025275	105 Registered
fwd.pulver.com:5060	651819	105 Registered
sip.voipbuster.com:5060	bsharif	105 Registered

Verbosity is at least 4
Core debug is at least 4

IAX2 Sip Registry			
Host	Username	Perceived	Refresh State

Verbosity is at least 4
Core debug is at least 4

IAX2 Peers					
Name/Username	Host	Mask	Port	Status	
TRIXBOX	192.168.1.101	(S) 255.255.255.255	4569	OK (1 ms)	
Malacca-IPBX/61	60.51.36.202	(S) 255.255.255.255	4569	Unmonitored	
CrossLink	60.51.36.202	(S) 255.255.255.255	4569	UNREACHABLE	
60600/60600	(Unspecified)	(D) 255.255.255.255	0	Unmonitored	

4 iax2 peers [1 online, 1 offline, 2 unmonitored]
Verbosity is at least 4
Core debug is at least 4

If all trunks are registered, you can be almost certain that at the very least, your TRIXBOX 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 Codecs, Routings and Dialling rules errors. Routings and Dialling rules are something that needs to be thought out logically with no simple way of determining, as different people has different requirements and different VSP have different dialling rules.

10 INTERFACING ASTERISK TO PSTN

If you are using TRIXBOX v1.0, please refer to section 10.3 before proceeding as there is a little problem of TRIBIX not recognising ZAP device. I hope this issue will be addressed in later versions of TRIXBOX.

10.1 DIGIUM WILDCARD X100P FXO PCI CARD

This card allows you to connect a POTS (plain Old Telephone System) line to your Asterisk box ([See Notes for Patch information](#)).

If this card is added after Asterisk has been configured, it may be necessary to configure it by using the zaptel card auto-config utility so the correct zaptel driver will be set up. To do that, enter the following from the command line.

```
rebuild_zaptel
```

Restart Asterisk

```
Shutdown -r now
```

After restart run the following;

```
genzaptelconf
```

Restart again.

```
Shutdown -r now
```

Next 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 dialing
5. Set the **Zap Identifier** to **1** (the default is g0)

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.

If you have this card installed, you may need to edit the following files (*usually you don't have to*); **zapata.conf**, **Zapata-auto.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 as **you cannot edit them through Config Edit**.

10.1.1 zapata.conf

Under [channels] add the following lines:

```
[channels]  
busydetect=yes  
busycount=6
```

The above is to ensure that the line is hung up if there is no answer or busy after 6 busy tones.

For my 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 a little with this setting)
txgain=2 (you may have to experiment a little with this setting)
```

10.1.2 Zapata-auto.conf

Ensure the following exist in `zapata_auto.conf`. It is located at the end of the file.

```
context=from-pstn
group=0
channel => 1
```

Leave the rest of the file as it is.

10.1.3 zaptel.conf

Change the `loadzone` and `defaultzone` to "au"

```
# Global data
loadzone = au
defaultzone = au
```

10.1.4 modprobe.conf

For usage in Australia, add the line highlighted in **Red** below:

```
alias char-major-196 torisa
options wcfxo opermode=AUSTRALIA ; add this line
install tor2 /sbin/modprobe --ignore-install tor2 && /sbin/ztcfg
```

10.2 DIGIUM TDM400P FXO/FXS CARD

Like the Digium Wildcard X100P, this card allows you to connect a POTS (plain Old Telephone System) line to your TRIXBOX box. Unlike the X100P, 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.

If this card is installed after Asterisk has been loaded, you will need to configure it just like the X100P by using the following command on the command line:

```
rebuild_zaptel
```

Restart Asterisk

```
Shutdown -r now
```

After restart run the following;

```
genzaptelconf
```

Restart again.

Shutdown -r now

10.2.1 zapata-auto.conf

Next, using config edit, look in the **zapata-auto.conf** file and you will see a list of all your channels in your TDM400P. Set up the trunks as trunks and the extensions as extensions in freePBX.

When you open the **Zapata-auto.conf** file, it will look something like the illustration below (see the red highlight)

```
; 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
;
callerid=asreceived

; Span 1: WCTDM/0 "Wildcard TDM400P REV E/F Board 1"
signalling=fxs_ks
; Note: this is a trunk. Create a ZAP trunk in AMP for Channel 1
context=from-pstn
group=0
channel => 1
; channel 2, WCTDM, inactive.
; channel 3, WCTDM, inactive.
; channel 4, WCTDM, inactive.
```

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

Once this is done, reboot your PC and when Asterisk starts, use freePBX 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 X100P card in the previous section.

10.2.2 modprobe.conf

You may need to edit the **modprobe.conf** to add the necessary option for usage in Australia.

Locate the line "install wctdm /sbin/ztcfg-- --ignore-install wctdm && /sbin/ztcfg" and edit it to reflect the following:

```
install wctdm opermode=AUSTRALIA fxshonormode=1 booster=1 /sbin/ztcfg-- --
ignore-install wctdm && /sbin/ztcfg
```

Note: as of Zaptel Drivers 1.2.4, by selecting **opermode=AUSTRALIA** the zaptel drivers automatically add the "booster=1 , fxshonormode=1". Also see [Appendix E.3](#) (Users Suggestions)

10.3 REBUILD ZAPTEL DRIVER

This section should really be in 10.1 but I hope in later versions of TRIBOX, this issue will be fixed.

If you are using TRIBOX v1.0 you will need to do a rebuild the Zaptel Driver as ZAP device does not work the way it should with TRIBOX v1.0. The /usr/src/zaptel directory is visibly missing.

To do this, you will need to download the zaptel-1.2.7.tar.gz file from Digium and install it as described *below (at the time of this update, it is version 1.2.7. By the time you read this, the version may have changed. It is a good idea for you to check what the latest version is).*

You will have to log on to the TRIBOX system. You can either do this at the System box itself or through your favourite SSH client. I use Putty.

At the command prompt, do the following:

```
cd /usr/src/  
wget http://ftp.digium.com/pub/zaptel/zaptel-1.2.7.tar.gz
```

once the file is downloaded, unpack the file;

```
tar -zxvf zaptel-1.2.7.tar.gz
```

After the files have been unpacked, move them to the /usr/src/zaptel directory:

```
mv zaptel-1.2.7 zaptel
```

Leave it there for now, as we need to fix an inherent bug in the Centos release of Red Hat.

The following is the fix (source Nerd Vittles <http://nerdvittles.com/index.php?p=123>)

Log into your new server as root and issue the following commands:

```
cd /usr/src/kernels/2.6.9-34.0.1.EL-i686/ include/linux  
(depending on the version you have. At the time of writing it is 2.6.9-34.0.1.EL-1686).  
mv spinlock.h spinlock.h.old  
wget http://nerdvittles.com/trixbox/spinlock.h
```

Alternatively, after you change to the directory, instead of getting the file from NerdVittles, you may edit the spinlock.h file manually. This is what you have to do:

Edit the spinlock.h file. I use nano to do it using the following command (depending on the version you have. At the time of writing it is 2.6.9-34.0.1.EL-1686).

```
nano /usr/src/kernels/2.6.9-34.0.1.EL-i686/include/linux/spinlock.h
```

Locate the line entry below, at line 407.

```
#define DEFINE_RWLOCK(x) rw_lock_t x = RW_LOCK_UNLOCKED
```

You need to change this entry to reflect the line below:

```
#define DEFINE_RWLOCK(x) rwlock_t x = RW_LOCK_UNLOCKED
```

Once done, save the file.

Now do the following:

```
cd /usr/src/zaptel
make clean
make install
```

Now it might be a good idea to create the fxotune.conf file that is missing. Do the following:

```
touch /etc/fxotune.conf
/usr/sbin/fxotune -s
```

Once done, reboot using the following command:

```
reboot
```

When the reboot completes, you can start rebuilding the support for your ZAP devices or for that matter, ztdummy if you don't have any ZAP devices.

Log in as root and type the following command:

(Every time there is a kernel update with yum (which is the case with Asterisk and CentOS), ZAP device support needs to be rebuilt using the new kernel and you will need to follow the same process below.)

```
rebuild_zaptel
```

Then reboot your system:

```
reboot
```

Now log in as root again and enter the following command:

```
amportal stop
genzaptelconf
```

Reboot once again:

```
reboot
```

..and you're done.

(See also user [Users' Suggestions](#))

Note: *While the procedure above seems to have fixed most TRIXBOX users' woes, it didn't with mine ... for me, it's back to reinstalling from scratch*

.. and after reinstall, still it will not detect my card. It seems that there is a detection issue with the procedures, so, this time armed with a bigger hammer I proceeded to use the brute force method.

10.3.1 IF ALL ELSE FAIL

If all else fail, do the following at the command prompt,

```
cd /etc/sysconfig
nano zaptel
```

Locate the line entry corresponding to your card e.g:

If you are using the X100P, you will be looking for this:

```
#MODULES="$MODULES wcfxo" # X100P - Single port FXO interface
Since I use TDM400P, in my case I need to locate the following line.
```

```
#MODULES="$MODULES wctdm" # TDM400P - Modular FXS/FXO interface (1-4 ports)
```

Remove the # so the line will read like this;

```
MODULES="$MODULES wctdm" # TDM400P - Modular FXS/FXO interface (1-4 ports)
```

Save the file and restart.

When the system restarts, log in and do the following at the command prompt:

```
amportal stop
genzaptelconf
```

Once completed, restart again;

```
Shutdown -r now
```

Your ZAP device will now be recognised.... Phewww, that was a task and a half.

10.3.2 How to overcome the Zaptel 1.2.5 dependency

Did you get this error message when updating your Zaptel driver?

```
Error: Missing Dependency: zaptel = 1.2.5 is needed by package zaptel-
modules-2.6.9-34.EL
```

Yes, it is a pain in the proverbial.

If you are one of many who have been bugged by this, the easiest way around is to install 1.2.5 and be done with it.

Log in to TRIXBOX as root.

```
cd /usr/src
wget http://ftp.digium.com/pub/zaptel/releases/zaptel-1.2.5.tar.gz
tar -zxvf zaptel-1.2.5.tar.gz
mv zaptel-1.2.5 zaptel
cd /usr/src/zaptel
```

```
make clean
make install
rebuild_zaptel
modprobe wcfxo
genzaptelconf
reboot
```

after you have completed the above procedure, you will have the correct zaptel that trixbox update insists on having.

10.4 SIPURA SPA3000 AS A PSTN INTERFACE

To those new to the SPA3000, 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'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 TRIXBOX 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.

10.4.1 Log in to SPA3000

Login to your SPA-3000 as admin/advanced.

Before you change anything, I'd suggest taking a snapshot (i.e. just save the .html page) of your current SPA-3000 configuration, just in case you ever need to refer back to your own customisations.

If you're not already running the latest SPA-3000 firmware, then upgrade it to the latest version (at the time of writing, it's 3.1.5a). 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 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.

10.4.2 Change the settings

System tab

DHCP: No

Static IP: *something on your local subnet* e.g.; 192.168.1.200

NetMask: 255.255.255.0

Gateway: *your router's IP address* e.g.; 192.168.1.254

Primary DNS: *your ISP's primary DNS address* e.g.; 203.12.160.35

Secondary DNS: *your ISP's secondary DNS address* e.g.; 203.12.160.36

Regional tab

Dial Tone: 400@-19,425@-19;10(*0/1+2)

Busy Tone: 425@-10;10(.4/.4/1)

Reorder Tone: 425@-10;10(.2/.2/1)

Ring Back Tone: 400@-19,425@-19,450@-19;*(.4/.2/1+2+3,.4/.2/1+2+3,0/2/0)

User 1 tab

Default Ring: 3

Default CWT: 8

10.4.3 Add SIP Trunk

Then in freePBX, add a SIP trunk.

General Settings:

Outbound Caller ID: <0298765432> *(for example)*

Maximum Channels: 1

Dial Rules: 0+NXXXXXXXXX *(for example)*
0011+ZXXXXXXXXXXXX.

Trunk Name: telstra *(for example)*

Peer Details:

canreinvite=no

context=from-pstn

host=*the IP address of your SPA-3000 (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 (for example; 192.168.1.200)*

insecure=very

nat=no

port=5061 *for example*

type=user

username=PSTN

Leave "Register String" empty

10.4.4 SPA3000 as an outbound PSTN Trunk

To make PSTN calls using the SPA3000 you simply select the SPA3000 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.4](#) in Sipura SPA-3000

11 PSTN TO VOIP GATEWAY

Note: If you are using *TRIXBOX*, the following instruction may not apply as you can create *DISA* from *FreePBX* option without having to resort to the manual configuration below – see [DISA](#),

Now that the SPA-3000 has been set-up to accept PSTN, *TRIXBOX* can be turned into a PSTN to VOIP Gateway. You will not need a SPA-3000 if you have an FXO card installed.

I need such a gateway because I often have to call overseas from my office. Needless to say, we don't make private overseas calls from the office phone. I overcame this by calling my Asterisk box from the office and when it answered, I can then use asterisk to call overseas.

I accomplished this through *DISA*.

To do this, I manually edit the **extensions_custom.conf**, by adding the following entries at the end of the file,

