

Trixbox-2 without Tears can be downloaded [here](#).

TRIXBOX without Tears

The ICT serial following
Asterisk@Home without Tears

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If you are in Australia and don't have Paypal, you may forward a cheque made to
:Ben Sharif, PO Box 70 Kingswood NSW 2747, Australia.

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Without Tears

<http://dumbme.mbit.com.au/trixbox/index1.htm>

The Other Dumb-Me Guide

(Australianised for AU&NZ users)

Versions 1.x with freePBX

(Version 1.2.3e)

by

Ben Sharif

(The Author of Asterisk@Home without Tears)

[About the Author](#)

Last updated on November 30, 2006

Please send comment or suggestions to me at the address below

<mailto:netvoice.solutions@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

Foreword	8
1 Introduction.....	9
1.1 The Components.....	9
1.1.1 The IP PBX	9
1.1.2 Phones.....	10
1.1.3 SIP Gateway	10
1.1.4 Home Network.....	10
1.1.5 VOIP Service Providers	10
2 Is VOIP for You?	11
2.1 What is it going to cost?	11
2.2 What will the Quality of the phone calls be?	11
3 Installation.....	12
3.1 Change default Settings	15
3.1.1 To get Help	16
3.1.2 Change IP Address (set IP address to Static)	16
3.2 Configure TRIXBOX using Web Browser.....	18
3.2.1 Log in to System Administration	18
3.2.2 Configuring Trixbox using freePBX.....	19
3.3 General Settings.....	22
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.....	37
5.1 Inbound Routes (Incoming Calls).....	37
5.2 Time Conditions	39
Simple Time Condition.....	39
Complex Time Condition.....	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	43
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 Outgoing Calls.....	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
7.3 Multi-Language IVR.....	55

7.3.1 Complimentary multi-language MOH.....	57
7.4 Multi-Tenants IVR.....	57
8 Get Under The Bonnet	60
8.1 Editing The .conf Files.....	61
8.1.1 sip.conf.....	61
sip_custom.conf	62
8.1.2 iax.conf.....	62
lax_custom.conf	62
8.1.3 Indications.conf.....	62
8.1.4 enum.conf	62
8.1.5 extensions_custom.conf	63
8.1.6 features.conf	63
9 Check your System	65
9.1 System Process Status	65
9.2 Asterisk Info.....	66
10 Interfacing Asterisk to PSTN.....	68
10.1 Digium Wildcard X100P FXO PCI Card.....	68
10.1.1 zapata.conf	68
10.1.2 zapata-auto.conf	69
10.1.3 zaptel.conf.....	69
10.2 Digium TDM400P FXO/FXS Card.....	69
10.2.1 zapata-auto.conf	69
10.2.2 modprobe.conf	70
10.3 Caller ID (CID) using ZAP device.....	70
10.3.1 zapata.conf	71
10.3.2 zapata-auto.conf	71
10.4 Rebuild Zaptel Driver.....	72
10.4.1 IF ALL ELSE FAIL.....	74
10.5 Sipura SPA3000 as a PSTN Interface.....	75
10.5.1 Log in to SPA3000	75
10.5.2 Change the settings	75
10.5.3 Add SIP Trunk.....	77
10.5.4 SPA3000 as an outbound PSTN Trunk.....	78
11 PSTN to VOIP Gateway	79
12 DISA	80
12.1 DISA - Hand Coding Method.....	80
12.2 DISA – Using freePBX	81
13 Setting Up a BOL SIPPhone	82
Profile Tab.....	82
Audio & Video Tab	82
Network Tab.....	83
STUN Tab	83
Call Forward.....	83
14 Operator Panel	84
14.1 Flash Operator Panel (FOP)	84
14.1.1 Setting the Admin Password.....	85
14.1.2 Hang-up a Call	85
14.1.3 Transfer a Call	85
14.1.4 Initiate a Call	85
14.1.5 Barge in or Create a Conference	85
14.2 HUDLite	86
14.2.1 Setting up HUD Manager.....	86
14.2.2 Setting Up HUDLite.....	88
14.2.3 What if it does not work?	90
15 Call parking and transfer	91
15.1 Call Transfer - Managed	91
features.conf	91
extensions.conf	91
15.1.1 How is it done?	92
15.2 Call Transfer – Blind.....	92

15.3 Put a Call On Hold.....	92
15.4 Call Pickup	92
15.5 Filter Your Incoming Calls - Only Accept Known Calls	93
16 MeetMe – Teleconference	95
meetme.conf	95
meetme_additional.conf.....	95
16.1 Create Conference	95
16.1.1 /var/spool/asterisk/meetme	97
16.2 Invalid Conference number	97
17 Voicemail	98
17.1 Enabling Email Notification	98
17.1.1 Installing sendmail	98
17.1.2 /etc/hosts.....	99
17.2 Use Postfix Instead of Sendmail	100
17.3 How to Retrieve Voicemail from External Phone	101
17.3.1 Method 1	101
17.3.2 Method 2	101
17.4 How to Change Voice Mail Greetings	102
18 Fax to E-Mail.....	103
18.1 Installing Fax for TRIXBOX	103
19 Weather Reports	104
19.1 Configure Weather Report – On Demand.....	104
19.1.1 Create Extension codes.....	105
19.2 Configure Weather Report – Background Method.....	105
19.2.1 Create Extension codes.....	106
19.3 Now put it in your Digital Receptionist.....	107
20 Remote Management	108
20.1 httpd.conf.....	108
21 How to set up a Remote Extension.....	109
21.1 Create a new extension	109
21.2 IAX.Conf.....	110
21.3 Sip_Nat.Conf.....	111
21.4 Setting up an IAX Softphone.....	111
21.4.1 First the general options	111
21.4.2 Accounts options.....	112
22 Calls Recordings	113
22.1 Record all calls by default	113
22.2 Record calls on demand.	113
23 STUN Servers.....	114
24 Dialing through MS Outlook®	114
24.1 Download AstTapi	114
24.2 Install AstTapi.....	115
24.3 Configure Outlook Address Book.....	115
25 Speed Dial	117
25.1 The Dumb-Me method	117
25.2 TRIXBOX's method	118
26 How to use Window Messenger® 5.x	119
27 How to interconnect 2 boxes.....	121
27.1 Method 1 - With the Peer Asterisk box as Extensions.....	121
27.2 Method 2 - In a Peer/User Arrangement.....	122
27.3 Method 3 – Friend/Friend Arrangement.....	123
27.4 Thinking of more than 2 boxes?.....	124
27.5 Redirect Incoming in Box A to Box B	124
28 Customised Voice	126
28.1 Customising Voice Prompts	126
28.2 Installing Other Languages	127
28.3 Converting WAV files	127
28.3.1 Converting WAV to GSM	127
28.3.2 Converting WAV to SLN	128
29 MOH (Music On Hold).....	129