```
[custom-pstn2voip]
; I use this to call VOIP from PSTN
exten => s,1,Answer
exten => s,2,DigitTimeout(5)
exten => s,3,ResponseTimeout(10)
exten => s,4,Authenticate(my password)
exten => s,5,DISA(no-password|from-internal)
```

I have Digital Receptionist answers all incoming calls (refer to Digital Receptionist in the previous section).

In the previous section I have 5 options in Digital receptionist e.g. Press 1 for Ben, 2 for Rohani, and 3 for Nosurya. I did not announce the 4th option as I use it for *DISA* and is only known to me, besides, it requires a password.

For option 4, I chose the radio button next to Custom App. In the custom app field I entered custom-pstn2voip,s,1

Custom App

When 4 is pressed, followed by the # key, during the announcement, the digital receptionist will ask for a **password**. On entering the correct **password**, the one defined in the example above, a dial tone will be heard. At this stage, you can start dialling the number you wish to call.

Option 5, (which I have changes to the letter 't', for timeout) is invoked if the caller does not select any option or if the DTMF tone is not recognised. This option can be directed to a Ring Group or to a human receptionist if you are using Asterisk in a business environment.

12 DISA

12.1 DISA - HAND CODING METHOD

I use this function in my PSTN to VOIP gateway above.

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.

The example below is how I have implemented my DISA.

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

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

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

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

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

12.2 DISA – USING FREEPBX

Those using TRIKBOX with freePBX as the User Interface will be able to configure DISA from freePBX without having to do the above manual configuration. The same thing can be achieved through freePBX as illustrated in the screenshot below.

The screenshot shows the freePBX web interface. At the top, there is a navigation bar with links for Setup, Tools, Reports, Panel, and Recordings. On the left, there is a sidebar menu with various system settings options. The main content area is titled 'DISA: PSTNVoIP (1)'. It includes a 'Setup' button and a 'Delete DISA PSTNVoIP' link. Below this is an 'Edit DISA' section with a form containing the following fields and values:

DISA name:	PSTNVoIP
PIN	4748
Response Timeout	10
Digit Timeout	5
Require Confirmation	<input type="checkbox"/>
Caller ID	"The Sheriffs" <61282150
Context	from-internal

A 'Submit Changes' button is located at the bottom of the form.

I have populated the various fields thus:

- **DISA name:** *PSTNVoIP* (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 PSTNtoVOIP as your DISA application.

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

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

13 SETTING UP A 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/download/sipphone/>

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. 201>

Password: 201 (I use the same number as password for simplicity. I use the same password when I set up the extension in Asterisk).

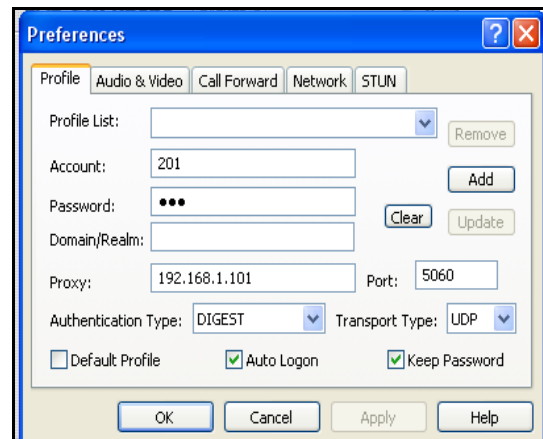
Domain/Realm: <leave it blank>

Proxy: Your Asterisk network address e.g. 192.168.1.101

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.

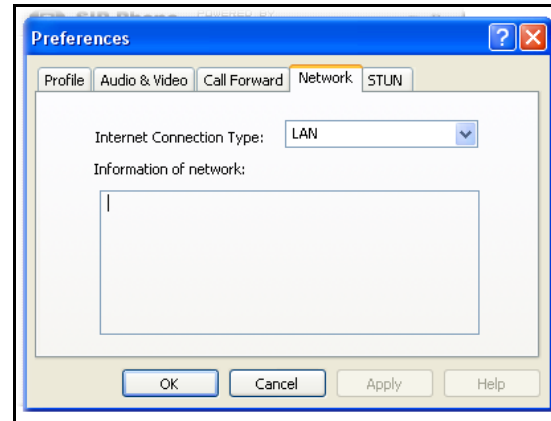
Click OK

Network Tab

Ensure that your Internet Connection Type is set to LAN.

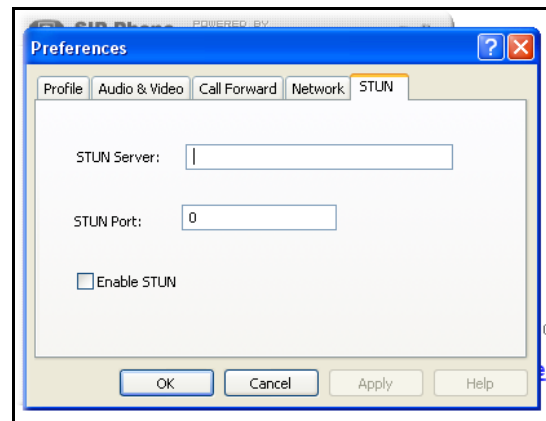
You may ignore the Information of Network field.

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.

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

14 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 TRIKBOX.

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 TRIKBOX and the support for HUD is already integrated within TRIKBOX.

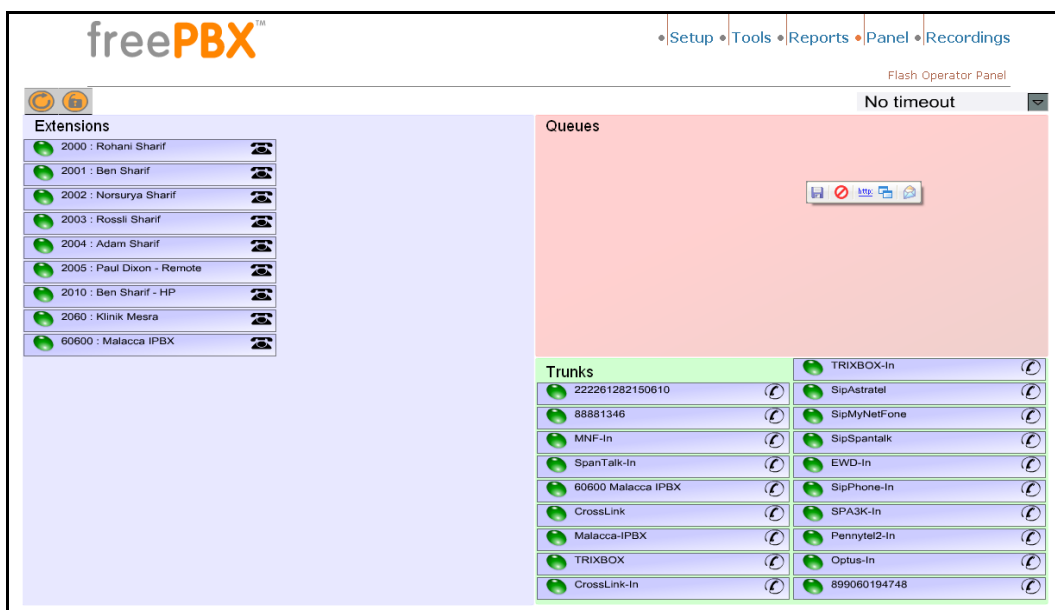
14.1 FLASH OPERATOR PANEL (FOP)

This is the standard Operator Panel that is already part of TRIKBOX – 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 either from within freePBX by clicking on the Panel option on the top right hand corner of the freePBX screen or from TRIKBOX Welcome screen.



The following information are displayed on FOP:

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

Functions you can perform on FOP:

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

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

14.1.1 Setting the Admin Password

It is desirable 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 TRIXBOX. Once logged in, change the directory to ***/var/www/html/panel***

```
cd /var/www/html/panel
```

Using **nano** as the editor, open the configuration file ***op_server.cfg***

```
nano op_server.cfg
```

Go to the line that says `security code=passw0rd`
(in FOP that comes with TRIXBOX, the default password is "passw0rd")

Replace the "passw0rd" with the password of your choice.

```
security_code=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.

14.1.2 Hang-up a Call

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

14.1.3 Transfer a Call

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

14.1.4 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.

14.1.5 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.

14.2 HUDLITE

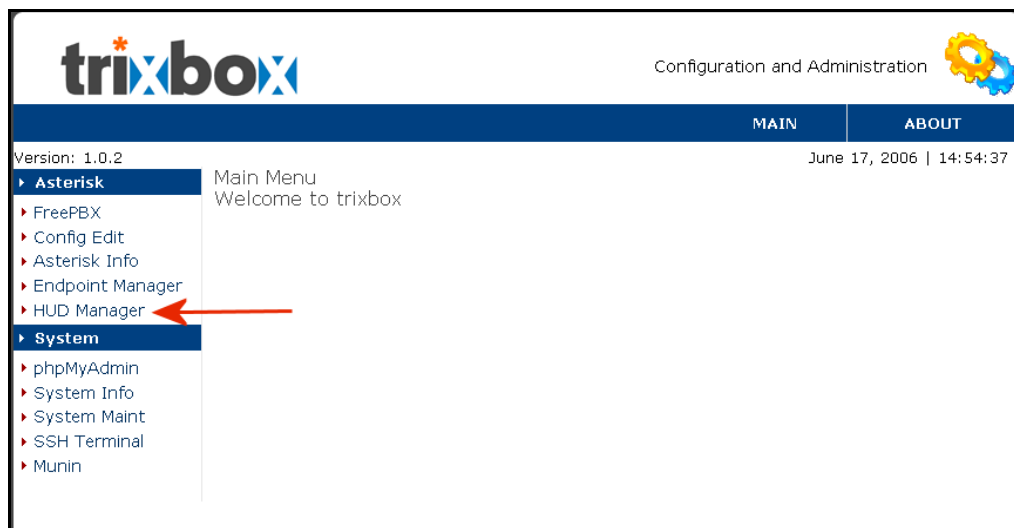
[HUDLite](#) from Fonality integrates very nicely with TRIXBOX and the HUD manager is already incorporated in TRIXBOX. It has most of the necessary functionalities (if not all) of a modern operator switch panel where you can tasks such as:

- Answer Calls
- Transfer calls
- Hang up calls
- Put calls on hold etc.
- Monitor status of extensions just like the default FOP that comes with TRIXBOX and more.

You can download HUDLite directly from the site [here](#).

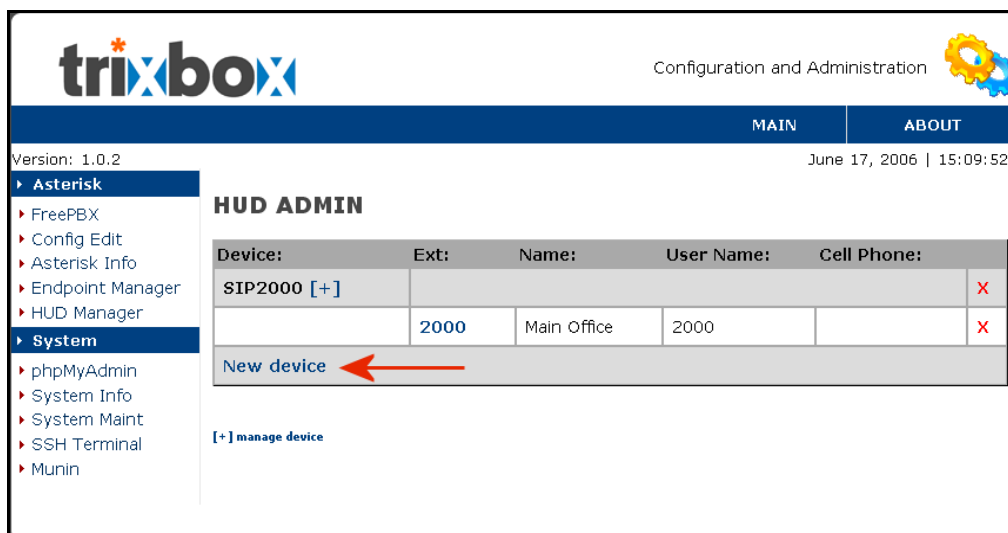
14.2.1 Setting up HUD Manager

Before you start installing HUDLite, you will need to populate HUD Manager from your TRIXBOX Configuration and Administration screen.



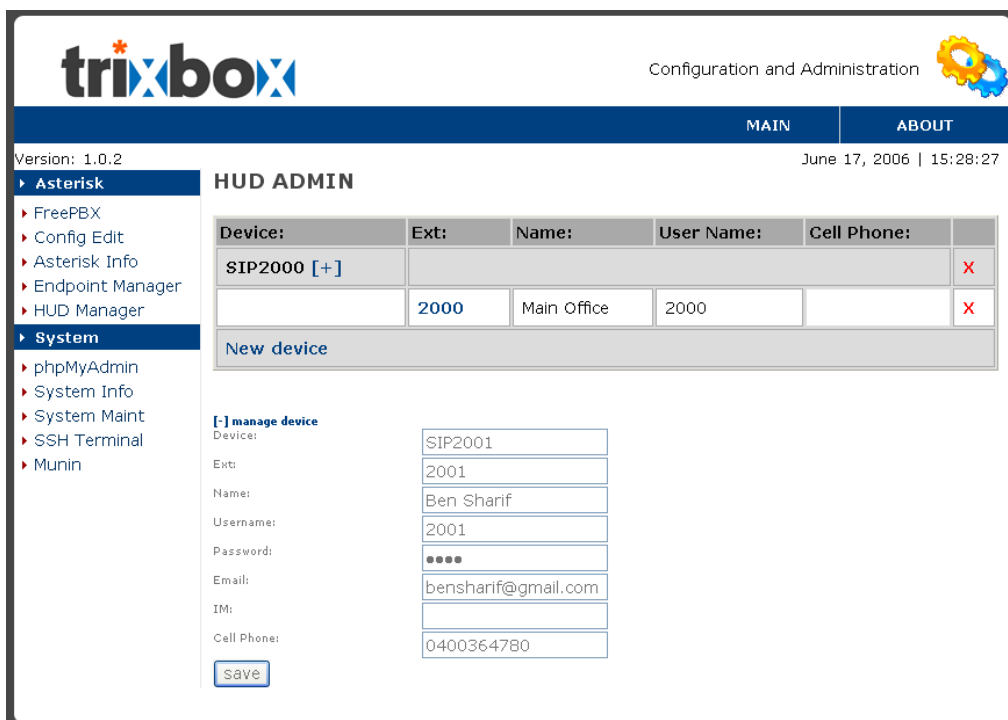
Click on HUD Manager to populate it.

You will be presented with the screen as illustrated below.



The screenshot shows the Trixbox Configuration and Administration interface. The top navigation bar includes 'trixbox' and 'Configuration and Administration'. Below this is a blue bar with 'MAIN' and 'ABOUT' tabs. The version is 1.0.2, and the date/time is June 17, 2006 | 15:09:52. A left sidebar lists navigation options under 'Asterisk' and 'System'. The main content area is titled 'HUD ADMIN' and contains a table with columns: Device, Ext, Name, User Name, and Cell Phone. The table has two rows: one for 'SIP2000 [+]' and another for '2000 Main Office 2000'. A red arrow points to a 'New device' link below the table. A '+ manage device' link is also visible.

To populate the HUD Manager, click on **New device** and the manage device fields will be expanded. This is where you enter the credentials of the extensions that you want to register with HUD Manager.



The screenshot shows the Trixbox Configuration and Administration interface with the 'manage device' form expanded. The top navigation bar and sidebar are the same as in the previous screenshot. The main content area is titled 'HUD ADMIN' and contains a table with columns: Device, Ext, Name, User Name, and Cell Phone. The table has two rows: one for 'SIP2000 [+]' and another for '2000 Main Office 2000'. Below the table, the 'manage device' form is expanded, showing fields for Device, Ext, Name, Username, Password, Email, IM, and Cell Phone. The values entered are: Device: SIP2001, Ext: 2001, Name: Ben Sharif, Username: 2001, Password: 2001, Email: bensharif@gmail.com, and Cell Phone: 0400364780. A 'save' button is located at the bottom of the form.

In the manage device fields I have entered the following:

Device: SIP2001 (*the device for my extension*)

Ext: 2001 (*my extension*)

Name: Ben Sharif (*that's my name if you don't know*)

Username: 2001 (*that's the extension ID*)

Password: 2001 (*I like it real simple*)

The other three fields are not mandatory and you can leave them blank.

Click **save**.

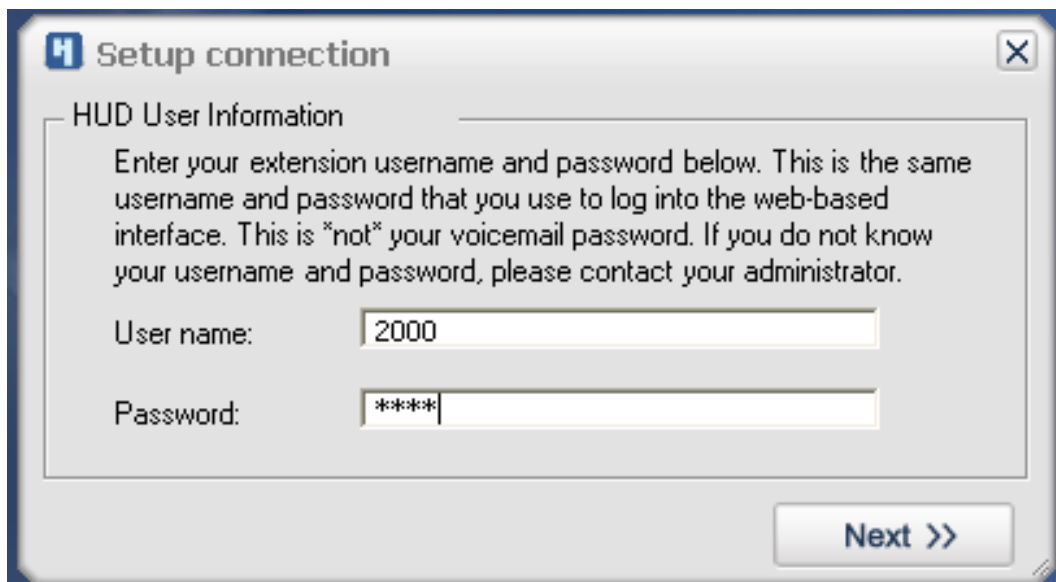
If you have more extensions to register, click on **New device** again and repeat this process as many extensions as you have.

Once this is done, you may proceed to install HUDLite.

14.2.2 Setting Up HUDLite

After downloading, simply click on the EXE file and HudLite will be installed in your Windows XP machine. Select the Typical setup option.

Once installed, run HUDLite you will be prompted to enter your Username and Password:



Enter your username and password as required. I would enter one of my extension number here and the extension password.

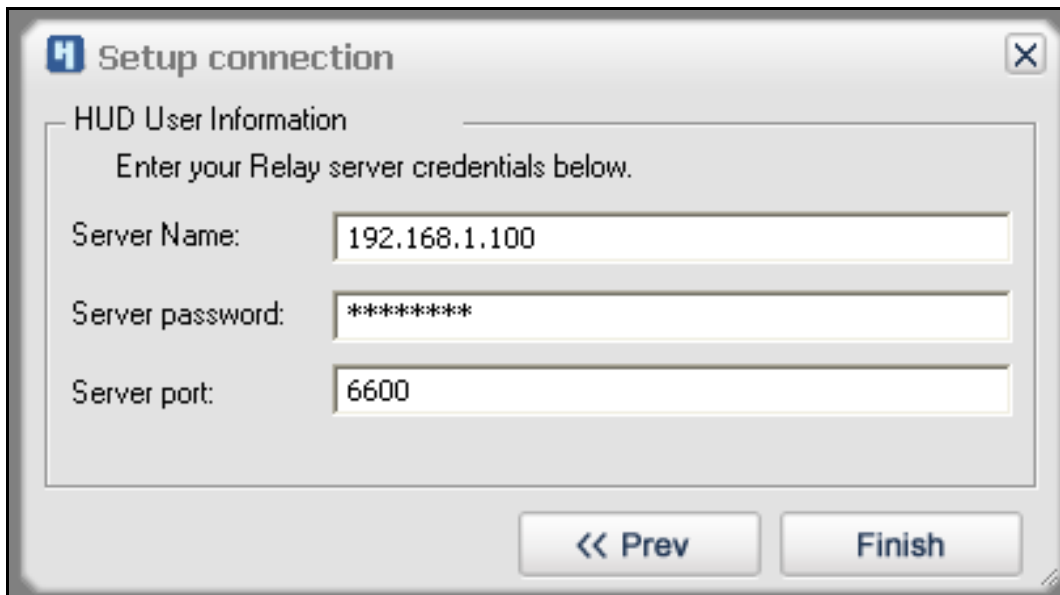
In my case I use my main office extension **2000**.

Username: 2000

Password: 2000

Click Next

Next you will be asked for the server details. This is your TRIXBOX credentials. However in the password field, don't use your server password, instead use HUDLite default password of **password**.



In the Server name I entered my **TRIXBOX IP address: 192.168.1.100**
Server password, I simply use the default password: **password**.
(I strongly urge that you use the default password for this initial installation. You can change it later when you are more comfortable with tinkering around)

Then click Finish.

HUDLite will now start. Initially you will see just a blank hudlite screen like the illustration below, as it will take a few seconds for it to gather some details.



After a few seconds, it will populate all the extensions that you have added in the HUD Manager.



If any of the extension is being called or making a call, you will see it highlighted just like the main office extension on the top left corner (*the main number rang as I was capturing the illustration*).

14.2.3 What if it does not work?

If after following the above instruction, and it does not work, make sure that you have the correct values in the setup above.

Click on **File -> Setting** on the HUDLite screen, expand the Advanced setting and make sure that the fields are correctly set as the set-up instructions above.

Username: **2000** (*you may have different extension*)
Password: **2000** (*you may have different password*)
Server Name: **192.168.1.100** (*your TRIXBOX IP may be different from mine*)
Server Password: **password** (*it must be password*)

Leave the other 3 fields at the default value.

Note: The default password for HUD admin is **password**.

15 CALL PARKING AND TRANSFER

TRIXBOX provides this facility. If you want to allocate other extensions rather than the one pre-configured, simply change the appropriate extension number in the **feature.conf** file.

15.1 CALL TRANSFER - MANAGED

The definition of this facility is found in **features.conf** and it contains the following lines:

features.conf

```
; Sample Parking configuration
;
[general]
parkext => 70 ; What ext. to dial to park
parkpos => 71-79 ; What extensions to park calls on
context => parkedcalls ; Which context parked calls are in
;parkingtime => 60 ; Number of seconds a call can be
; parked for (default is 45 seconds)

[featuremap]
;blindxfer => ## ; Blind Transfer (default is # - change if having
; problem with external DTMF that requires #)

;disconnect => ** ; Disconnect Call
automon => *1 ; One Touch Record
;atxfer => *2 ; Attended Xfer
```

Note: If you are having problem with # when you are calling external number that requires you to input #, you may change the blind transfer key to ## which will solve the problem but you must remember to use ## when you want to do blind transfer.

The companion entry is found in **extensions.conf**

Ensure the following entry exist in the **extensions.conf** under **[from-internal]** context.

extensions.conf

```
[from-internal]
include => parkedcalls
```

If that entry does not exist, add it to **extensions.conf**.

In the above example, the extension to park a call to is ext.70.

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 transfer. 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.

15.1.1 How is it done?

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

1. Dial #70 - the call will be transferred to a park extension and the number will be announced to you. In this case it will be 71. After the announcement, you may hang up and dial the extension of the person you wish to transfer the call to and inform him to dial 71 to accept the call.
2. While all this is happening, the caller will be listening to the music on hold.
3. If for any reason the person refused to take the call, you then hang up and dial 71 to pick up the call back to you and speak to the caller making the usual excuses *“I am afraid Mrs. Smith, your husband is busy with his secretary and his office door is locked from the inside”* ☺ and the next time Mr. Smith will always accept the calls.

15.2 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.

15.3 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.

15.4 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 dialing *8# or *8 from another phone in the call pickup group.

You can also change *8 to something else if you wish.

see <http://www.voip-info.org/wiki/view/Asterisk+config+features.conf>

15.5 FILTER YOUR INCOMING CALLS - ONLY ACCEPT KNOWN CALLS

(Good for filtering Telemarketers and ex-girlfriends)

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.

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

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

I add the following scripts in my ***extensions_custom.conf***

```
[from-internal-custom]

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

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

At the end of the ***extensions_custom.conf***, I added the following 2 blocks of extra codes.

```
; experimental as per above 30-10-2005

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

[parkedcalls]
exten => 70,1,Answer
```

```
exten => 70,2,SetMusicOnHold(default)
exten => 70,3,ParkAndAnnounce(asterisk-friend:/tmp/asterisk-
stranger:vm-isonphone:at-following-
number:PAKED|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.

This will really screws up my Ex ☺ - got my revenge.

16 MEETME – TELECONFERENCE

Meetme Tele-conference can be configured through FreePBX.

Meetme is pretty easy to set up, as there is very little to do. However, to set up Meetme, a Zaptel card will be required.(at least in my case as I don't seem to be able to get it working without a Zaptel card installed). There may be a workaround for it and I hope someone will be able to tell me what it is so it can be included in this guide (in the latest TRIXBOX 1.1, this seems to have been resolved).

The Meetme 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 TRIXBOX, there are 2 **.conf** files that relates to conferencing. They are:

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

meetme.conf

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

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

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

meetme_additional.conf

This is where the extensions for the meeting rooms is 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|
```

16.1 CREATE CONFERENCE

FreePBX allows you to create one or more conference room through the **Setup - > 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:

freePBX™ • Setup • Tools • Reports • Panel • Recordings

Setup

<ul style="list-style-type: none"> Administrators Callback <li style="background-color: #f0f0f0;">Conferences DISA Digital Receptionist Extensions Feature Codes Follow Me Gabcast General Settings Inbound Routes Misc Destinations On Hold Music Outbound Routes PIN Sets Paging and Intercom Queues Ring Groups System Recordings Time Conditions Trunks VoicePulse 	<h2 style="margin: 0;">Conference: 8200</h2> <p style="margin: 0;">Delete Conference 8200</p> <hr/> <p style="margin: 0;">Edit Conference</p> <p style="margin: 0;">conference name: <input type="text" value="General Conference"/></p> <p style="margin: 0;">user PIN: <input type="text" value="1234"/></p> <p style="margin: 0;">admin PIN: <input type="text" value="1235"/></p> <hr/> <p style="margin: 0;">Conference Options</p> <p style="margin: 0;">join message: <input type="text" value="No_Marketing"/></p> <p style="margin: 0;">leader wait: <input type="text" value="Yes"/></p> <p style="margin: 0;">quiet mode: <input type="text" value="No"/></p> <p style="margin: 0;">user count: <input type="text" value="Yes"/></p> <p style="margin: 0;">user join/leave: <input type="text" value="Yes"/></p> <p style="margin: 0;">music on hold: <input type="text" value="Yes"/></p> <p style="margin: 0;">allow menu: <input type="text" value="No"/></p> <p style="text-align: center; margin: 10px 0;"><input type="button" value="Submit Changes"/></p>	<p style="margin: 0;">Add Conference</p> <p style="margin: 0; color: red;">8200:General Conference</p>
--	--	--

freePBX™ Version 2.1.1 on 192.168.1.100

- This conference room number is **8200**
- I call this conference room **General Conference**
- Users are required to enter a PIN number of **1234**
- Admin PIN number is **1235** *for the conference to wait for the leader to conduct the conference.*
- I have a joining message greeting when someone joins the conference.
- I have **Yes** 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 join or leave 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, 8200 and you will be asked to enter the PIN number and to confirm it.

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 8200. You should hear an announcement (if you are listening on the first phone), indicating that someone has joined the conference.

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

16.1.1 /var/spool/asterisk/meetme

It seems that in some installations, the meetme directory may be missing from the /var/spool/asterisk.

If this is the case, It may be necessary for this directory to be created otherwise if you have the **user join/leave** feature enabled, meetme will not function correctly.

If you have the need to create this directory, do the following at root command prompt:

```
mkdir /var/spool/asterisk/meetme
```

then change the ownership of this directory

```
chown asterisk.asterisk /var/spool/asterisk/meetme
```

next you need to change the permission,

```
chmod 775 /var/spool/asterisk/meetme
```

16.2 INVALID CONFERENCE NUMBER

If you get this error, chances are, it is caused by not having a zaptel timing source. This often happen when you have upgraded your kernel, but you haven't recompiled your zaptel to suit.

Here is the quick fix compliment of Rob, our freePBX guru.

Log in to your Asterisk box as root, and at the command prompt, do the following:

```
cd /usr/src/zaptel
make install
/etc/init.d/zaptel stop
/etc/init.d/zaptel start
```

17 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.

You must run `setup-mail` from the command line initially for Voice Mail and Notification to work in TRIXBOX

To enable voicemail on an extension simply "enable" it when you create the extensions from the AMP GUI. If you require email notification of your voicemail, you may enter your email address in the email address field.

17.1 ENABLING EMAIL NOTIFICATION

Asterisk can send email to you informing you that you have voicemail waiting in your mail box. In fact it can even send you the voicemail itself in the form of a .wav file if you so desire.

17.1.1 Installing sendmail

This is not always the case in every installation. If this happens to you, you may need to re-install sendmail.

I have been having some problem to get mail notification to be sent to my email address. I later discovered that sendmail need to be installed for this function to work (*thanks Thunderbird1 for pointing this out to me*). Sadly, this is not mentioned anywhere and has caused me major grief.

To install sendmail with yum, you can do it on the Asterisk box itself or connect through SSH (I use putty to connect), by **logging in as root**.

Do these commands at the command prompt:

```
yum install sendmail-cf
```

This installs the necessary files for configuration of the sendmail client. Select "y" for yes when prompted and sendmail will be installed.

Next you will need to do a little editing. I use nano to edit, thus:

```
nano /etc/mail/sendmail.mc
```

Locate the following line:

```
dn1 define(`SMART_HOST',`smtp.yourprovider')
```

and edit it as per the example below

```
define(`SMART_HOST',`smtp.yourprovider')
```

Replace 'smtp.yourprovider' with your real smtp provider e.g. 'smtp.ozemail.com.au'

Then press CTRL-X, and type in Y, hit enter. You'll be asked for the file name you want "File Name To Write: /etc/mail/sendmail.mc", and hit enter.

Next, you will need to rebuild your /etc/mail/sendmail.cf file.

At the command prompt, do the following:

```
make -C /etc/mail
```

Once done, restart sendmail thus;

```
/etc/init.d/sendmail restart
```

You should now be able to have your voice mails sent to you via email.

The above instruction is also mentioned in this link:

http://sourceforge.net/forum/message.php?msg_id=3387743

and http://sourceforge.net/forum/message.php?msg_id=3432175

17.1.2 /etc/hosts

There is a couple of things that you will need to do otherwise your email, although sent to the email address, your email host may not accept the email if you do not have a valid DNS.

By default, your host will look like this:

```
# Do not remove the following line, or various programs
# that require network functionality will fail.
127.0.0.1      localhost
127.0.0.1      asterisk1.local
```

Your mail server will not accept this as a valid DNS. Therefore you will need to use your valid Domain Name e.g. **pbx.myasterisk.com**. If you are like me, without a valid Domain Name, you should obtain one through DynDNS.

Using Nano, you will need to edit your **/etc/hosts** file from command line to this:

```
# Do not remove the following line, or various programs
# that require network functionality will fail.
127.0.0.1      localhost
127.0.0.1      pbx.myasterisk.com asterisk1.local
```

You may use an editor (such as nano) to edit **/etc/hosts** file through CLI.

*If you are using **SipBroker** or **Pennytel**, its best not to make this change as describe above, as SipBroker and Pennytel will resolve your DNS name as the local host address of 127.0.0.1 as defined in the hosts file and not your actual external IP address. To overcome this I substitute my DNS hostname with another valid DNS name such as **yahoo.com** (as I have an account with Yahoo – not sure about the legality side of this though).*

Refer to [Notes](#) for a kludge fix if you are experiencing difficulties.

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.

```
; 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 entry in red to reflect your requirement e.g you may change **vm@asterisk** to **My Super PBX** and change **Voicemail System** to read **My Cool System ☺**

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: My Cool System **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.myasterisk.com:8080/cgi-bin/vmail.cgi?action=login&mailbox=201](http://pbx.myasterisk.com:8080/cgi-bin/vmail.cgi?action=login&mailbox=201)
to check your voicemail with a web browser.

17.2 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 TRIXBOX system.

This is how I do it (you may have been told of other different methods).

17.2.1 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.

17.2.2 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.

17.3 HOW TO CHANGE VOICE MAIL GREETINGS

The default Voicemail greeting is fine but it is rather bland and quite boring. No doubt some user would like to change the standard greeting to something more personal and meaningful.

Note: *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 wife’s phone number and I promise that I will call her when you are not home”*
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 folks ☺

18 FAX TO E-MAIL

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 receiving Faxes is one of the available functions in TRIXBOX, there have been reports of various degree of success. This is very much dependant on the type of equipment used such as PC, processor speed, line quality etc.

In the earlier incarnation of TRIXBOX (it was AAH for those old enough to remember), for the fax facility to work reliably, you will also need [Newman Telecom's NVfaxdetect](#). This can be downloaded FREE. This may require some modification and recompiling.

SpanDSP allows Asterisk to receive faxes by converting the received data to .pdf format. This service must be installed. If you want to have this function, you will need to install the support for encoding incoming faxes as PDF. This support is not pre-installed with TRIXBOX although the necessary files are there for you to manually install.

18.1 INSTALL-PDF

To install this support, you need to log in to Centos either locally or remotely.

At the command prompt type:

```
install-pdf
```

This will install support for encoding incoming faxes as PDFs. Once this is done, your chances of getting fax through TRIXBOX is increased dramatically.

The fax received is converted into a **.pdf** file in **/var/spool/asterisk/fax** and emailed to you.

18.2 INSTALLING FAX FOR TRIXBOX

The developer of Asterisk have included NVfaxdetect in Asterisk starting versions 1.2.2 which allows incoming faxes to be received over SIP, IAX & ZAP channel, unlike previous versions where it has to be installed separately. However, installing fax is not without its woes.

The Whirlpool Forum community have been working on solutions to get this facility to play nicely and a solution was published by one of the participants which have been reproduced below:

You also need to run install-pdf, from the console, if you require faxes to be emailed as pdf files.

The following is a method found to be suitable for TRIXBOX with FreePBX.