29.1 Customising MOH	129
29.2 MOH for different trunks	130
29.3 Streaming Music on Hold	131
30 Backup and restore	133
30.1 Schedule a Backup	133
30.2 To Restore a Backup.....	134
31 SugarCRM configuration	135
32 USB Phone Support	137
33 Codec (Coder Decoder)	138
33.1 Installing G729 and G723 CODECs.....	139
34 PSTN Trunk – Special Handling	141
34.1 Dial 0 for Outside Line.....	141
34.1.1 Using Zaptel Card Method.....	141
34.1.2 Using SPA3K Method	141
34.2 Giving Priority to Emergency Numbers (E.g. 000).....	142
35 Port Forwarding and QOS – Routers.....	143
35.1 Port Forwarding.....	143
35.2 QOS – Routers.....	143
36 Updating TRIXBOX	145
37 Tools	148
37.1 Webmin	148
37.1.1 To update WebMin.....	149
37.2 Putty	149
37.3 WinSCP	149
38 My TRIXBOX® PBX Network	150
39 Trouble Shooting	151
39.1 Asterisk Feature Codes not working	151
39.2 Asterisk Drops Calls after a few seconds	151
39.3 Pennytel Asterisk Problem	151
39.4 Unable to receive Incoming Calls.....	151
39.5 Sipura SPA3000 – Not hanging up after PSTN Calls	151
39.6 Trixbox 1.2 will not play nicely.....	152
Appendix A	153
A.1 Asterisk Feature Codes	153
A.2 Asterisk CLI commands	154
A.2.1 General commands	154
A.2.2 AGI Commands.....	155
A.2.3 Database Handling	155
A.2.4 IAX Channel Commands.....	155
A.2.5 SIP Channel commands.....	155
A.2.6 Server management	155
A.3 Asterisk Special Extensions.....	156
A.4 Asterisk Common Variables	156
A.5 Indications for Various Countries.....	158
Appendix B	167
B.1 – Australian VSPs (trunks).....	167
B.1.1 BBPglobal.....	167
B.1.2 Engin BYO.....	167
B.1.3 Faktortel (IAX)	168
B.1.4 iPrimus.....	168
B.1.5 Koala.....	169
B.1.6 MyNetFone	169
B.1.7 Nehos (IAX)	170
B.1.8 Nodephone	170
B.1.9 SipMe.....	170
B.1.10 TelephoneGlobal	171
B.2 – New Zealand VSPs (trunks)	172
B.2.1 iTalk	172
B.2.2 Double Zero 64 (IAX).....	172
B.3 – Other VSPs (trunks).....	173

B.3.1 Broadvoice.....	173
B.3.2 FWD aka Pulver(IAX)	173
B.3.3 SipBroker.....	173
B.3.4 SipPhone	174
B.3.5 VOIP Buster.....	174
B.4 Related Links – (for vsp TRUNK settings).....	175
B.4.1 Trunk and Route Settings.....	175
Appendix C.....	176
C.1 Voice for Asterisk Messages	176
C.2 Australian Voice and Language Set for Asterisk	177
C.3 More Australian Voice for Asterisk	177
Appendix D (ATAs and Routers).....	178
D.1 Alcatel SpeedTouch 510/530 ADSL Router	178
D.2 Billion 7402 VL.....	178
D.3 Linksys PAP2 and Sipura SPA2000.....	179
Appendix E (Users' Contributions).....	181
E.1 Suggestions for Dial Plan and configuration.....	181
E.1.1 Astratel	181
E.1.2 Further .conf modifications by Colin	182
E.2 Multiple TDM400P Installation	184
E.3 Eliminating Echo Problems in SPA-3000	186
E.4 Immediate Answer of PSTN Calls	188
E.5 Rebuild Zaptel Driver (Users' Suggestions)	188
E.5.1 As experienced by marner – a Whirlpool Forum participant.	188
E.5.2 As suggested by Rob Thomas (the FreePBX guru).....	188
E.6 Zaptel card configuration for Australia	189
E.7 RAID1, LVM (logical volumes) - howto	190
Appendix F (Publications and References).....	196
F.1 DUNDi – a guide to a Simple Setup.....	196
F.2 The future of Telephony by O'Reilly Publishing	196
F.3 Other Asterisk Forums, Tutorials and Wikis	196
F.4 Voice over IP – Per call bandwidth consumption.....	196
Notes.....	197
What version of Asterisk is installed?.....	197
A2Billing.....	197
Delete Call Records from CDR	197
Extension Numbers to avoid using.....	197
sip_nat.conf	198
Trunks and Outbound Routes – a circus of confusion.....	199
Outgoing Route Dial Pattern	199
Trunk Outgoing Dial Rules.....	199
Adding the missing Maintenance Modules to freePBX 2.1.0	200
Bugs Reports	201
Missing Zaptel support	201
Missing /var/spool/asterisk/meetme	201
Editing email notification message.....	201
FreePBX 2.1.2 caused Asterisk not to load	201
Trixbox 1.2 distribution major bug	203
Asterisk re-load issue	203
Are you still having problem?	203
Hangs on shutdown.....	204
Cheat Sheet.....	205
Exploit Alerts.....	206
Live Case Study	207
Bibliography	208
Changes Made since last Update.....	209

FOREWORD

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. Since I am putting a lot of efforts into providing personal assistance to those who have sent me donations, the volume of email I received has become unmanageable. . 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 am not an expert on Asterisk and therefore I do not have all the answers. I ask those requiring assistance to post their questions in the Whirlpool Forum or Trixbox 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)
- 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, and 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, and 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 Oztell (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 comparable to your mobile phone, not quite but close to your existing PSTN calls and be 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 notwithstanding, 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

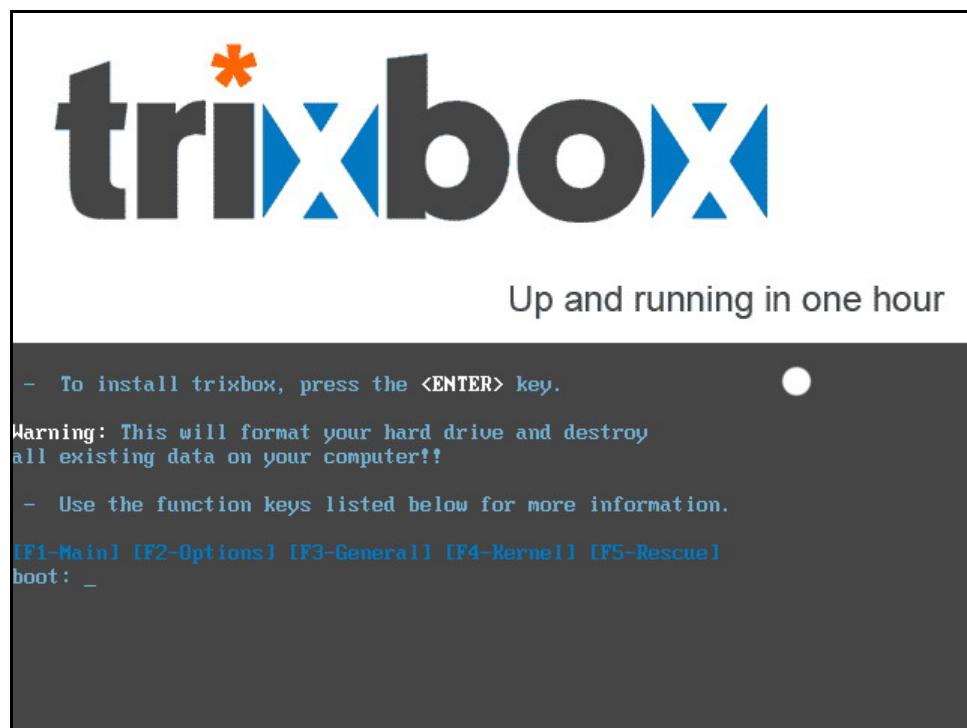
3 INSTALLATION

(The dumb-me way)