Note:

Remember to **run install-pdf** to enable Trixbox to convert your faxes to pdf. Also, to reliably receive incoming PSTN faxes, you will probably also need to increase `rxgain=` in `zapata.conf` from the default of 0.0 to something higher (I have mine set at 7.0) otherwise you will probably end up with a blank pdf being emailed. You need to stop & restart Trixbox after making any changes to `zapata.conf`:

```
amportal stop
amportal start
```

It is found that incoming PSTN faxes are received & emailed fine but the resulting pdf formatting left too much white space at the top & bottom of my first page with subsequent pages missing the text header as well as the last few lines. Incoming SIP faxes are received ok but are not converted to pdf nor emailed.

To correct the pdf formatting issue you need to modify the tiff to pdf conversion line in the **fax-process.pl** script (by removing the page width & length parameters) using a text editor like nano:

```
nano /var/lib/asterisk/bin/fax-process.pl
```

Scroll down to find & change the line from:

```
open PDF, "tiff2pdf -w8.5 -l11 $file|";
```

to:

```
open PDF, "tiff2pdf $file|";
```

Save & exit.

To receive incoming SIP faxes, modify the ext-fax context in **extensions.conf** from the existing script below:

```
[ext-fax]
exten => s,1,Answer
exten => s,2,Goto(in_fax,1)
exten => in_fax,1,StopPlayTones
exten => in_fax,2,GotoIf(${FAX_RX}" = "system?3:analog_fax,1)
exten => in_fax,3,Macro(faxreceive)
exten => in_fax,4,system(/var/lib/asterisk/bin/fax-process.pl --to
${EMAILADDR} --from ${FAX_RX_FROM} --subject "Fax from
${CALLERID(number)} ${CALLERID(name)}" --attachment
${CALLERID(number)}.pdf --type application/pdf --file ${FAXFILE});
exten => in_fax,5,Hangup
exten => analog_fax,1,GotoIf(${FAX_RX}" = "disabled?4:2) ;if fax
is disabled, just hang up
exten => analog_fax,2,Set(DIAL=${DB(DEVICE/${FAX_RX}/dial)});
exten => analog_fax,3,Dial(${DIAL},20,d)
exten => analog_fax,4,Hangup
;exten => out_fax,1,wait(7)
exten => out_fax,1,txfax(${TXFAX_NAME}|caller)
exten => out_fax,2,Hangup
exten => h,1,Hangup()
```


to the modified one below:

```
[ext-fax]
exten => s,1,Answer
exten => s,2,Goto(in_fax,1)
exten => in_fax,1,StopPlayTones
exten => in_fax,2,GotoIf(${FAX_RX}" = "system?3:analog_fax,1)
exten => in_fax,3,Macro(faxreceive)
exten => in_fax,4,Hangup
exten => analog_fax,1,GotoIf(${FAX_RX}" = "disabled?4:2) ;if fax
is disabled, just hang up
exten => analog_fax,2,Set(DIAL=${DB(DEVICE/${FAX_RX}/dial)});
exten => analog_fax,3,Dial(${DIAL},20,d)
exten => analog_fax,4,Hangup
;exten => out_fax,1,wait(7)
exten => out_fax,1,txfax(${TXFAX_NAME}|caller)
exten => out_fax,2,Hangup
exten => h,1,system(/var/lib/asterisk/bin/fax-process.pl --to
${EMAILADDR} --from ${FAX_RX FROM} --subject "Fax from
${CALLERID(number)} ${CALLERID(name)}" --attachment
${CALLERID(number)}.pdf --type application/pdf --file ${FAXFILE});
exten => h,2,Hangup()
```

Highlighted, in **red**, are the changes made from the original and the **bold red** highlight indicates that the line should be one continuous line.

Next, log on to FreePBX.

- In freePBX go to General Settings -> Fax Machine & set to *system* & enter your email address.
- Go to Inbound Routes & create an incoming route if you haven't already done so.
- Set Fax Extension to *freePBX default*.
- Set Immediate Answer to *Yes* & Pause after answer to *4*

Testing

Testing consisted of connecting a fax machine to a SPA-2000 configured as ext 200 & then dialling 7777 to simulate an incoming call. To monitor the progress of the call, open an Asterisk console by typing *asterisk -rvvvvv* at the CLI. To exit, type *exit*.

For testing incoming PSTN faxes, send a normal fax to **1300 368 999** (Telstra's test fax service) & within a short period of time you will get a response fax. If that call fails, the service will re-try after 3 mins. Depending on the quality of your line, you will probably need to re-adjust the *rxgain=* setting in *zapata.conf*. Using a TDM400P I had to set my *rxgain* to 7.0 to receive PSTN faxes. At lower settings the faxes failed - YMMV ☺

(Source Reference: *ThunderBird1*)

19 WEATHER FORECAST

The existing weather forecast information that comes with TRIXBOX is configured for New York. However, Asterisk 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 TRIXBOX **extensions_trixbox.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.

19.1 CONFIGURE WEATHER REPORT – ON DEMAND.

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:

```
/usr/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.

19.1.1 Create Extension codes.

The next step is to include the following in the **extensions_trixbox.conf** through the **Config Edit** option of TRIXBOX Maintenance 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.

19.2 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 for the moment. 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.

19.2.1 Create Extension codes.

Next include the following in **extensions_trixbox.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.

19.3 NOW PUT IT IN YOUR DIGITAL RECEPTIONIST

If you want the 2 procedures to be incorporated in your Digital receptionist, simply duplicate them with some minor changes. You will want the digital receptionist to know which whether 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

```

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 App

20 REMOTE MANAGEMENT

Some may want to manage their Asterisk 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 80 on your router to point to the local IP address of your Asterisk box.

To log on to your Asterisk box remotely using your browser, use
`http://your_DynDNS_name:80` or
`http://your_public_ip_address:80`

20.1 HTTPD.CONF

If you are using Optus Cable, you may not be able to use port 80. However this can be overcome by changing the port to 8080. To do this, you will be required to change one setting in the **httpd.conf** file which resides in the **/etc/httpd/conf** directory.

If you are not a Linux fan, I suggest you use either webmin or putty to access the file. I use Putty to log on remotely to the console and edit the file with Nano using the following command..

```
nano /etc/httpd/conf/httpd.conf
```

In the file, look for the following entry (marked in red) - below :

```
# Change this to Listen on specific IP addresses as shown below to
# prevent Apache from glomming onto all bound IP addresses (0.0.0.0)
# e.g. "Listen 12.34.56.78:80"
#
# To allow connections to IPv6 addresses add "Listen [::]:80"
#
Listen 0.0.0.0:80    ← change this from 0.0.0.0:80 to read 0.0.0.0:8080
#
```

After doing the above, reboot Asterisk.

Don't panic if you can no longer access AMP through your web browser. You simply append :8080 after the address e.g. **192.168.1.101:8080**

21 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*

21.1 CREATE A NEW EXTENSION

Create a new extension in Asterisk – 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.*

IAX2 Extension: 205

[Delete Extension 205](#)

Edit Extension

Display Name:

Extension Options

Outbound CID:

Record Incoming:

Record Outgoing:

Device Options

secret	<input type="text" value="205"/>
nottransfer	<input type="text" value="yes"/>
context	<input type="text" value="from-internal"/>
host	<input type="text" value="dynamic"/>
type	<input type="text" value="friend"/>
port	<input type="text" value="4569"/>
qualify	<input type="text" value="no"/>
disallow	<input type="text"/>
allow	<input type="text"/>
dial	<input type="text" value="IAX2/205"/>
accountcode	<input type="text"/>

Voicemail & Directory:

voicemail password:

email address:

pager email address:

email attachment: yes no

Play CID: yes no

Play Envelope: yes no

Delete Vmail: yes no

vm options:

vm context:

AAH 2.x IAX Extension Screen
It differs from AAH 1.x but essentially
it serves the same purpose

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" instead of "never" as illustrated in the screen below - see red arrow.

SIP Extension: 205

[Delete Extension 205](#)

Edit Extension

Display Name:

Extension Options

Outbound CID:

Record Incoming: ▼

Record Outgoing: ▼

Device Options

secret	<input type="text" value="205"/>
dtmfmode	<input type="text" value="rfc2833"/>
canreinvite	<input type="text" value="no"/>
context	<input type="text" value="from-internal"/>
host	<input type="text" value="dynamic"/>
type	<input type="text" value="friend"/>

nat	<input type="text" value="yes"/>
port	<input type="text" value="5060"/>
qualify	<input type="text" value="no"/>
callgroup	<input type="text"/>
pickupgroup	<input type="text"/>
disallow	<input type="text"/>
allow	<input type="text"/>
dial	<input type="text" value="SIP/205"/>
accountcode	<input type="text"/>

Voicemail & Directory: ▼

voicemail password:

email address:

pager_email address:

email attachment: yes no

Play CID: yes no

Play Envelope: yes no

Delete Vmail: yes no

vm options:

vm context:

Once you have done this, you will need to ensure a couple of things in your .conf files are set correctly.

21.2 IAX.CONF

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

iax.conf

```
[general]
bindport = 4569 ; Port to bind to (IAX is 4569. Forward this port to your Asterisk box)
externip = <your fixed external IP> or
externhost = <myasterisk.freedns.com>
localnet = 192.168.1.0/255.255.255.0
bindaddr = 0.0.0.0 ; Address to bind to (all addresses on machine)
delayreject = yes
disallow=all
allow=g729
allow=ilbc
allow=gsm
allow=g723
allow=alaw
allow=ulaw

jitterbuffer = yes
mailboxdetail = yes

#include iax_additional.conf
#include iax_custom.conf
```


21.3 SIP_NAT.CONF

The only other .conf file that requires attention is the **sip_nat.conf** in case you want to use SIP. If you do, ensure that you have forward ports 5004 – 5082 as well. My file is as per the example below.

sip_nat.conf

```
nat=yes
externip=<your fixed external IP> or
externhost=< myasterisk.freedns.com> (see note)
localnet=192.168.1.0/255.255.255.0
externrefresh=10
```

Those are the only 2 .conf files that require changing if needed.

21.4 THE CORRECT SOFTPHONE FOR IAX

I use a 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.

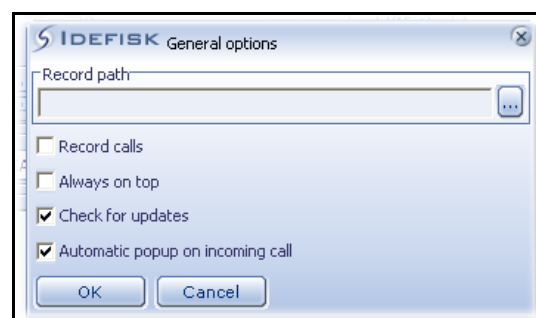
Click on the idefisk.exe and just configure it like the example below.



Once the softphone is active, to start configuring, simply place your cursor anywhere in the middle of the image and right click.

The configuration option window will be displayed.

21.4.1 First the general options



In the general options, checked the Check for update and Automatic popup on incoming call.

21.4.2 Accounts options

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

The screenshot shows the 'IDEFISK Account options' dialog box. On the left, there is a list of accounts with 'Ben Sharif IAX' selected. On the right, there are several input fields: 'Host' (myasterisk.freedns.com), 'User name' (205), 'Password' (isndhrtt), 'Caller ID' (Ben on IAX), and 'Number' (Remote). Below these fields is a checkbox for 'Register on startup' which is unchecked. At the bottom of the dialog, there are four buttons: 'New', 'Update', 'Set Default', and 'Delete'. An 'OK' button is located at the bottom left, and the text 'Default account : Ben Sharif IAX' is displayed at the bottom center.

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.

22 TOOLS

The following are tools that I use regularly to maintain my Asterisk systems.

22.1 WEBMIN

Webmin is an invaluable web based tool for managing a Linux box. Webmin makes it easy to configure applications like SMTP mail, editing files etc (see [Note](#)).

Those who want to use Web Admin to maintain the Asterisk System may download Webmin from [here](#) or from CLI, do the following:

```
wget superb-east.dl.sourceforge.net/sourceforge/webadmin/webmin-1.270-1.noarch.rpm
```

Install it with the following command through CLI:

```
rpm -Uvh webmin-1.270-1.noarch.rpm
```

Or be totally lazy like me and do the whole lot in a one liner;

```
rpm -Uvh http://superb-east.dl.sourceforge.net/sourceforge/webadmin/webmin-1.270-1.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

You may connect to Webmin remotely through your browser using the following address **`http://<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 **`http://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 **`http://192.168.0.101:10000`** ☺

(Note, 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
;
rtptest=10000 < - - change this to 10002(totally unnecessary.. ☺)
rtptest=20000
```

22.1.1 To update WebMin

Anytime you want to update Webmin, simply do the following.

Log on to your Asterisk box (SSH or at the console).

At the command prompt, issue the following command:

```
yum -y install webmin
```

Sorted ☺

22.2 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>

22.3 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>

23 STUN SERVERS

STUN (Simple Traversal of UDP through NATs (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.

Below is a list of public STUN 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

24 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.

24.1 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 TRIXBOX Config Edit (the easiest way!)

TRIXBOX -> Config Edit -> 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.

24.2 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.

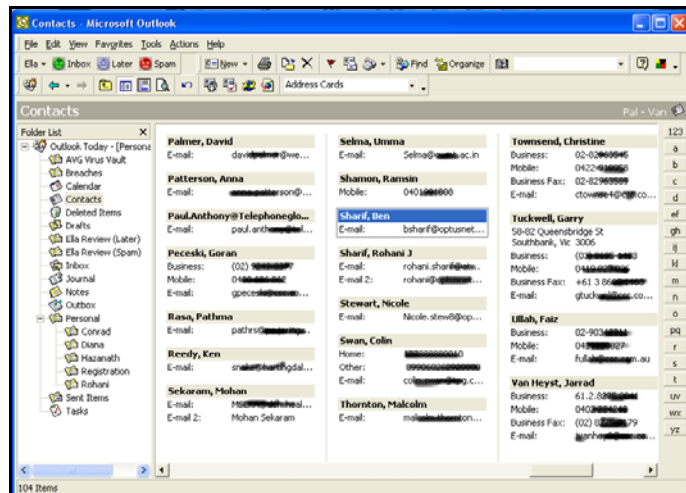
24.3 CONFIGURE OUTLOOK ADDRESS BOOK

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).

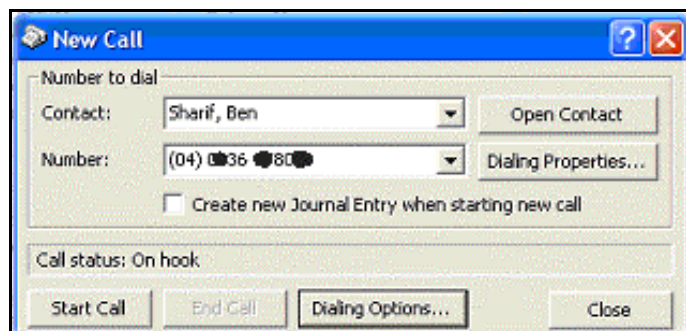
Click on the phone icon.



Click on the Telephone icon.

The New Call dialog box will be displayed:

Select Dialing Options.



The Dialing Option box will be displayed.

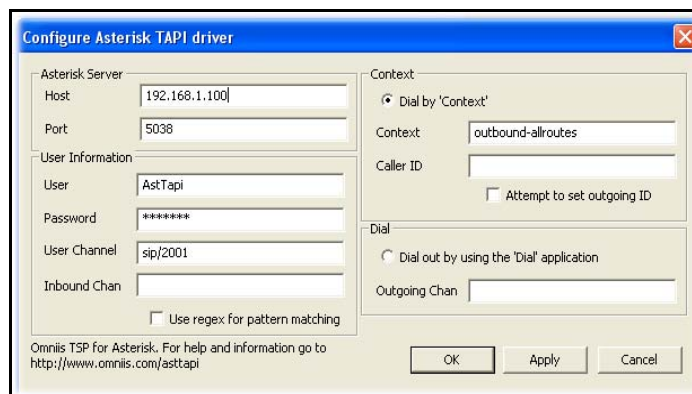
In the "connect using line" drop down box, select Asterisk.

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 ☺

25 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 TRIXBOX's way.

25.1 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 **TRIXBOX -> Config Edit**

```
; #####
; Speed Dial to some predefined numbers (added by Ben 2/12/2005)
; #####

exten => 236,1,playback(pls-wait-connect-call)
exten => 236,2,Dial(SIP/6140036xxxx@pennytelsip,60,m) ; Ben's Mobile
exten => 236,3,Hangup

exten => 764,1,playback(pls-wait-connect-call)
exten => 764,2,Dial(SIP/6141242xxxx@pennytelsip,60,m) ; Rohani's Mobile
exten => 764,3,Hangup

exten => 667,1,playback(pls-wait-connect-call)
exten => 667,2,Dial(SIP/6140470xxxx@pennytelsip,60,m) ; Nor's Mobile
exten => 667,3,Hangup

exten => 767,1,playback(pls-wait-connect-call)
exten => 767,2,Dial(SIP/6140512xxxx@pennytelsip,60,m) ; Ros's Mobile
exten => 767,3,Hangup

exten => 232,1,playback(pls-wait-connect-call)
exten => 232,2,Dial(SIP/6140936xxxx@pennytelsip,60,m) ; Adam's Mobile
exten => 232,3,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.

25.2 TRIXBOX'S METHOD

There is however, a simpler way (if you are too lazy to write extra codes – like me☺)

TRIXBOX has included a built-in speed dialling utility amongst the goodies that comes with it. Numbers 300 to 399 has been reserved for this purpose.

To add a number to your speed dial list, you simply pick up one of your extensions and punch in 300**3XX**61212345680 (assuming 61212345680 is the number you want to save in your speed dial list) where 3XX is the speed dial code for 61212345680 (substitute 3XX for the position e.g. 301).

After doing that, Asterisk will confirm the code and the number that you have just registered. If you want to make correction to or change the telephone number that you have entered simply repeat the process.

In the above example, every time you dial 301 Asterisk will dial out the number associated with that code. In this case, it is 61212345680.

Ensure that you enter the number to be called in the same format that is supported by your Asterisk dialplan.

To look up speed dial numbers, simply dial * followed by the 3-digit speed dial code, e.g. *301. This will tell you the number stored in speed dial position 301.

26 HOW TO USE WINDOW MESSENGER® 5.X

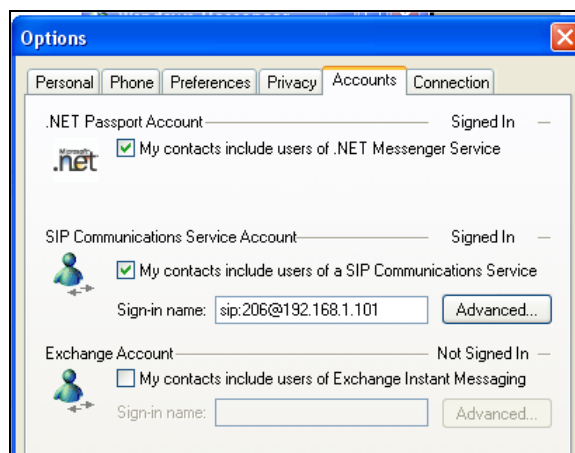
Note: Windows Messenger is **NOT** MSN Messenger. You may need to download Windows Messenger from Microsoft.

Set up Windows Messenger 5.x as a SIP client on your PC:

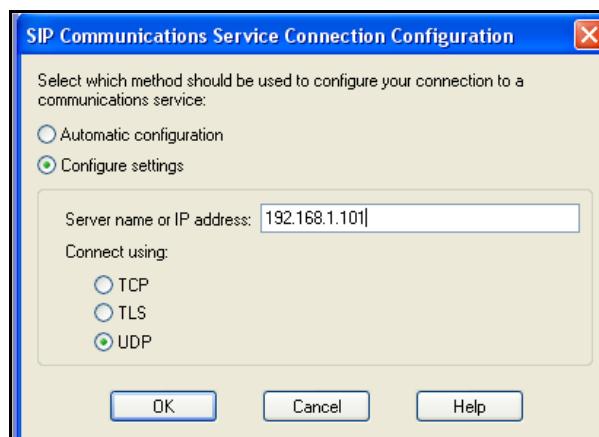
1. Start Windows Messenger from your PC.

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



1. 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.

Note: *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.

freePBX -> Tools -> Asterisk Information. – *this is available as an addon (see NOTES on [Adding the missing Maintenance Modules](#))*

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.

27 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 Asterisk 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).

27.1 METHOD 1 - WITH THE PEER ASTERISK BOX AS EXTENSIONS

For the purpose of registering the peers to each other, I created 1 extension on each box eg: 90000 on System 1 and 91000 on System 2– using extension numbers that I am not likely to use as local extensions (while some users have had success using common extension, but I prefer 2 separate extensions as I have them working). For simplicity, I gave a common password xxxyyy to both boxes. Avoid using extension starting with 8 as it may clash with conferencing.

	System 1 <i>def.com.au</i>	System 2 <i>xyz.com.au</i>
<u>IAX Trunk</u>		
Outgoing Dial Rules:	XX.	XX.
Trunk Name	Parramatta	MainOffice
Peer Details	host=xyz.com.au (or System 2 IP) secret=xxxxyy type=peer username=91000	host=def.com.au (or System 1 IP) secret=xxxxyy type=peer username=90000
User Context	Leave blank	Leave blank
User Details	Leave blank	Leave blank
Register String	<i>80000:xxxxyy@xyz.com.au</i>	<i>90000:xxxxyy@def.com.au</i>

Note: Registration isn't really necessary. It will still work without it unless you use Dynamic IP.

	System 1	System 2
<u>Extensions</u>		
Phone Protocol	IAX	IAX
Extension Number	90000	91000
Extension Password	xxxxyy	xxxxyy
Fullname	Parramatta	Main Office
Voicemail & Directory	Disabled	Disabled

	System 1	System 2
<u>Outbound Routing</u>		
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 City 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.

The above example assumes 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, than you will have no need to worry about DynDns and what not.

Note: *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.*

27.2 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.

Note: *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.

(Note: A little tutorial on DUNDi can be found [here](#)).

Rather than being verbose, I will illustrate this method using tables as follows;

	System 1 <i>def.com.au</i>	System 2 <i>xyz.com.au</i>
<u>IAX2 Trunk</u>		
Outgoing Dial Rules:	6XXX	XX.
Trunk Name	InterOffice	InterOffice
Peer Details	host=xyz.com.au (<i>or System 2 IP</i>) Qualify=yes type=peer	host=def.com.au (<i>or System 1 IP</i>) Qualify=yes type=peer
User Context	InterOffice-In	InterOffice-In
User Details	context=from-internal host=xyz.com.au (<i>or IP</i>) type=user	context=from-internal host=def.com.au (<i>or IP</i>) type=user

	System 1	System 2
<u>Outbound Routing</u>		
Route Name	InterOffice	InterOffice
Route Password	Leave Blank	Leave Blank
Dial Patterns	6XXX (<i>6001 to 6009 are Parramatta Office extensions</i>)	XX. (<i>Apart from Local extensions, all others go via City Office</i>)
Trunk Sequence	IAX2/InterOffice	IAX2/InterOffice

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). 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 advise to you is to hire a VOIP consultant.

28 CUSTOMISED VOICE

The default pre-recorded voice prompts and announcements in TRIXBOX 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.

28.1 CUSTOMISING VOICE PROMPTS

To satisfy this requirement, custom voice prompts can be recorded individually by using the **System Recording** facility of freePBX as covered in the chapter referring to [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>

28.2 INSTALLING OTHER LANGUAGES

If you wish to use a different language other than 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/library/asterisk/sounds** directory, **e.g. /var/library/asterisk/sounds/au** and send all your prompts sound files there. Also create the language directories in the digits, letters etc directories, and send all the 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 TRIXBOX, 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).

29 MOH (MUSIC ON HOLD)

TRIXBOX supports native MP3. To change or add to your music on hold collections, simply upload all your MP3 to TRIXBOX through the **Onhold Music** screen of freePBX.

You may also need to add the following, under **[channels]** in your **zapata.conf** file:

```
musiconhold=default
```

If using TRIXBOX, pretty much any mp3 will work. 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.

You will now have to get used to the newer approach (it no longer uses the `[classes]` tag):

You need to change the default directory in **musiconhold.conf** (see below) from `moh-native`, to `mohmp3` (see sample below), otherwise your music when doing web upload will be in a wrong directory.

musiconhold.conf

```
[moh_files]
;default => /var/lib/asterisk/moh-native,r
default => /var/lib/asterisk/mohmp3,r
#include musiconhold_additional.conf
; class definitions For A2Billing
acc_1 => mp3:/var/lib/asterisk/mohmp3/acc_1
acc_2 => mp3:/var/lib/asterisk/mohmp3/acc_2
acc_3 => mp3:/var/lib/asterisk/mohmp3/acc_3
acc_4 => mp3:/var/lib/asterisk/mohmp3/acc_4
acc_5 => mp3:/var/lib/asterisk/mohmp3/acc_5
acc_6 => mp3:/var/lib/asterisk/mohmp3/acc_6
acc_7 => mp3:/var/lib/asterisk/mohmp3/acc_7
acc_8 => mp3:/var/lib/asterisk/mohmp3/acc_8
acc_9 => mp3:/var/lib/asterisk/mohmp3/acc_9
acc_10 => mp3:/var/lib/asterisk/mohmp3/acc_10
```

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
```

29.1 STREAMING MUSIC ON HOLD

Streaming music on hold is not directly supported using the GUI of TRIXBOX or freePBX, however if you are prepared to do a little editing of the musiconhold.conf file, you will be able to get it working somehow.

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.

First I have to download and install mpg123 as it is not included in TRIXBOX implementation of Asterisk. To get mpg123 I have to log on as root and at the command prompt I did the following:

```
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.

```
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.

```
cd stream
touch stream.mp3
```

Now I have to do some editing of my **musiconhold.conf** file.

Open the musiconhold.conf file using the Config edit option on your TRIXBOX and add the following lines on top of the file and leave everything else as they are.

```
[classes]
default => quietmp3:/var/lib/asterisk/mohmp3/stream,http://radioio.sc.llnwd.net:8075/
```

The **http://radioio.sc.llnwd.net:8075/** is where I get my streaming music from.

This is how my **musiconhold.conf** looks like.

```
[classes]
default => quietmp3:/var/lib/asterisk/mohmp3/stream,http://radioio.sc.llnwd.net:8075/
;
; Leave all the default stuff underneath unchanged
;
[moh_files]
default => /var/lib/asterisk/mohmp3,r
#include musiconhold_additional.conf
; class definition for Streaming MOH
;stream => quietmp3:/var/lib/asterisk/mohmp3/stream
; class definitions For A2Billing
acc_1 => mp3:/var/lib/asterisk/mohmp3/acc_1
acc_2 => mp3:/var/lib/asterisk/mohmp3/acc_2
acc_3 => mp3:/var/lib/asterisk/mohmp3/acc_3
acc_4 => mp3:/var/lib/asterisk/mohmp3/acc_4
acc_5 => mp3:/var/lib/asterisk/mohmp3/acc_5
acc_6 => mp3:/var/lib/asterisk/mohmp3/acc_6
acc_7 => mp3:/var/lib/asterisk/mohmp3/acc_7
acc_8 => mp3:/var/lib/asterisk/mohmp3/acc_8
acc_9 => mp3:/var/lib/asterisk/mohmp3/acc_9
acc_10 => mp3:/var/lib/asterisk/mohmp3/acc_10
```

The lines in BOLD are the lines that was added to the musiconhold.conf file.

After having done that you may reboot or restart asterisk.

```
amportal restart
```

The above is not necessarily being the best way to approach it but that is the only way that I know of for now. Those with alternative way, I would like to hear from them.

30 BACKUP AND RESTORE

Backup and Restore are two of the mandatory functions of any application. FreePBX is no exception. With this in mind, the developer of freePBX has incorporated this facility in freePBX under the Tools section.

Provision for flexibilities was built in this tool. You can schedule your backup, Daily, Weekly, Monthly, Ad-Hoc and even hourly if you wish to. Therefore there is no excuse for not making a regular backup.

30.1 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.

If this is the first time you schedule a backup, it is advisable to do the backup right away by selecting **Now** from the Run Backup drop down selection box. After this is done, you may start scheduling your regular backup by selecting the appropriate option presented to you.

I have chosen to schedule a **Monthly (On the 1st)** backup as per the illustration below:

- In the Schedule Name, I have given it a name Monthly (to signify that it is a monthly job)
- I have also selected all the items to be backed up – VoiceMail, System Recordings, System Configurations, CDR and Operator Panel.
- Next from the Run Backup drop down choice, I have chosen Monthly (on the 1st)

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***.

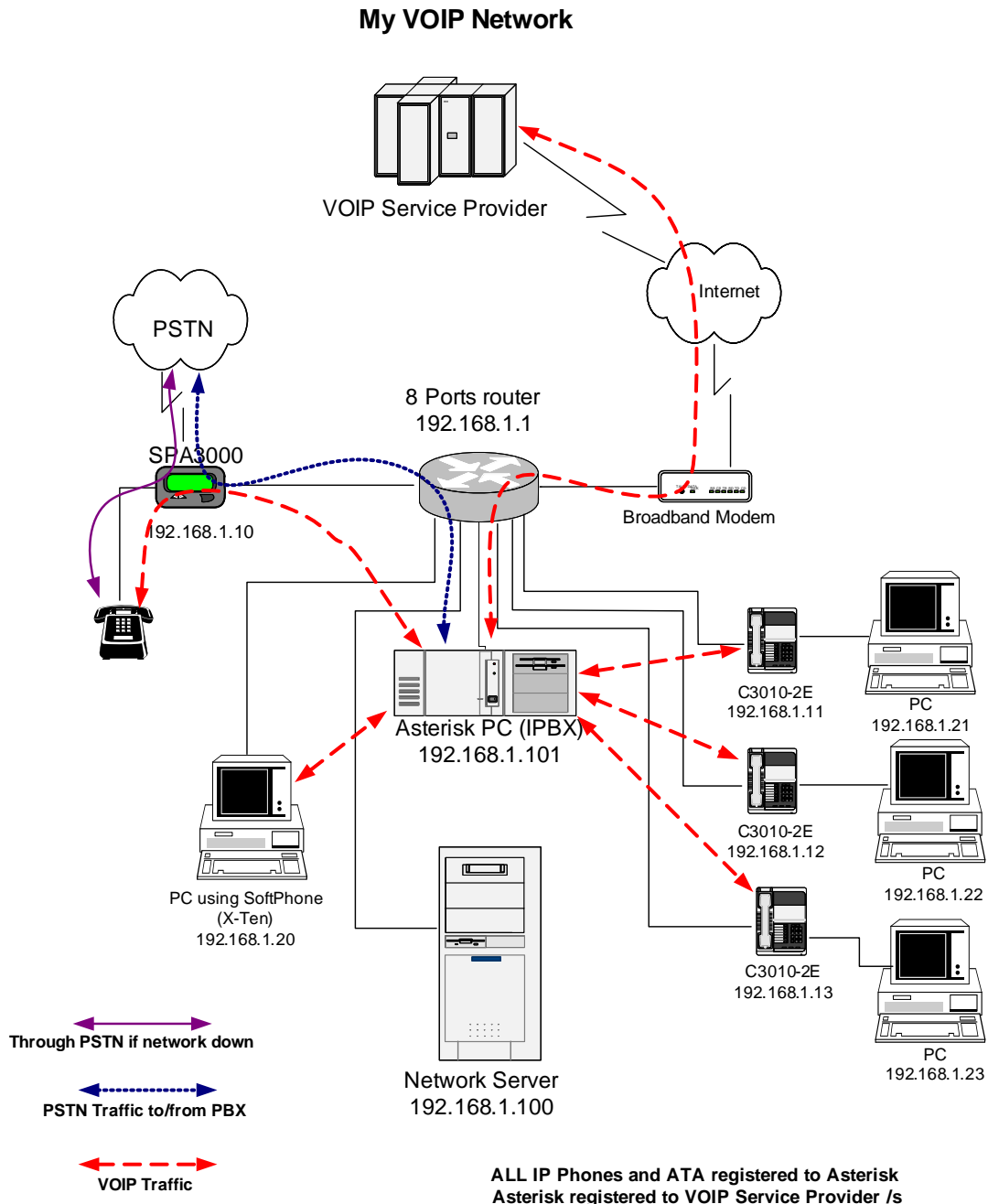
30.2 TO RESTORE A BACKUP

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 and the rest are all a matter of choosing what it is that you want to restore.