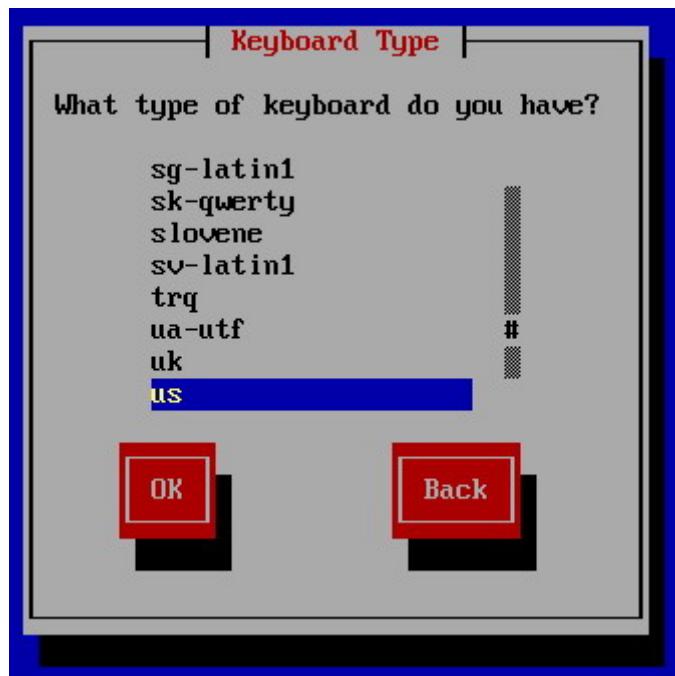
Download the ISO from here

<http://www.trixbox.org/modules/smartsession/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



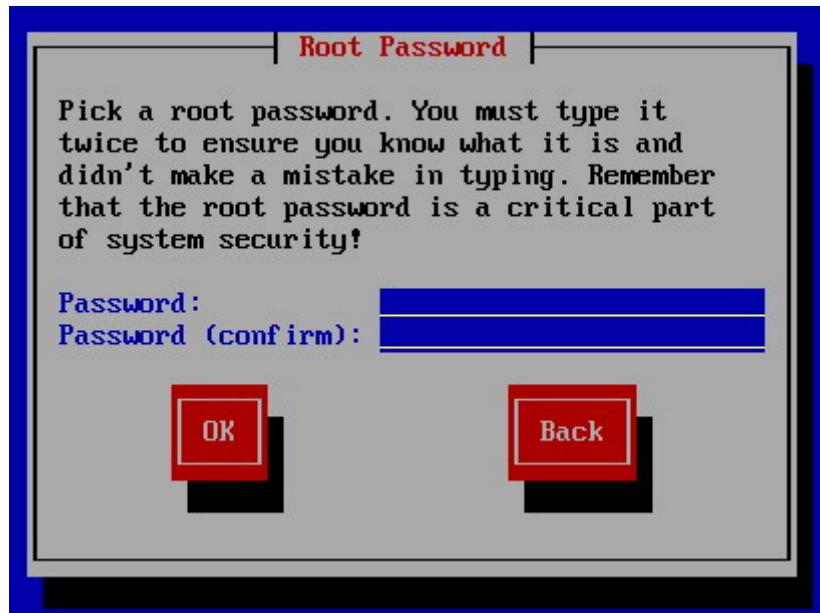
- Press the Enter key to start the installation.
- After initial system detection, you will be asked to select the type of keyboard that you will be using. Use your cursor keys to navigate the keyboard types and use the Tab key to move the focus to the **OK** or **Back** option. Select the appropriate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



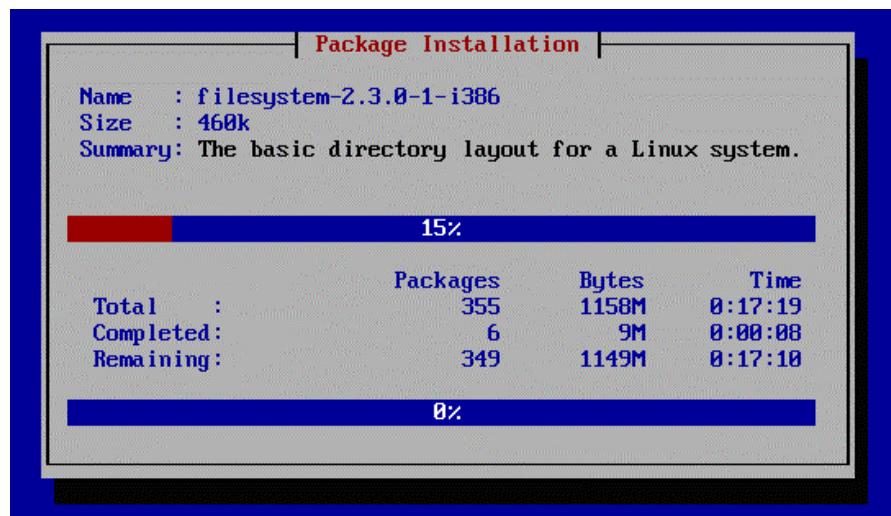
- After system hardware probing is completed, you will be asked to select the Time Zone you are in.



- 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 to secure your installation.

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 [UPDATING 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

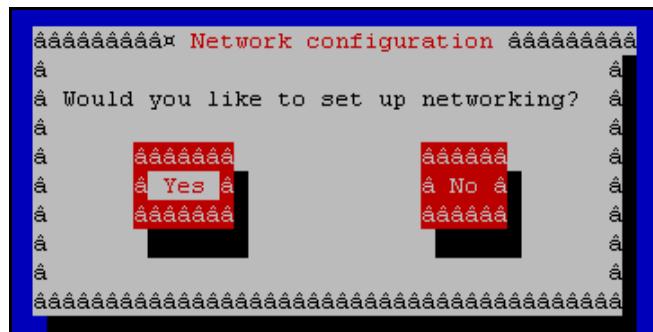
trixbox - HELP

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)