31 MY TRIXBOX® 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.



32 PUBLICATIONS AND REFERENCES

32.1 DUNDI – A GUIDE TO A SIMPLE SETUP

You will find a guide to a simple DUNDi setup at the GoldfishNet Blog:

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

32.2 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.

32.3 OTHER ASTERISK FORUMS, TUTORIALS AND WIKIS

<http://asteriskathome.sourceforge.net/>

<http://www.voip-info.org/wiki/view/Asterisk%40home+Handbook+Wiki>

<http://asteriskathome.sourceforge.net/faq.html>

http://sourceforge.net/forum/?group_id=123387

<http://amp.coalescentsystems.ca/>

http://sourceforge.net/docman/?group_id=121515

The following are two tutorial sites that I recommend everyone who is serious about Asterisk to visit.

<http://mundy.org/blog/index.php?p=81>

<http://www.asteriskguru.com/tutorials/>

General VOIP related:

<http://www.freewebs.com/voipformula/VoIP-HOWTO.html>

32.4 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

33 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-plk.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.sjllabs.com/usbphone/SJphoneDriverATCOM-AU-100.exe>

34 BUGS REPORTS

Software is not software if it is release without any bug. It just goes against nature's logic, not to mention our Mr. Murphy ☺

34.1 MISSING ZAPTEL SUPPORT

First bug reported for TRIXBOX, straight off the oven, is that zaptel support is missing. To remedy this, please refer to the chapter relating to Rebuild Zaptel Driver earlier on in the document.

34.2 MISSING /VAR/SPOOL/ASTERISK/MEETME

It seems that the meetme directory does not exist in /var/spool/asterisk.

It will be necessary for this directory to be created otherwise if you have the *user join/leave* feature enabled, meetme will not function correctly.

34.3 EDITING EMAIL NOTIFICATION MESSAGE

This is hardly a bug but a feature ☺. When customising or editing your e-mail notification message in */etc/asterisk/vm_email.inc* you are advised to use an editor (such as nano) as I have found that AMP tends to loose the embedded formatting character.

34.4 WAKEUP CALLS

There is a problem with the Wakeup Calls - the *wakeconfirm* is not working.

35 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.

35.1 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 intercept the codes thinking that it is its own feature codes.

35.2 ASTERISK DROPS CALLS AFTER A FEW SECONDS

If Asterisk rings an extension but drops the calls too soon before it was answered, chances are, you may have set the '*Number of seconds to ring phones before sending callers to voicemail:*' too short. This setting can be found in the **General Setting** under **Voicemail**.

35.3 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.

35.4 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 TRIXBOX 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 sections in this guide:

[Calls Handling](#)

36 PORT FORWARDING AND QOS – ROUTERS

36.1 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 TRIXBOX 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.
RTP ports: 10000 – 10500 (UDP) – this should be enough.

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

Port Range					
Application	Start	End	Protocol	IP Address	Enable
IAX	4569	to 4569	Both	192.168.1.101	<input checked="" type="checkbox"/>
AMP*	8080	to 8080	Both	192.168.1.101	<input checked="" type="checkbox"/>
Azareus	50001	to 50010	Both	192.168.1.3	<input checked="" type="checkbox"/>
SIP	5060	to 5065	Both	192.168.1.101	<input checked="" type="checkbox"/>
RTP	10000	to 11000	UDP	192.168.1.101	<input checked="" type="checkbox"/>

Port Range Forwarding :
Certain applications may require to open specific ports in order for it to function correctly. Examples of these applications include servers and certain online games. When a request for a certain port comes in from the Internet, the router will route the data to the computer you specify. Due to security concerns, you may want to limit port forwarding to only those ports you are using, and uncheck the **Enable** checkbox.

These should be enough for starters.

36.2 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.

Applications & Gaming

Setup | Wireless | Security | Access Restrictions | **Applications & Gaming** | Administration | Status

Port Range Forward | Port Triggering | DMZ | QoS

Wired QoS Enable Disable

Upstream Bandwidth: Auto [0] Kbps

Device name	Priority	MAC Address
Asteris	Highest	00:0E:29:66:51:30
Family	Low	00:0F:3D:57:9C:B4
	Low	00:00:00:00:00:00

Two types of Quality of Service features are available, Wired QoS which controls devices plugged in to the router with a Ethernet cable, and Wireless QoS, which control devices that are wirelessly connected to the router.

Wired QoS

Device Priority: You may specify priority for all traffic from a device on your network.

In the application priority section (see below), I have defined the various ports that I want priority to be accorded the highest priority. In this case I have defined the port ranges related to my VOIP requirement.

Application Priority Optimize Gaming Applications

Application Name	Priority	Specific Port #
SIP	Highest	5060-5065
RTP	Highest	10000-11000
RTP	Highest	4569
	Low	0

Wireless QoS is also referred to as **Wi-Fi MultiMedia™ (WMM)** by the Wi-Fi Alliance™. Select Enable to utilize WMM if you are using other wireless devices that are also WMM certified.

No Acknowledgement: Enable this option if you would like to disable acknowledgement. If this option is enabled, the router will not re...

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.

(Note: This may not be the best solution. If anyone has a better solution for the Linksys WRT54G, please let me know).

37 UPDATING TRIXBOX

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 TRIXBOX as soon as there is a new update.

The following instruction may get you there. It's best to stop asterisk. Through SSH or the command prompt: type:

```
ampportal stop
```

Then followed by:

```
trixbox-update.sh
```

```
Installed trixbox-update.sh version is 1.0.0.0
Automatic update check...Version 1.0.0.5 available on the server.
Updating to the latest version ( 1.0.0.5 )...

--18:05:51-- http://upgrade.trixbox.org/trixbox-update.sh
=> `trixbox-update.sh'
Resolving upgrade.trixbox.org... 66.234.135.90
Connecting to upgrade.trixbox.org[66.234.135.90]:80... connected.
HTTP request sent, awaiting response... 200 OK
Length: 7,356 (7.2K) [application/x-sh]

100%[=====] 7,356 33.96K/s

18:05:51 (33.95 KB/s) - `trixbox-update.sh' saved [7356/7356]

New version successfully downloaded.
Can I put the new script in /usr/local/sbin (y/n)? █
```

You will get a prompt: 'Can I put the new script in /usr/local/sbin (y/n) ?

Type 'y' for yes. Then you will get the following:

```
New version successfully downloaded.
Can I put the new script in /usr/local/sbin (y/n)? y
chmod 500 /usr/local/sbin/trixbox-update.sh

Exiting so you can restart with the latest script.

[root@asterisk1 ~]# █
```

Now issue the following command:

```
trixbox-update.sh update
```

This process will take quite a few minutes while it downloads all the necessary updates and proceeded to install them. Make yourself a cup of coffee and wait.

You may get the following warning message.

```
Updating : mysql-devel          ##### [11/48]
Updating : mysql-server        ##### [12/48]
Updating : Pdp-pear            ##### [13/48]
Installing: kernel-devel       ##### [14/48]
Updating : sendmail            ## [15/48] warning: /etc/mail/submit.cf created as /etc/mail/submit.cf.rpmnew
Updating : sendmail            ##### [15/48]
Updating : tzdata              ##### [16/48]
Updating : xxx_gherkin         ##### [17/48]
```

Just ignore the warning and let the process continue.

```

--> Downloading header for munin-node to pack into transaction set.
munin-node-1.2.4-7.noarch 100% |=====| 19 kB 00:00
--> Package munin-node.noarch 0:1.2.4-7 set to be updated
--> Downloading header for perl-DateManip to pack into transaction set.
perl-DateManip-5.42a-3.no 100% |=====| 2.4 kB 00:00
--> Package perl-DateManip.noarch 0:5.42a-3 set to be updated
--> Running transaction check

Dependencies Resolved

=====
Package Arch Version Repository Size
=====
Installing:
munin noarch 1.2.4-7 trixbox 98 k
munin-node noarch 1.2.4-7 trixbox 141 k
perl-DateManip noarch 5.42a-3 base 135 k
perl-HTML-Template noarch 2.8-1.2.el4.rf trixbox 61 k
perl-Net-Server noarch 0.93-1.el4.rf trixbox 140 k
perl-rrdtool i386 1.0.50-3.2.el4.rf trixbox 177 k
rrdtool i386 1.0.50-3.2.el4.rf trixbox 519 k

Transaction Summary
=====
Install 7 Package(s)
Update 0 Package(s)
Remove 0 Package(s)
Total download size: 1.2 M
Downloading Packages:
(1/7): munin-1.2.4-7.noar 100% |=====| 98 kB 00:00
(2/7): perl-HTML-Template 100% |=====| 61 kB 00:00
(3/7): perl-Net-Server-0. 100% |=====| 140 kB 00:00
(4/7): rrdtool-1.0.50-3.2 100% |=====| 519 kB 00:02
(5/7): perl-rrdtool-1.0.5 100% |=====| 177 kB 00:00
(6/7): munin-node-1.2.4-7 100% |=====| 141 kB 00:00
(7/7): perl-DateManip-5.4 100% |=====| 135 kB 00:10
Running Transaction Test
Finished Transaction Test
Transaction Test Succeeded
Running Transaction
Installing: perl-DateManip ##### [1/7]
Installing: rrdtool ##### [2/7]
Installing: perl-rrdtool ##### [3/7]
Installing: perl-Net-Server ##### [4/7]
Installing: perl-HTML-Template ##### [5/7]
O:adding group munin...
O:adding user munin...
Installing: munin ##### [6/7]
O:group munin already present
O:user munin already present
Installing: munin-node ##### [7/7]

Installed: munin.noarch 0:1.2.4-7 munin-node.noarch 0:1.2.4-7 perl-DateManip.noar
ch 0:0.93-1.el4.rf perl-rrdtool.i386 0:1.0.50-3.2.el4.rf rrdtool.i386 0:1.0.50-3.
Complete!
done.

Please reboot by typing reboot

[root@asterisk1 ~]# █

```

At the end of it, you will be asked to reboot.

reboot.

You may or may not see the following error. In my case I did not get this error. If

you do not get the following error, skip this section. After reboot, and logging back in as root, go straight to rebuild_zaptel.

```
Error:
--> Finished Dependency Resolution
Error: Missing Dependency: zaptel = 1.2.5 is needed by
package zaptel-modules-2.6.9-34.EL
-----
done.
```

In my case I did not get this error.

If you do get the above error, you may have to update and reinstall the zaptel driver after rebooting. Otherwise skip the 2 'yum' instructions below and go straight to rebuild_zaptel.

```
yum -y remove zaptel

yum -y install zaptel-1.2.6

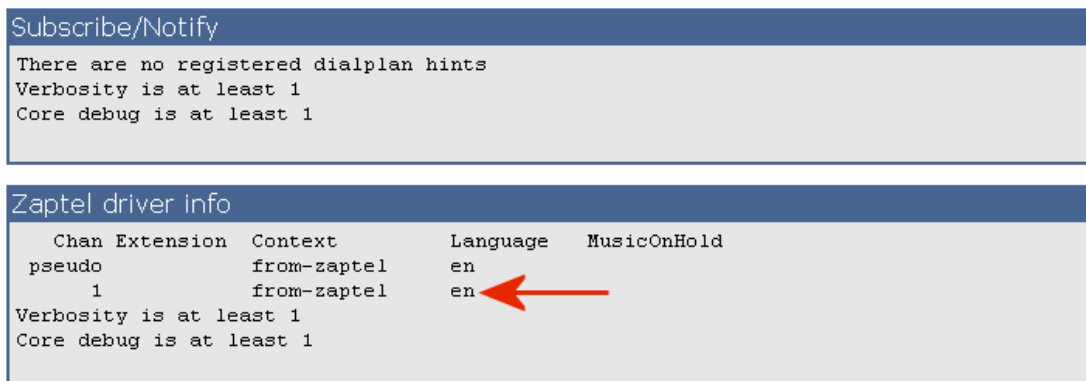
rebuild_zaptel

genzaptelconf
```

Reboot yet again.

During the reboot you will see some activities while Trixbox is updating your modules. Cross your fingers and you should have the latest version. AMEN!

Using your browser, connect to TRIXBOX and using the Asterisk Info option, check to see if zaptel is loaded.



The screenshot shows two sections of the Asterisk Info interface. The top section, titled 'Subscribe/Notify', contains the text: 'There are no registered dialplan hints', 'Verbosity is at least 1', and 'Core debug is at least 1'. The bottom section, titled 'Zaptel driver info', contains a table with the following data:

Chan	Extension	Context	Language	MusicOnHold
	pseudo	from-zaptel	en	
	1	from-zaptel	en	

Below the table, it also shows 'Verbosity is at least 1' and 'Core debug is at least 1'. A red arrow points to the 'en' language entry in the second row of the table.

Yesssss! Zaptel found.

APPENDIX A

A.1 ASTERISK FEATURE CODES

*8	Call Pickup
*43	Echo Test
*60	Time
*61	Weather
*62	Schedule wakeup call
*65	Festival test (your extension is XXX)
*69	Last Caller number/ID
*70	Activate Call Waiting (deactivated by default)
*71	Deactivate Call Waiting
*72	Call Forwarding System
*73	Disable Call Forwarding
*77	IVR Recording
*78	Enable Do-Not-Disturb
*79	Disable Do-Not-Disturb
*90	Call Forward on Busy
*91	Disable Call Forward on Busy
*97	Message Center (does not ask for extension)
*97 + 0	Change email greeting message.
*98	Enter Message Center
*99	Playback IVR Recording
*411	Directory (while in IVR – similar to #)
70	Call Parking (or 700 see Call Parking and Transfer)
666	Test Fax
888	Barge in (there are variations)
7777	Simulate incoming call

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 TRIKBOX – **amportal stop**

To start TRIKBOX – **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 TRIKBOX and
amportal stop: Restart TRIKBOX.
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 verbosity
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 is 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.

<code>\${ACCOUNTCODE}</code>	Account code (if specified)
<code>\${CALLERID}</code>	Caller ID
<code>\${CALLERIDNAME}</code>	Caller ID Name only
<code>\${CALLERIDNUM}</code>	Caller ID Number only
<code>\${CALLINGPRES}</code>	PRI Caller ID presentation for incoming calls
<code>\${CHANNEL}</code>	Current channel name
<code>\${CONTEXT}</code>	Current context
<code>\${DATETIME}</code>	Current date time in the format: YYYY-MM-DD_HH:MM:SS
<code>\${DNID}</code>	Dialed Number Identifier
<code>\${ENUM}</code>	Result of application EnumLookup
<code>\${EPOCH}</code>	Current unix style epoch
<code>\${EXTEN}</code>	Current extension
<code>\${ENV(VAR)}</code>	Environmental variable VAR
<code>\${HANGUPCAUSE}</code>	Asterisk hangup cause
<code>\${INVALID_EXTEN}</code>	The invalid called extension (used in the "i" extension)
<code>\${LANGUAGE}</code>	Current language
<code>\${LEN(VAR)}</code>	String length of VAR (integer)
<code>\${MEETMESECS}</code>	Number of seconds a user participated in a MeetMe
<code>\${PRIORITY}</code>	Current priority
<code>\${RDNIS}</code>	Redirected Dial Number ID Service
<code>\${TIMESTAMP}</code>	Current date time in the format: YYYYMMDD-HHMMSS
<code>\${TXTCIDNAME}</code>	Result of application TXTCIDName
<code>\${UNIQUEID}</code>	Current call unique identifier
<code>\${SIPCALLID}</code>	SIP Call-ID: header verbatim (for logging or CDR matching)
<code>\${SIPDOMAIN}</code>	SIP destination domain of an inbound call (if appropriate)
<code>\${SIPUSERAGENT}</code>	SIP user agent

The following are set by the dial() application:

<code>\${DIALEDPEERNAME}</code>	Dialed peer name
<code>\${DIALEDPEERNUMBER}</code>	Dialed peer number
<code>\${DIALEDTIME}</code>	Total time for the call in seconds (Network time).
<code>\${ANSWEREDTIME}</code>	Time from answer to end of call in seconds.
<code>\${DIALSTATUS}</code>	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,!400/100,
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,
!440/100,440

[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

[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,!1400/330,!1800/330,!0/1000,0
stutter = 470/400,425/400

[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,!1400/330,!1800/330,!0/1000,0
stutter =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,425

[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,!350+440/100,!0/100,350+440

[n!]

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,!425/100,
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,!1800/332,
!0/2024,!950/332,!0/24,!1400/332,!0/24,!1800/332,!0/2024,!950/332,!0/24,!1400/332,!0/
24,!1800/332,!0/2024,!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,!425/100,
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,!425/600,
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,!350+440/100,!0/100,!350+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,!350+440/100,!0/100,!350+440/100,!0/100,350+440

A.6 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 won't be able to decompress it and vice versa.

Note: 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 TRIKBOX 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 ☺

APPENDIX B

B.1 – MORE TRUNKS

The following is a collection of TRUNK settings recommended for other VSPs that are not covered above. Details of trunks configuration are correct at the time they were compiled, however changes may have been made. Users are advised to consult their respective VSPs for confirmation.

Note: In Peer Details, insert **qualify=yes** if you want fail-over to PSTN should you loose your internet connection.

B.1.1 [BBPglobal](#)

Peer Details

Trunk Name: bbpglobal

```
allow=g723&gsm
authuser=<user number>
disallow=all
fromdomain=sip2.bbpglobal.com
fromuser=<user number>
host= sip2.bbpglobal.com
insecure=very
qualify=yes
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>
```

Register String:

```
<user number>:<password>@sip2.bbpglobal.com/<user number>
```

B.1.2 [Engin BYO](#)

Peer Details

Trunk Name: engin

```
allow=ulaw&alaw
auth=md5
canreinvite=yes
disallow=all
dtmfmode=rfc2833
fromdomain=voice.mibroadband.com.au
fromuser=02321XXXX
host=byo.engin.com.au
insecure=very
musiconhold=framed
nat=yes
port=5060
qualify=no
realm=mobileinnovations.com.au
reinvite=yes
secret=<password>
type=friend
username=02321XXXX
```

User 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**

```
Defaultexpiry=600
Maxexpiry=3600
```

Register String:

```
02321XXXX:<password>@byo.engin.com.au/02321XXXX
```

B.1.3 Faktortel (IAX)**Peer Details**

Trunk Name: faktortel

```
allow=g729&gsm&ulaw&alaw&ilbc
disallow=all
host=iax.faktortel.com.au
qualify=3000
secret=<password>
type=friend
username=xxxxxxx
```

User Details

User Context: xxxxxxx

```
context=from-trunk
host=iax.faktortel.com.au
qualify=3000
type=friend
username=xxxxxxx
```

Register String:

```
xxxxxx:<password>@iax.faktortel.com.au
```

Faktortel supports the following codecs: ulaw, alaw, ilbc, gsm, g729

B.1.4 FWD aka Pulver(IAX)**Peer Details**

Trunk Name: fwd

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

User Details

User Context: 65xxxx

```
auth=rsa
context=from-trunk
inkeys=freeworlddialup
type=user
```

Register String:

```
65xxxx:<password>@iax2.fwdnet.net
```

B.1.5 iinet

The following are the various iinet sip proxies for the different states (at the time of writing). You may use the sip proxy designation or the dot format IP address when configuring your iinet setting in TRIKBOX.

State	SIP Server	IP
act	sip.act.iinet.net.au	203.55.231.193
nsw	sip.nsw.iinet.net.au	203.55.231.199
nt	sip.nt.iinet.net.au	203.55.229.193
qld	sip.qld.iinet.net.au	203.55.228.194
sa	sip.sa.iinet.net.au	203.55.229.193
tas	sip.tas.iinet.net.au	203.55.229.193
vic	sip.vic.iinet.net.au	203.55.229.193
wa	sip.wa.iinet.net.au	203.59.49.5

Since VSP may change their sip IP address, it is advisable to use the sip proxy designation rather than the dot format IP address.

Peer Details

Trunk Name: iinetout

```

allow=alaw&ulaw
canreinvite=no
context=ex-did
disallow=all
fromdomain=iinetphone.iinet.net.au
fromuser=073XXXXXXX
host=203.55.228.193
insecure=very
nat=no
pedantic=no
secret=<password>
type=peer
username=073XXXXXXX

```

User Details

User Context: iinet-inbound

```

Canreinvite=no
context=from-trunk
fromuser=073XXXXXXX
host=203.55.228.193
insecure=very
qualify=no
secret=<password>
type=user
username=073XXXXXXX

```

Register String:

```
073XXXXXXX@iinetphone.iinet.net.au:Password:073XXXXXXX@iinetout/073XXXXXXX
```

B.1.6 iTalk (New Zealand)**Peer Details**

Trunk Name: italk

```

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

```

User Details

User Context:

You may prefix outgoing calls with 0197 to disable outgoing CID.

Inbound route 64997xxxxx required to be forwarded to a destination ext/menu

Register String:

```
64997xxxxx:<your password>@akl.italk.co.nz/64997xxxxx
```

*(Source Ref: Steve Biddle)***B.1.7 Koala****Peer Details**

Trunk Name: koalaSip

```

allow=g729&gsm&alaw&ulaw
disallow=all
fromuser=<User-sip id>
host=203.122.248.173
nat=yes
port=5060
qualify=no
secret=<password>
type=friend

```

User Details

User Context: <User-sip id>

```

context=from-pstn
fromdomain=203.122.248.173
host=203.122.248.173
secret=<password>
type=user
username=<User-Sip id>

```

Register String:

```
<User SIP ID>:<password>@koalavoip.com.au
```

(Source Ref: [Curtis of Koala](#)) also refer here <https://www.koalavoip.com.au/billing/node/75>

B.1.8 [MyNetFone](#)

Peer Details

Trunk Name: MyFonesip

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

User Details

User Context: 091XXXXX

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

Register String:

```
091xxxxx@sip.myfone.com.au:<password>:091xxxxx@sip.myfone.com.au/091xxxxx
```

B.1.9 [Nehos \(IAX\)](#)

Peer Details

Trunk Name: Nehos

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

User Details

User Context: 661XXXX

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

Register String:

```
661xxxx:<password>@iax.ifone.com.au
```

B.1.10 [Nodephone](#)

Peer Details

Trunk Name: Nodephone

```
allow=g729
canreinvite=no
disallow=all
dtmfmode=rfc2833
fromdomain=sip.internode.on.net
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
secret=<password>
type=user
username=<usernumber>
```

Register String:

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

B.1.11 [SipBroker](#)

Peer Details

Trunk Name: sipbroker-out

```
allow=g729&ulaw&alaw
canreinvite=no
disallow=all
dtmfmode=rfc2833
fromdomain=<your existing sip provider>
fromuser=<your user ID of existing provider>
host=sipbroker.com
insecure=very
nat=yes
port=5060
secret=<password for existing provider>
type=peer
```

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

You do not need to register either.

If you are using SipBroker, you must ensure that your sip_nat.conf is modified to the following;

```
nat=yes
```

```
externip=<Fix Ip address> ; if you have fix IP or
```

```
externhost=<your DNS hostname> ; if you are using Dynamic IP
```

```
localnet=192.168.1.0/255.255.255.0
```

B.1.12 [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.1.13 [SipPhone](#)

Peer Details

Trunk Name: sipphone

```
fromdomain=proxy01.sipphone.com
host=proxy01.sipphone.com
insecure=very
secret=<password>
type=peer
username=1747xxxxxxx
```

User Details

User Context: 1747xxxxxxx

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

Register String:

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


B.1.14 [TelephoneGlobal](#)

Peer Details

Trunk Name: Teleglobal

```
allow=g729
canreinvite=no
disallow=all
host=210.80.182.142
insecure=very
secret=<password>
type=peer
username=2222xxxxxxxx
```

User Details

User Context: 2222xxxxxxxx

```
context=from-pstn
fromuser=2222xxxxxxxx
secret=<password>
type=user
```

Register String:

```
2222xxxxxxxx:<password>@210.80.182.142/2222xxxxxxxx
```

B.1.15 [VOIP Buster](#)

Peer Details

Trunk Name: VoipBuster

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

User Details

User Conext: <leave blank>

If you have an inbound number, an Inbound route is required to forwarded your incoming call to a destination extension or IVR menu:

Inbound Route:

DID: <User inbound Number>
No Fax

Register String:

```
<username>:<password>@sip1.voipbuster.com/<username>
```

(Source Ref: Florent Chandelier)

B.2 RELATED LINKS – (FOR VSP TRUNK SETTINGS)

The following links contain very useful information when setting up your TRIXBOX. I recommend that you should, at least, give them a cursory glance.

B.2.1 Trunk and Route Settings

The following link contains Trunks and Route settings put together by Colin Swan. It is quite comprehensive and certainly worth a look.

http://members.ozemail.com.au/~bsharif/aah/trunk_settings.htm

B.2.2 MyFone settings

<http://www.myfone.com.au/downloads/sip.conf>

<http://www.myfone.com.au/downloads/extensions.conf>

B.2.3 Setting up Faktortel for TRIXBOX

The following is the link on the discussions for setting up [Faktortel](#) on TRIXBOX.

<http://voipinfo.faktortel.com.au/viewtopic.php?t=135>

Faktortel supports the following codecs:

- ulaw
- alaw
- ilbc
- gsm
- g729

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.cbsservers.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 Acoustica 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/TVR.htm>

<http://www.voip-info.org/tiki-index.php?page=Asterisk+sound+files>

<End quote>

C.2 AUSTRALIAN VOICE AND LANGUAGE SET FOR ASTERISK

VOIPSHOP has made available the files free to non commercial users. Log on to http://www.voipshop.com.au/product_info.php?products_id=72 to download

The following is extract of the instructions by VOIPSHOP on their web site. Two files are available; one is as a replacement for the current asterisk files. The other is as an additional Asterisk language (if you are going to use the SetLanguage option).

Replace Current Asterisk Files: ausounds.tar.gz

Quick Steps for TRIXBOX.

Log into your TRIXBOX server and at the command prompt, do the following:

```
wget http://files.voipshop.com.au/ausounds.tar.gz
tar cvfz backupsounds.tar.gz /var/lib/asterisk/sounds/ (for
backing up your existing files)
tar -zxvf ausounds.tar.gz -C /var/lib/asterisk/sounds/
```

Done. Your asterisk sound files are replaced. Make sure you back up, just in case.

Additional Asterisk Language Set: ausoundset.tar.gz

At the command prompt execute the following commands:

```
wget http://files.voipshop.com.au/ausoundset.tar.gz
tar -zxvf ausoundset.tar.gz -C /var/lib/asterisk/sounds/
```

Done. Your asterisk sound files now contain an au soundset.

Note : **You can only download these files from voipshop,**

Please do not re-distribute. These are free for non-commercial use only. Contact VOIPSHOP <mailto:support@voipshop.com.au> for commercial license details.

C.3 MORE AUSTRALIAN VOICE 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 setup described here is for use with TRIXBOX 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.

1. Select **Admin Login** and then select *(switch to advanced view)*
2. Select the **System** Tab
 - a. Under **Internet Connection Type**, change the **Gateway** address to the address of your router.
3. Select **Regional** Tab
 - a. Under **Vertical Service Activation Codes**, delete all the fields.
4. 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**.
5. 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**.

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 (USERS' SUGGESTIONS)

E.1 SUGGESTIONS FOR DIAL PLAN AND CONFIGURATION

E.1.1 Astratel

Colin Swan submitted the following Astratel configuration on TRIXBOX.

General Settings

Outbound Caller ID: <8888xxxx>

Maximum Channels: 1

Outgoing Dial Rules

Dial Rules: (which tie in with his Outbound Routing method)

```
02+4XXXXXXXX
02+6XXXXXXXX
02+80XXXXXXXX
02+81XXXXXXXX
02+82XXXXXXXX
02+83XXXXXXXX
02+84XXXXXXXX
02+85XXXXXXXX
02+86XXXXXXXX
02+87XXXXXXXX
02+89XXXXXXXX
02+880XXXXX
02+881XXXXX
02+882XXXXX
02+883XXXXX
02+884XXXXX
02+885XXXXX
02+886XXXXX
02+887XXXXX
02+889XXXXX
02+8880XXXX
02+8881XXXX
02+8882XXXX
02+8883XXXX
02+8884XXXX
02+8885XXXX
02+8886XXXX
02+8887XXXX
02+8889XXXX
02+9XXXXXXXX
0+NXXXXXXXXX
0011+1NXXNXXXXXXXX
0011+44ZXXXXXXXXXXXX
0011+49NXXXXXXXX.
```

Outbound Dial Prefix: (empty)

Outgoing Settings

Trunk Name: astratel

Peer Details:

authuser=8888xxxx

context=from-trunk


```
fromdomain=sip01.astrasip.com.au
fromuser=8888xxxx
host=sip01.astrasip.com.au
insecure=very
secret=xxxx
type=peer
username=8888xxxx
```

Incoming Settings

User Context: 8888xxxx

User Details:

```
context=from-trunk
insecure=very
secret=xxxx
type=user
```

Registration

Register String:

```
8888xxxx:xxxx@sip01.astrasip.com.au/8888xxxx
```

E.1.2 Further .conf modifications by Colin

extensions_trixbox.conf

add...

; only for FWD

```
[from-fwd-custom] ; won't work to 's' extension for me.
exten => _X.,1,Wait(2) ; the following is so that any callers to my FWD
exten => _X.,2,Answer ; number will be "reminded" as to what my local
exten => _X.,3,Wait(1) ; time is, and then for good measure, they'll have to
exten => _X.,4,SayUnixTime(, ,Imp); provide the correct password before continuing
exten => _X.,5,DigitTimeout(5) ; to my "family" menu
exten => _X.,6,ResponseTimeout(10)
exten => _X.,7,Authenticate(****) ; change '****' to your chosen digits
exten => _X.,8,Goto(aa_2,s,1)
```

; only for sixTel

```
[from-sixtel-custom]
exten => _X.,1,Goto(aa_2,s,1) ; won't work to 's' extension for me
```

; only if you want a "DISA" facility – point one of your IVR options to: *custom-disa,s,1*

```
[custom-disa]
exten => s,1,Answer
exten => s,2,DigitTimeout(5)
exten => s,3,ResponseTimeout(10)
exten => s,4,Authenticate(****); change '****' to your chosen DISA password
exten => s,5,DISA(no-password|from-internal)
```

iax.conf

```
[general]
externip= *.*.*.* ; ip address or hostname, DynDNS is OK
```

```
localnet=192.168.1.0/255.255.255.0 ; change to suit
bindport = 4569 ; Port to bind to (IAX is 4569)
bindaddr = 0.0.0.0 ; Address to bind to all addresses on machine
delayreject=yes
disallow=all
allow=ulaw ; allow desired codecs in preferred order
allow=alaw
allow=g729
allow=ilbc
allow=gsm
jitterbuffer=yes
mailboxdetail=yes

#include iax_additional.conf
```

indications.conf

change...

```
country=au ; default is "us"
```

rtp.conf

change...

```
rtpend=10100 ; was 2000. 10,000 ports seems like overkill
```

sip.conf

```
[general]
port = 5060 ; Port to bind to (SIP is 5060)
bindaddr = 0.0.0.0 ; Address to bind to all addresses on machine
disallow=all
allow=ulaw ; allow desired codecs in preferred order
allow=alaw
allow=g729
allow=ilbc
allow=gsm
context = from-trunk ; Send unknown SIP callers to this context
callerid = Unknown

#include sip_nat.conf
#include sip_additional.conf
```

sip_nat.conf

add...

```
nat=yes
externip= *.*.*.* ; ip address or hostname – even DynDNS is OK
localnet=192.168.1.0/255.255.255.0 ; change to suit
```

E.2 MULTIPLE TDM400P INSTALLATION

Neilmc, a participant I the Whirlpool Forum provided the following feedback on the use of multiple TDM400P Digium cards.