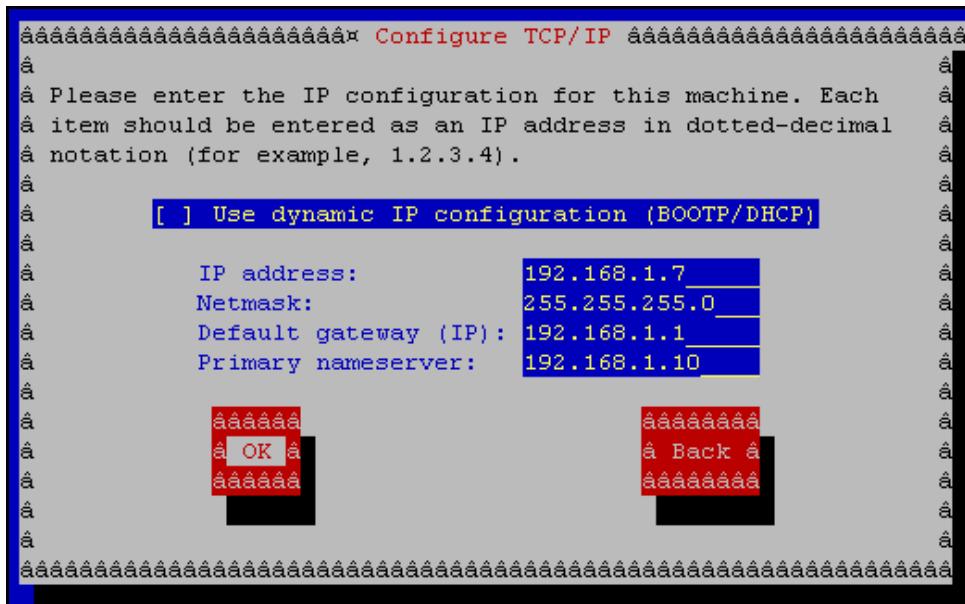
Change Asterisk IP address from DHCP to Static. At the command prompt enter:

Netconfig



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

You will then see the following screen.



Use the Tab key to cycle through the fields. Enter the IP address that is to be allocated to the Asterisk box, the Netmask (subnet mask), Default Gateway and Primary nameserve as per the example above. In my example above, I used my existing network regime.

- **In the IP address field**, enter an IP address TRIXBOX making sure it is within your network range eg: 192.168.1.7.
- **Netmask** is normally 255.255.255.0 unless your network has different Network mask.
- **Default gateway IP** is the address of your router. In my case, my router address is 192.168.1.1
- **Primary nameserver** is the address of your Name Server, usually your domain server if you have a network domainserver but if you are running workgroup, use the address of your default gateway eg: 192.168.1.1

Once done, select OK.

Those are the initial inputs that require immediate attention. Others, such as Admin password etc, are optional although it is recommended that you change them. Once done, log off Linux and reboot.

```
amportal stop  
Shutdown -r now
```

Asterisk will now start with the new IP address.

3.2 CONFIGURE TRIXBOX USING 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.



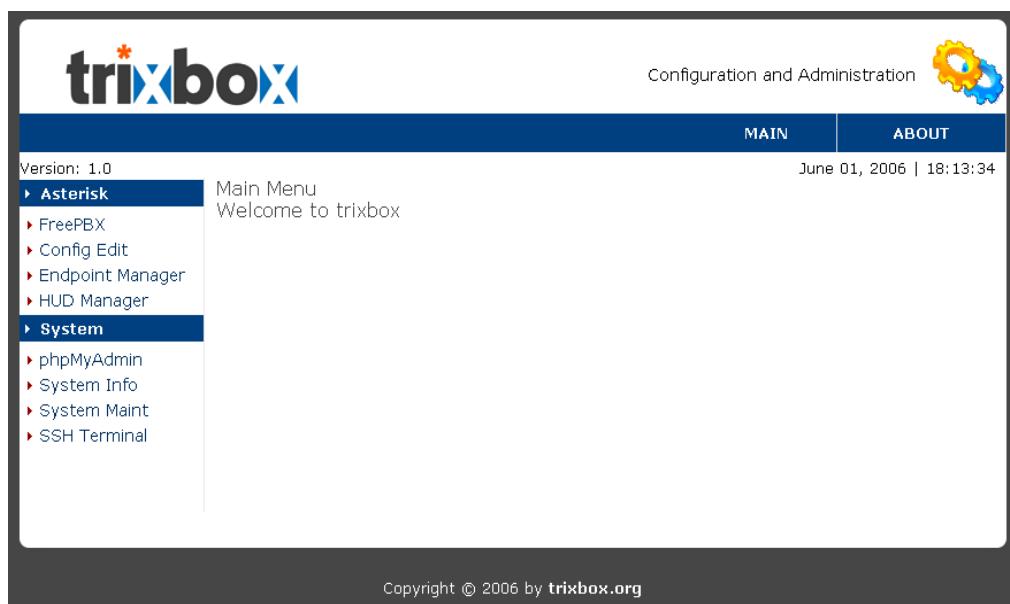
Main Welcome Screen

3.2.1 Log in to System Administration

Note

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,



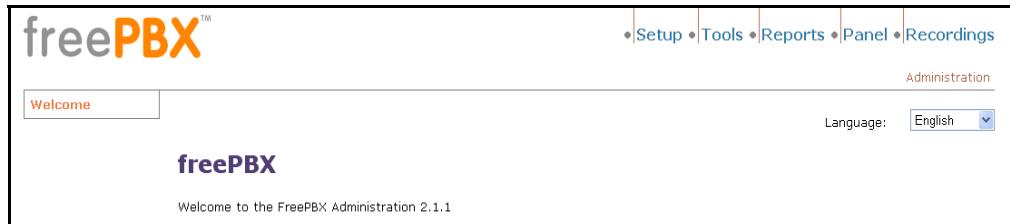
Main Configuration Screen

3.2.2 Configuring Trixbox using freePBX

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

From now on, we will be configuring Trixbox using freePBX web GUI.

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



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, and 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™

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

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™

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

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:

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Module Administration (online)

Available Module Updates (online)

Module	Version	Type	Category
CIN Sets (cincsets)	1.0.6	setup	Module
Recordings (recordings)	2.2	setup	Module

Download and Update selected Submit

Modules Available (online)

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

Download and Install selected Submit

Enabled Modules

Module	Version	Type	Category
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.

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

Module Administration (online)

Enabled Modules

Module	Version	Type	Category
Gabcast (gabcast)	1.0	setup	Addon
Asterisk API (manager)	1.0.2	tool	Basic
Core (core)	1.1	setup	Basic
Feature_Code_Admin (featurecodeadmin)	1.0	setup	Basic
Follow_Me (findmefollow)	1.2.5	setup	Basic

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.

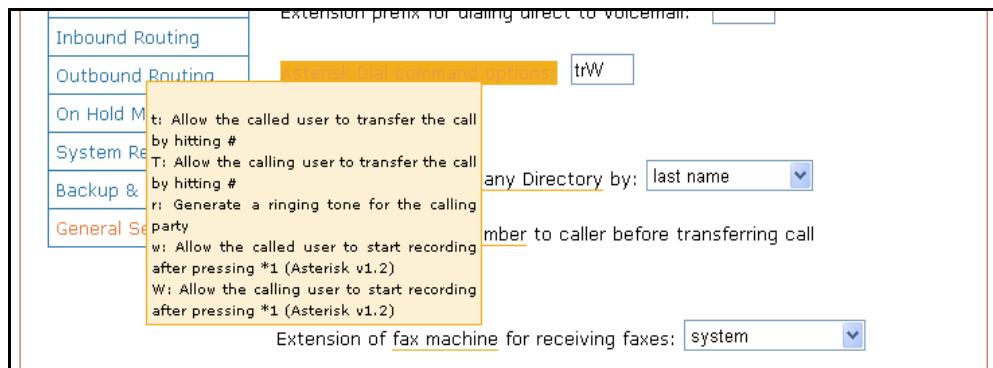
I have set mine to **Unavailable**.

The screenshot shows the 'General Settings' page of the freePBX web interface. The left sidebar lists various configuration categories. The main content area is divided into several sections:

- Dialing Options**: Asterisk Dial command options: `ltn`, Asterisk Outbound Dial command options: `r`.
- Voicemail**: Number of seconds to ring phones before sending callers to voicemail: `40`, Extension prefix for dialing direct to voicemail: `*`, Direct Dial to Voicemail message type: `Default`, Use gain when recording the voicemail message (optional): `0`.
- Company Directory**: Find users in the Company Directory by: `first name`, Play extension number to caller before transferring call.
- Fax Machine**: Extension of fax machine for receiving faxes: `system`, Email address to have faxes emailed to: `bensharif@gmail.com`, Email address that faxes appear to come from: `freepbx@gmail.com`.
- International Settings**: Country Indications: `Australia`, Security Settings: Allow Anonymous Inbound SIP Calls? `yes`.

At the bottom right is a `Submit Changes` button.

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 but for now, lets leave it alone:

Options:

A(x)	Play an announcement to the called party, using 'x' as the file
C	Reset the CDR for this call
d	Allow the calling user to dial a 1 digit extension while waiting for a call to be answered. Exit to that extension if it exists in the current context, or the context defined in the EXITCONTEXT variable, if it exists.
D([called]):[calling])	Send the specified DTMF strings *after* the called party has answered, but before the call gets bridged. The 'called' DTMF string is sent to the called party, and the 'calling' DTMF string is sent to the calling party. Both parameters can be used alone.
f	Force the callerid of the *calling* channel to be set as the extension associated with the channel using a dialplan 'hint'. For example, some PSTNs do not allow CallerID to be set to anything other than the number assigned to the caller.
g	Proceed with dialplan execution at the current extension if the destination channel hangs up
G(context^exten^pri)	If the call is answered, transfer both parties to the specified priority. Optionally, an extension, or extension and context may be specified. Otherwise, the current extension is used.

h	Allow the called party to hang up by sending the '*' DTMF digit.
H	Allow the calling party to hang up by hitting the '*' DTMF digit.
i	Jump to priority n+101 if all of the requested channels were busy.
L(x[:y][:z])	<p>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:</p> <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 called party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in features.conf (default *1 in Asterisk v1.2)
W	Allow the calling party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in features.conf (default *1 in Asterisk v1.2)

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	<h3>Add SIP Extension</h3> <p>Add Extension</p> <p>Extension Number: <input type="text" value="2001"/></p> <p>Display Name: <input type="text" value="Ben Sharif"/></p> <p>Extension Options</p> <p>Direct DID: <input type="text"/></p> <p>DID Alert Info: <input type="text"/></p> <p>Outbound CID: <input type="text"/></p> <p>Emergency CID: <input type="text"/></p> <p>Record Incoming: <input type="button" value="On Demand"/></p> <p>Record Outgoing: <input type="button" value="On Demand"/></p> <p>Device Options</p> <p>secret <input type="text" value="2001"/></p> <p>dtmfmode <input type="text" value="rfc2833"/></p> <p>Voicemail & Directory: <input type="button" value="Disabled"/></p> <p style="text-align: center;"><input type="button" value="Submit"/></p>	Add Extension Ben Sharif <2001> Norsurya Sharif <2002>
--	--	--

freePBX™

Version 2.1.1 on 192.168.1.29

Add Extension Screen

Click on the red bar on the top of the screen every time 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	<h3>Add SIP Extension</h3> <p>Add Extension</p> <p>Extension Number: <input type="text"/></p> <p>Display Name: <input type="text"/></p> <p>Extension Options</p> <p>Direct DID: <input type="text"/></p> <p>DID Alert Info: <input type="text"/></p> <p>Outbound CID: <input type="text"/></p>	Add Extension 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.

Note:

*If you want this extension to be a remote extension, you will need to edit the extension and change the entry under **Device options**. Nat: Never to Nat: yes.*

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:

The screenshot shows the freePBX interface under the 'Setup' tab. On the left, a sidebar menu includes 'Follow Me' which is currently selected. The main content area is titled 'Follow Me' and contains the sub-instruction 'Choose a user/extension:'. To the right of the main area is a vertical list of extensions, each with a status indicator (e.g., 'add') and a link to its configuration page.

Test 210 <210> add
Test 220 <220> add
Rohani Sharif <2000> add
Ben Sharif <2001> add
Norsurya Sharif <2002> add
Rossli Sharif <2003> add
Adam Sharif <2004> add
Ben Sharif - Remote <2005> add
Ben HP <2010> add
Klinik Mesra <2060> add
Malacca IPBX <60600> add

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**’

The screenshot shows the 'Edit Extension 2001' configuration page. The left sidebar still shows 'Follow Me' as selected. The main form is titled 'Follow Me: 2001' and contains several configuration fields: 'ring strategy' set to 'hunt', 'extension list' containing '2001 0404044444#', 'CID name prefix', 'ring time (max 60 sec)' set to '15', and 'announcement' set to 'None'. Below these are sections for 'Edit Follow Me' and 'Destination if no answer', which lists various options like 'Follow Me: <2001>', 'Core: voicemail box 2001', etc. A 'Submit Changes' button is at the bottom.

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:

The screenshot shows the freePBX Ring Group configuration page for Ring Group 10. The left sidebar lists various system settings like Administrators, Callback, Conferences, DISA, Digital Receptionist, Extensions, Feature Codes, Follow Me, Gacast, General Settings, Inbound Routes, Misc Destinations, On Hold Music, Outbound Routes, PIN Sets, Paging and Intercom, Queues, Ring Groups (which is currently selected), System Recordings, Time Conditions, Trunks, and VoicePulse. The main content area is titled "Ring Group: 10". It includes fields for "group description" (AllPhones), "ring strategy" (ringall), and an "extension list" containing 2000, 2001, 2002, and 0400123456#. There are also fields for "CID name prefix", "ring time (max 60 sec)" (40), "announcement" (None), and "Alert Info". Under "Destination if no answer", the "Ring Groups" option is selected with "AllPhones <10>". At the bottom, there is a "Submit Changes" button. A status bar at the bottom right indicates "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 and make calls to the outside worlds.

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

This is a simplified guide to give the budding TRIXBOX 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 dialled 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 dialled 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).