This is being discussed in this thread:

<http://forums.whirlpool.net.au/forum-replies.cfm?t=483550&p=1>

<quote>

There are plenty of forum posts (on WP and elsewhere) + various how-to and guides that say don't use more than one if you want a reliable system with no audio problems.

I'm sure that on the old Digium site they had a caution not to use more than one TDM or TE card (That's changed, they now brag about 1 box running 5 quad span TE cards).

This machine is now working in a busy environment (medical centre) nicely, sometimes under full Zap call load. It has 5 x FXO & 5 x FXS on 3 cards. CPU is under 10% pretty much all the time.

I asked them to be very fussy about audio quality and let me know if they have any echo, pops, clicks, distortion etc.

So far everyone is saying that it's perfectly fine, not a bad call yet.

There is good reason behind the old advice not to do it though. Lots of people have had grief.

Make sure you don't grab any old motherboard that is lying about (especially if it isn't PCI 2.2 compliant) Even though minimum system specs aren't high don't go for a bottom of the range el-crapo brand motherboard. There is a list of some incompatible motherboards on the Digium site, but no doubt there will be others that have problems.

Choose a board that has plenty of PCI slots (5 or so). Make sure that it has plenty of control over IRQ in CMOS. eg the ASUS P5P-800 I used could use APIC to assign IRQ or you could manually assign an IRQ to a particular PCI slot.

Be very sure that each Digium card is not sharing an IRQ with anything else.

Disable any onboard devices you don't need inc. serial, parallel & USB ports.

It may not be a problem, but to minimise chances of problems disable Hyper threading if it is supported on your CPU. Use 32 bit OS rather than 64 bit.

Use plenty of RAM. You don't want a PABX to be thrashing about with a swap file. I used 1GB of decent quality RAM which is definitely far more than the system needs. (It's using about 230MB at the moment), but 1GB of RAM doesn't really cost much any more.

Check that your hard drives are running in DMA mode (or use SCSI drives). Sometimes they default to PIO, which might cause problems if there is a sudden burst of disk activity.

Sometimes APIC can cause you grief. You might be able to tweak your kernel, but you might get what you need by turning it off.

Have a read through

http://www.asteriskguru.com/tutorials/pci_irq_apic_tdm_ticks_te410p_te405p_noise.html

If you have all of the driver stuff loading properly and still have audio problems.

Use recent zaptel drivers (unless you become aware of an issue with them).

- Make sure that the driver loads with the module option `opermode=AUSTRALIA`.
- Run `fxotune -i 4` (usually from `/usr/src/zaptel`) It will create a file `/etc/fxotune.conf` with settings for your cards based on tests with your lines (takes about 5 minutes per FXO port).
- You will have to stop Asterisk before running `fxotune`.
- Have `/usr/src/zaptel/fxotune -s` run at startup to have the card set. Apparently the values in the conf file are often zero if you have the modules in the correct opermode. (have one module at the office where the first value is 10 rather than zero. I'll try `fxotune` again and see if it changes at all).

I just put `/usr/src/zaptel/fxotune -s` at the bottom of `/etc/rc.d/rc.local` in A@H 2.6 and it loaded okay. You need it to run after the drivers load but before asterisk starts.

- Check `/var/log/messages` after bootup. Each FXO port should have loaded with AUSTRALIA mode. If it is FCC mode then the card is set for North American lines.
- Each FXS port should have an entry for boosting ringer.

If `fxotune` is loading settings okay you should see a line saying something like after all of the other init stuff.

```
kernel: -- Setting echo registers:  
kernel: -- Set echo registers successfully
```

If you have 4 FXO ports, you should see this 4 times.

<\quote>

E.3 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 it 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 which 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 its 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 too 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.

E.4 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.

E.5 REBUILD ZAPTEL DRIVER (USERS' SUGGESTIONS)

E.5.1 As experienced by marner – a Whirlpool Forum participant.

On his Dell box, marner had to do the following to rebuild the driver and for it to work.

Log in as root and issue the following command:

```
cd /usr/src/kernels/2.6.9-34.EL-smp-i 686/include/linux
wget http://nerdvittles.com/aah27/spinlock.h
```

The rest of the procedure is as per earlier chapter [Rebuilding Zaptel Driver](#)

E.5.2 As suggested by Rob Thomas (the FreePBX guru)

Rob's suggestion is to edit the file
/usr/src/kernels/2.6.9-34.EL-i686/include/linux/spinlock.h
and change 'rw_lock_t' on line 407 to 'rwlock_t'

Once that is done, do the following:

```
rebuild-zaptel
```

That seems to be a lot easier.

E.6 ZAPTEL CARD CONFIGURATION FOR AUSTRALIA

Submitted by Stephen Gleeson – Technical Manager for Community Information Strategies Australia Inc (CISA).

<http://gleesos.wordpress.com/2006/02/08/voip-with-asterisk/>

Stephen has attached samples of Zapata configuration files – with correct settings for Australia – loop start.

zaptel.conf

```
# Autogenerated by /usr/local/sbin/genzaptelconf -- do not hand
edit
# Zaptel Configuration File
#
# This file is parsed by the Zaptel Configurator, ztcfg
#
# It must be in the module loading order
```



```
# Span 1: WCTDM/0 "Wildcard TDM400P REV I Board 1"
#
# Update for regional - ls - loopstart au - zone
#
fxols=1
fxsls=2
fxsls=3
fxsls=4

# Global data

loadzone = au
defaultzone = au
```

zapata_addiotional.conf

```
;;;;[230]
signalling=fxo_ls
record_out=Adhoc
record_in=Adhoc
mailbox=230@device
echotraining=800
echocancelwhenbridged=no
echocancel=yes
context=from-internal
callprogress=no
callerid=device <230>
busydetect=no
busycount=7
channel=>1
```

zapata.conf

```
;
; Zapata telephony interface
;
; Configuration file

[trunkgroups]

[channels]

language=en
context=from-pstn
signalling=fxs_ls
rxwink=300 ; Atlas seems to use long (250ms) winks
;
; Whether or not to do distinctive ring detection on FXO lines
;
;usedistinctiveringdetection=yes

usecallerid=yes
hidecallerid=no
callwaiting=yes
usecallingpres=yes
callwaitingcallerid=yes
threewaycalling=yes
transfer=yes
```

```
cancallforward=yes
callreturn=yes
echocancel=yes
echocancelwhenbridged=no
echotraining=800
rxgain=0.0
txgain=0.0
group=0
callgroup=1
pickupgroup=1
immediate=no

;faxdetect=both
faxdetect=incoming
;faxdetect=outgoing
;faxdetect=no

;Include genzaptelconf configs
#include zapata-auto.conf

;Include AMP configs
#include zapata_additional.conf
```

NOTES

A2BILLING

A2Billing is a calling card platform to manage calling card users' accounts. To log on to A2Billing through Web Interface, connect to ***asterisk.ip.address/a2billing*** e.g. ***192.168.0.101/a2billing***

A2Billing is not being covered in this guide, as it is a whole documentation exercise on its own. Those interested in A2Billing may have to refer to other source for instructions available on the net such as the following link:

http://sourceforge.net/forum/forum.php?thread_id=1398290&forum_id=420324
<http://www.voip-info.org/wiki/view/Asterisk+billing>

CONVERSATION RECORDINGS

There have been numerous discussions on the legality of recording telephone conversations. Some say its OK while others say its illegal. If you are recording conversation without the consent of all parties, you may be breaking the law – so be warned.

For those who do record conversations, the recording are kept in the ***/var/spool/asterisk/monitor*** directory or you can monitor the recording through the ARI interface of freePBX.

If you want to monitor all the calls, you will need to log in as ***admin*** with the default password of ***ari_password***.

To monitor the calls for any individual extension, log in using the ***extension number*** and the ***extension password***.

The configuration file for ARI is ***/var/www/html/recordings/includes/main.conf.php***

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.

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})  
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.**

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 ***extensions_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.

DELETE CALL RECORDS FROM CDR

There will be time when you need to delete the CDR records from your call details record. One example is, those calls you made white testing will need to be deleted before going live where you will need clean record.

- These records are being kept in the SQL database and to delete them, do the following:
- From Tools Menu
- select phpmyadmin
- In the Database dropdown field, top left, select asteriskcdrdb. Select the 'cdr' table in the tick box
- At the bottom of the list of table you will find a dropdown field "With Selected", select EMPTY
- You will be asks to 'confirm truncate table cdr',
- If you click Yes at this point the contend of the table will be erased.

EXTENSION NUMBERS TO AVOID USING

Unless you are prepared to edit and change some codes, its best to avoid the following extension numbers:

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

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 that "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
```

SIP_NAT.CONF

To those who use this configuration (not all do), I have an issue with **externip = <DNS name>**. I wasn't able to accept incoming calls although SIP was registered. Changing to **externip=<my fixed external IP>** if you have a fixed IP or **externhost=<DNS name>** if you don't have a fixed IP and use Dynamic DNS service, solved the issue or do the alternative as I have outlined below.

This issue may not be apparent at first as Asterisk will show that it is registered and you will still be able to make a call out (except calls to another Asterisk that have this property set). Another telltale sign is, you may not hear the ring tone when you are calling a party or you may have audio problem.

This is caused by my **/etc/hosts** file where I have an entry (in red)

```
127.0.0.1          localhost
127.0.0.1          pcnovation.homelinux.org asterisk1.local
```

SipBroker will resolve pcnovation.homelinux.org as 127.0.0.1 which is what it is suppose to be as it is being defined as such by the offending line. In actual fact pcnovation.homelinux.org should be resolved to the external IP address. However I need that entry for my VoiceMail e-mail notification since most email server will not accept email from unknown source – grief time.

Taking my DNS name out solved the issue, but I cannot send e-mail notification of my voicemail. To fix this problem (a kludge) I substitute **pcnovation.homelinux.org** with another valid domain name different to my DynDNS domain name such as **yahoo.com** thus;

```
127.0.0.1          yahoo.com asterisk1.local
```

and problem solved... for now until something else pops up ☺

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.

Greg Hind (a participant in the Whirlpool Forum) provided a solution with the following settings:

PSTN Line Tab

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

I deleted my original Disconnect Tone setting of: 425@-30,425@-30,1(.375/.375/1+2)

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.

TRUNKS AND OUTBOUND ROUTES – A CIRCUS OF CONFUSION

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

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

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

```
0|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.

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.

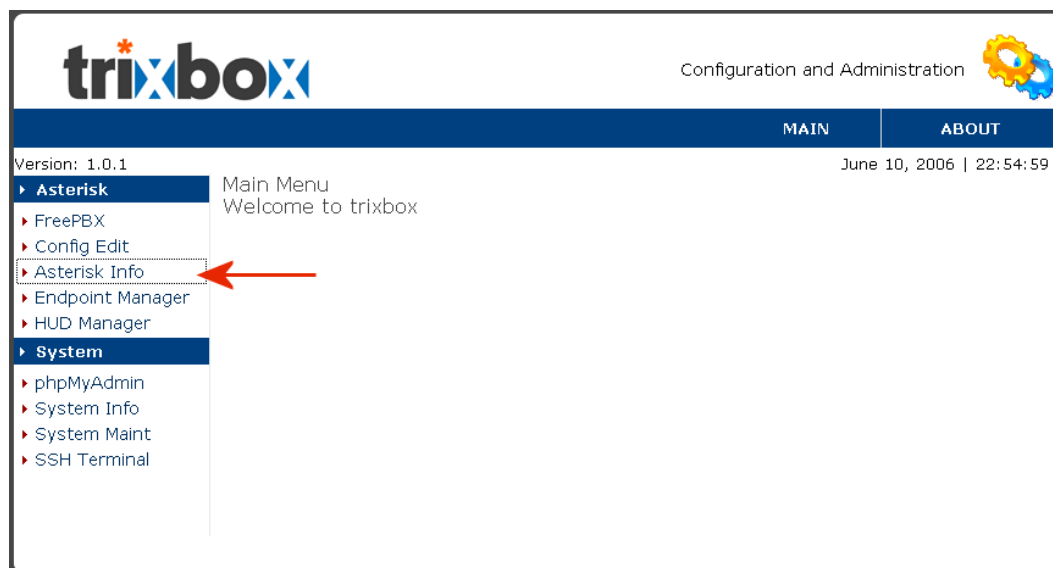
In your Trunk Outgoing Dial Rule, you will need the following pattern.

```
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 the previous chapter earlier on in this document.

ADDING THE MISSING MAINTENANCE MODULES TO FREEPBX 2.1.0

Note: If you are already using TRIKBOX v1.0.1 or later, you will not be required to do the solution as described below. The Asterisk Info is included in the latest TRIKBOX (see arrow).



Some of us have gotten used to and became quite attached to the maintenance modules in AMP but unfortunately they were taken out of FreePBX. 2.1.x that comes with TRIKBOX. If you are like me, still wanting the Asterisk Info module, this is what you have to do.

- Download the missing modules here: www.voipeye.com.au/invoke/maint-modules.zip (compliment of Mark Brooker).
- Copy the folder **asterikinfo** (don't bother with the others as they are in TRIKBOX) in the zip into the **var/www/html/admin/modules** folder of TRIKBOX (you could use WinSCP to do this).
- Next, ensure that 'status.pl' is executable. At the linux command prompt enter the following command:

```
chmod 775 /var/www/html/admin/modules/sysstatus/status.pl
```
- Then go into your modules admin in the browser and install the new Asteriak Info module. Once this is done go to tools and you should have Asterisk Info as one of the option - Showing all your extensions and trunks.

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.

Andrew Gillis	<i>The man who started it all – The Maha Guru</i>
Asterisk Guru	<i>http://www.asteriskguru.com</i>
Brian	<i>Advise tips and tricks- Voice for Asterisk</i>
Bob Fryer	<i>A friend from a long way back. From the days I learned to use a PC and according to Bob, I still cant use it.</i>
Brian	<i>Advise tips and tricks- Voice for Asterisk</i>
Colin Swan	<i>Major assistance, advice, and basically a place to go when things go wrong.</i>
Graham Foote	<i>Always giving me tips and corrections</i>
Greg Hind	<i>Advise, Tips and tricks</i>
Jack Zimmermann	<i>Helping me with e164</i>
Jeffrey Borg	<i>Advise, Tips and tricks</i>
lusyn.com	<i>X100P Patch for UK Caller ID http://www.lusyn.com</i>
Mark Brooker	<i>Another of my source for help when things go wrong</i>
Matt (daggo)	<i>Advise, Tips and tricks</i>
Neilmc	<i>Advise, Tips and tricks</i>
Nathan Poyner	<i>Assistance with Scripts and also proofing the documents</i>
Openvoice	<i>Source for Australian Voice</i>
Peter Quodling	<i>Testbed and resident cynic</i>
Rob Thomas	<i>freePBX Developer and Super Hero – without which there will be no Trixbox</i>
Sammy/sz	<i>Taken time to point out some of my mistakes</i>
Sean Mahon	<i>Solution for Billion 7402 BL</i>
Shaun (Ewing)	<i>Advise, Tips and tricks on codecs and dial plan etc</i>
Stefan Keller-Tuberg	<i>Helping me out with Cron Jobs and Scripts</i>
SteveM	<i>Advise, Tips and tricks on Webmin</i>
Steven D	<i>Help with detecting some error and proofing the document</i>
Thunderbird1/Chippy	<i>Giving me clues to some obscure processes. It is Chippy who solved the Fax problem that has been bugging us all.</i>
VoipShop	<i>Source for Australian voice and others</i>
Voxilla.com	<i>http://voxilla.com</i>
Ward Mundy	<i>My hero http://mundy.org my place of salvation ☺.</i>

CHANGES MADE SINCE LAST UPDATE