General Setting (This applies to all the trunks):

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

Maximum channels: Leave blank for maximum.

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

4.3.1 Pennytel

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

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

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

Pattern	Explanation
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 dialling Australian Mobile phone. The 61 is the Australian Country code that will be added to the mobile phone number.

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

```
601XXXXXXXX  
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: **pennytelsip**

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=sip.pennytel.com  
insecure=very  
secret= Password  
type=peer  
username=8880XXXX
```

Incoming Settings

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

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

```
canreinvite=no  
context=from-trunk  
fromuser=888XXXXXX  
qualify=no  
secret= Password  
type=user  
username=888XXXXXX
```

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

We need various patterns here:

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

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

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

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

Outgoing Settings

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

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

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

allow=ulaw&alaw
canredirect=no
disallow=all
dtmfmode=rfc2833
fromuser=899060xxxxx
host= sip2.syd.net2max.com

```
insecure=very  
secret=Password  
type=peer  
username=899060XXXXX
```

Incoming Settings

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

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

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

Registration

In the registration field, enter the following:

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

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

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: **oztelliiax** (as an example)

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

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

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

Incoming Settings

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

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

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

Registration

In the registration field, enter the following:

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

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

4.3.4 Astratel

To make an international call, Astratel requires the numbers dialled 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 dialling 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:

The screenshot shows the 'Add SIP Trunk' configuration page in freePBX. The left sidebar lists various system settings, and the main area is divided into several sections: General Settings, Outgoing Dial Rules, Outgoing Settings, Incoming Settings, and Registration. The 'Outgoing Dial Rules' section contains a list of dial rules and a 'Clean & Remove duplicates' button. The 'Outgoing Settings' section includes a 'Trunk Name' field set to 'SipPennytel' and a large text area for 'PEER Details' containing configuration parameters. The 'Incoming Settings' section shows a 'USER Context' field set to 'Pennytel-In' and a 'USER Details' text area with similar configuration parameters. The 'Registration' section displays a 'Register String' field with the value '8882593347:xxxxxx@sip.pennytel.com/88825'. At the bottom right is a 'Submit Changes' button.

* 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 with the title 'freePBX™'. On the left, there is a vertical sidebar menu with various options like Administrators, Callback, Conferences, DISA, Digital Receptionist, Extensions, Feature Codes, Follow Me, Gabcast, General Settings, Inbound Routes (which is highlighted in orange), Misc Destinations, On Hold Music, Outbound Routes, PIN Sets, Paging and Intercom, Queues, Ring Groups, System Recordings, Time Conditions, Trunks, and VoicePulse. The main content area has a header 'Add Incoming Route' with a 'Setup' link and a 'Add Incoming Route' button. The form fields include:

- Add Incoming Route**:
 - 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> (radio button selected)
 - Core: Rohani Sharif <2000> (dropdown menu)
 - Custom App: [empty]

At the bottom right of the form is a 'Submit' button. At the very bottom of the page, there is a footer with the 'freePBX™' logo and the text 'Version 2.1.1 on 192.168.1.29'.

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 with a red banner at the top stating "You have made changes - when finished, click here to APPLY them". Below the banner, the title "Add Incoming Route" is displayed. On the left, there is a sidebar menu with links like Administrators, Callback, Conferences, DISA, Digital Receptionist, Extensions, Feature Codes, Follow Me, and Gabcast. The main form area contains fields for DID Number (set to 8882593347), Caller ID Number (left blank), Zaptel Channel (left blank), and Fax Handling (left blank). A note "OR" is present between the DID Number and Caller ID Number fields. To the right of the form, there is a "Setup" link and a box containing the text "Add Incoming Route" and "8882593347 / any CID".

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 calls may not work.

5.2 TIME CONDITIONS

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

The screenshot shows the freePBX web interface with the title 'Add Time Condition'. The left sidebar contains a list of administrative functions: Administrators, Callback, Conferences, DISA, Digital Receptionist, Extensions, Feature Codes, Follow Me, Gacast, General Settings, Inbound Routes, Misc Destinations, On Hold Music, Outbound Routes, PIN Sets, Paging and Intercom, Queues, Ring Groups, System Recordings, Time Conditions (which is highlighted), Trunks, and VoicePulse. The main content area has several sections: 'Time Condition name:' with a text input field; 'Time to match:' with dropdown menus for 'Time to start' and 'Time to finish'; and sections for 'Week Day Start', 'Month Day Start', 'Month Day Finish', 'Month Start', and 'Month Finish'. Below these are two sections for 'Destination if time matches:' and 'Destination if time does not match:', each containing a list of destination options: DISA (PSTN2VoIP), Queues (General <100>), Ring Groups (AllPhones <10>), Misc Destinations (VoiceMail), IVR (Main Menu), Conferences (General Conference <8200>), Core (Rohani Sharif <2000>), and Custom App. At the bottom is a 'Submit Changes' button.

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.

Simple Time Condition

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

- Add a new time condition by selecting **Setup -> Time Condition -> Add time condition**.
- In the **Time condition name** field, give it a name eg: **DayNight**

- Under **Time to match**, do the following:
 - Time to start: 08:00
 - Time to finish: 20:00
 - Week Day start: Monday
 - Week Day finish: Sunday
- Under **Destination if time matches**, select IVR (or Core extension or ring group)
- Under **Destination if time does not match**, select Core mailbox: eg voicemail box 2001
- Submit and click on the **red bar** on top of the page.

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

Complex Time Condition

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

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

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

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

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

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

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 for a route named 'Pennytel'. The left sidebar lists various system settings, and the top navigation bar includes 'Setup', 'Tools', 'Reports', 'Panel', and 'Recordings'. The main form has sections for 'Route Name' (Pennytel), 'Route Password' (empty), 'PIN Set' (dropdown), 'Emergency Dialing' (checkbox), 'Dial Patterns' (list of numbers: 777, 84848810, 88801575, 88836804, 88897446, 17361XX, 60386572XX, 888ZXX, with a 'Clean & Remove duplicates' button), 'Insert' (dropdown), 'Trunk Sequence' (list of trunks: SIP/SipPennytel, SIP/SipPennytel-2, with an 'Add' button), and a 'Submit Changes' button. A vertical list of existing routes is on the right.

Add Route
0 PSTN
1 SPA3K
2 Oztell
3 Pennytel
4 TeleGlobal
5 AstrateL
6 MyNetFone
7 Spantalk
8 VOIPBuster
9 Domestic
10 MobileAUST
11 International
12 12Number
13 13N1800Numbers
14 USFree

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.

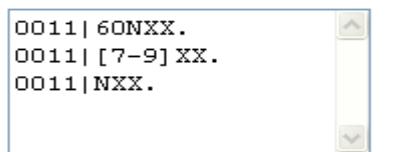
5.3.3 International

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 dialling 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 Sequent. 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

0	SIP/astratelsip	▼	X	↶
1	SIP/pennytelsip	▼	X	↶
		▼		
<input type="button" value="Add"/>				

Any number dialled 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]XXXXXXXXX	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]XXXXXXXXX	If you dial local number and not use the STD prefix, it will send to the trunk as is and the trunk will insert the 02 if required (this is a NSW pattern. If you are in Victoria you will need to modify the trunk).
ZXX.	This is the catchall dialling for numbers that does not start with a 0 such as 1300, 1800, 13 etc.

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



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

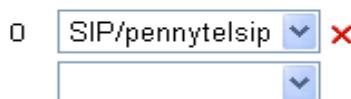
5.3.5 MobileAust

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

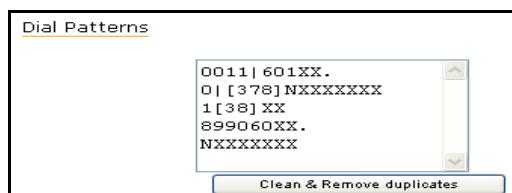


5.3.6 Oztellonly

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]NXXXXXXX	You will dial Australian telephone number as normal using 03, 02 prefix but the route will strip the leading 0 before it is sent to the trunk. If the trunk requires the 0, the trunk will insert it.
1[38]XX.	This is for dialling 1300, 1800 and 13 numbers.
899060XX.	For dialling Oztell inter Oztell users
NXXXXXXX	Catchall for dialling a local number that doesn't start with 1

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



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

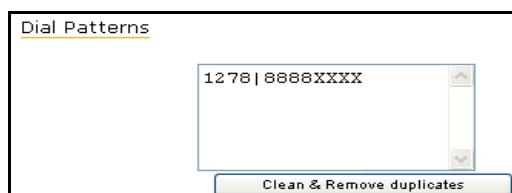
Trunk Sequence



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

The screenshot shows the 'Edit ENUM Trunk' configuration page in freePBX. The left sidebar lists various system settings like Administrators, Callback, Conferences, DISA, etc. The main panel has a title 'Edit ENUM Trunk' and a note 'In use by 1 route'. It includes sections for 'General Settings' (Outbound Caller ID set to "Ben Sharif" <612473641E>, Maximum channels empty), 'Outgoing Dial Rules' (Dial Rules list containing '612+NXXXXXXX', '61+NXXXXXXXXX', and 'ZXXXXXXXXX.', with a 'Clean & Remove duplicates' button), 'Dial rules wizards' (set to '(pick one)'), and 'Outbound Dial Prefix' (empty). A 'Submit Changes' button is at the bottom right.

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]NX. 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|NX. 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|NX. This enables me to dial NSW local number without having to dial 02.

Dial Patterns

1640011 NX.
1640 [378]NX.
164 NX.

Clean & Remove duplicates

Insert: Pick pre-defined patterns

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.

Trunk Sequence

0	ENUM/
1	SIP/SipPennytel

Add

Submit Changes

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 OUTGOING CALLS

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

There is no easy way of imposing this restriction through the freePBX GUI. The only way that can be done currently is to introduce special scripts 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 TRIXBOX, open `extensions_additional.conf` and locate the **[outbound-allroutes]** section of the file. You will notice something like my example below.

```
[outbound-allroutes]
include => outbound-allroutes-custom
include => outrt-001-PSTN
include => outrt-002-SPA3K
include => outrt-003-Oztell
include => outrt-004-Pennytel
include => outrt-005-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 and will get back to this a little later.

Alternatively you can paste it directly to `extensions_custom.conf`. Do not do this if Trixbox is active as it may cause some problem while people using it.

3. Through **Config Edit** of TRIXBOX, open `extensions_custom.conf` and create a section like my example below. You can get the codes from **[from-internal-additional]** of `extensions_additional.conf`. So will have to do some juggling because you can't cut and paste this from another `.conf` file without having to close one (you can cut and paste from here if you like).

```

[from-restricted]
;
; These are all the applications that you will require
;
include => app-cf-busy-off
include => app-cf-busy-off-any
include => app-cf-busy-on
include => app-cf-off
include => app-cf-off-any
include => app-cf-on
include => app-cf-unavailable-off
include => app-cf-unavailable-on
include => app-calltrace
include => app-callwaiting-cwoff
include => app-callwaiting-cwon
include => app-dialvm
include => app-directory
include => app-dnd-off
include => app-dnd-on
include => app-echo-test
include => app-recordings
include => app-speakextennum
include => app-speakingclock
include => app-userlogonoff
include => app-zapbarge
include => app-vmmain
include => ext-group
include => ext-fax
include => ext-meetme
include => ext-findmefollow
include => ext-paging
include => ext-queues
include => ext-test
include => ext-local
include => parkedcalls
;
; ##### OutBound Routes #####
; # Below are all restricted routes #
; #####
;
```

Now copy your [outbound-allroutes] that was saved to the text file directly underneath the last line of the above context. Remember to delete the [outbound-allroutes] label. Unfortunately you can't cut and paste my example because they are my routes NOT yours.

Your final handiwork will look something like the example below:

```

[from-restricted]
;
; These are all the applications that you will require
;
include => app-cf-busy-off
include => app-cf-busy-off-any
include => app-cf-busy-on
include => app-cf-off
include => app-cf-off-any
include => app-cf-on
include => app-cf-unavailable-off
include => app-cf-unavailable-on
include => app-calltrace
```

```

include => app-callwaiting-cwoff
include => app-callwaiting-cwon
include => app-dialvm
include => app-directory
include => app-dnd-off
include => app-dnd-on
include => app-echo-test
include => app-recordings
include => app-speakextennum
include => app-speakingclock
include => app-userlogonoff
include => app-zapbarge
include => app-vmmain
include => ext-group
include => ext-fax
include => ext-meetme
include => ext-findmefollow
include => ext-paging
include => ext-queues
include => ext-test
include => ext-local
include => parkedcalls
;
; ##### OutBound Routes #####
; # Below are all restricted routes #
; ##### Outbound-Restricted-Routes #####
;
; new outbound-restricted-routes
include => outbound-allroutes-custom
include => outrt-001-PSTN
include => outrt-002-SPA3K
include => outrt-003-Oztell
include => outrt-004-Pennytel
include => outrt-005-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-restricted-routes

Now add the following codes underneath that to finish it off.
;
exten => h,1,Hangup
exten => s,1,Macro(hangupcall)
exten => h,1,Macro(hangupcall)

```

4. You may now comment out the outbound routes you do not want the restricted extensions to use.

E.g. If you comment out `include => outrt-001-PSTN`, the restricted extension cannot use PSTN.

5. After you have done that, you need to go back to FreePBX and edit the extension that you want to restrict by change the context to **from-restricted** instead of from-internal.

Now go enjoy yourself with it.

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 'System Recordings' page in the freePBX web interface. On the left is a sidebar with links like Administrators, Callback, Conferences, DISA, Digital Receptionist, Extensions, Feature Codes, Follow Me, Gabcast, General Settings, Inbound Routes, and Misc Destinations. At the top right are links for Setup, Tools, Reports, Panel, and Recordings. Below the header, there's a section titled 'Add Recording' with two steps: Step 1: Record or upload and Step 2: Name. Step 1 includes fields for extension number and wav file upload. Step 2 includes a name field and a 'Save' button. A sidebar on the right lists recordings: Add Recording, Local_Weather_Report, and No_Marketing.

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.

This screenshot is similar to the previous one but includes instructions for direct recording. It says 'Using your phone, dial *77 and speak the message you wish to record.' and 'After recording or uploading, dial *99 to listen to your recording.' The rest of the interface is identical to the first screenshot.

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 dialling *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 Set-up tab, click on the Digital Receptionist option and get the following screen.

The screenshot shows the freePBX interface under the 'Setup' tab. On the left, a sidebar lists various configuration options like Administrators, Callback, Conferences, DISA, Digital Receptionist (which is highlighted in orange), Extensions, and Feature Codes. The main content area is titled 'Digital Receptionist' and contains an 'Instructions' section with text about using the digital receptionist to make IVR's, Interactive Voice Response systems. It also includes a note about menu options and invalid button handling. On the right, there are two buttons: 'Add IVR' and 'MainMenu'. A 'Setup' link is located at the top right of the content area.

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

The screenshot shows the 'Edit Menu Unnamed' configuration screen. At the top, a message says 'You have made changes - when finished, click here to APPLY them'. The main area is titled 'Digital Receptionist' and 'Edit Menu Unnamed'. It features a 'Delete Digital Receptionist Unnamed' button. Below are several configuration fields: 'Change Name' set to 'Unnamed', 'Timeout' set to '10', 'Enable Directory' checked, 'Directory Context' set to 'default', 'Enable Direct Dial' checked, and 'Announcement' set to 'None'. There are three buttons at the bottom: 'Increase Options', 'Save', and 'Decrease Options'. Below these buttons is a group of radio buttons with dropdown menus: 'Ring Groups: AllPhones <10>', 'IVR: MainMenu', 'Core: Rohani Sharif <2000>', 'Time Conditions: Normal', 'DISA: PSTN-VoIP', and 'Custom App:'. The 'Save' button is highlighted in blue.

You then need to fill in the following fields:

Name (Change Name): MainMenu (*that's what I called my 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)

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/6140XXXXXX@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 screen.

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
```

7.3 MULTI-LANGUAGE IVR

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

See [Installing Other Languages](#) in the chapter referring to [Customised Voice](#) in later part of this document.

I am assuming that you already have your **MAIN IVR** in English set up for selecting various options. This IVR is known as IVR-2 in your extension_additional.conf file. Any subsequent IVR created will IVR-3, IVR-4 and so on.

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

Being the second IVR you have created, this IVR will be known as IVR-3 in your extension_additional.conf file. Any subsequent IVR created will be IVR-4 and so on.

In ***LanguageChoice***, we will have 3 options.

- Select 1 for Australian English
- Select 2 for Italian
- The third option is unannounced as it will be the time out option represented by the letter "t". This option is the option the IVR will defaulted to if no choice is made or if the caller's DTMF tone is not recognised by Trixbox.

In choice 1, select the Custom App: radio button and enter the following in the box:

Custom App

In choice 2, select the Custom App: radio button and enter the following in the box:

Custom App

The third choice, instead of a number, you will enter the letter "t" in the option box, select Custom App: radio button and enter the following:

Custom App

Next, you will need to do a little editing of the extensions_custom.conf file. You need to add the following towards the end of your extensions_custom.conf.

You will also need to create another IVR called ***MAINIVR_IT*** with options to be presented in Italian. This is similar to your main IVR except that it is presented to the caller in Italian. This IVR being the third that you have created will be referred to as IVR-4 in the extensions_additional.conf file.

```
[custom-language_au]
exten => s,1,Set(LANGUAGE()=au)
exten => s,n,Playback(vm-dialout)
exten => s,n,Goto(ivr-2,s,begin)

[custom-language_it]
exten => s,1,Set(LANGUAGE()=it)
exten => s,n,Playback(vm-dialout)
exten => s,n,Goto(ivr-4,s,begin)
```

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

Now you will need to change your inbound route to point to the ***LanguageChoice*** IVR. All calls will then be greeted by the LanguageChoice IVR and when the caller select 1 (or 2), the IVR will call the appropriate exten. If the caller selected 1, the ***[custom-language_au]*** will be selected and the language will be set to Australian English, an announcement in the appropriate language will be made and the caller will be sent to the ***MainIVR (or MainIVR_IT)*** for call options (you must also have recorded the call options announcement in the various languages).

7.3.1 Complimentary multi-language MOH

To take the multi-language IVR one step further, we can also have multi-language (or multi category) MOH to compliment the language.

If the language used is Italian, it is safe to assume that the caller is Italian and would much prefer to listen to Italian music instead of Chinese or Arabic music..

To do this you will need to have the appropriate MOH music in Trixbox.

Assuming that you already have the default music category that comes with Asterisk, you will need to create a new music category using freePBX On Hold Music option. Let us call this new category ***ItalianTune***.

See the chapter relating to [MOH](#) in the later chapter of this document.

Once you have created this category, upload a few MP3 Italian music/Tunes to it. If you want the tune to be randomised, click the Enable Random button and you are set to go.

You will need to modify the 2 scripts above to the following:

```
[custom-language_au]
exten => s,1,Set(LANGUAGE()=au)
exten => s,n, SetMusicOnHold(default)
exten => s,n,Playback(vm-dialout)
exten => s,n,Goto(ivr-2,s,begin)

[custom-language_it]
exten => s,1,Set(LANGUAGE()=it)
exten => s,n, SetMusicOnHold(ItalianTune)
exten => s,n,Playback(vm-dialout)
exten => s,n,Goto(ivr-4,s,begin)
```

You may have to restart asterisk after this (although it may not be necessary).

Now you will have multi language IVR as well as the complimentary MOH.

7.4 MULTI-TENANTS IVR

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

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

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

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

The main requirements are pretty straightforward:

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

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

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

The full case study can be found in the Appendix titled, [Live Case Study](#) at the end of this document.

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

PSTN Interface

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

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

IVR Prompts

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

- MedicalIVR (represented by ivr-3 in extensions_additional.conf)
- DentalIVR (represented by ivr-4 in extensions_additional.conf)
- PatholIVR (represented by ivr-5 in extensions_additional.conf)

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

Inbound Routings

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

Extensions

1. Extensions 2000 to 2004 allocated to GP

- 2000 for Receptionist/Clerk with voicemail
- 2001 for Doctor A
- 2002 for Doctor B
- 2003 for Surgery Room
- 2004 for Dispensary

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

2. Extensions 3000 to 3003 allocated to Dental

- 3000 for receptionist/clerk with voicemail

- 3001 for Dentist A
- 3002 for Dentist B
- 3003 Spare extension

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

3. Extensions 4000 to 4002 allocated to Pathology

- 4000 for receptionist/clerk with voicemail
- 4001 for Radiographer
- 4002 for Assistant

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

Outbound Calls Handling context

Create 3 different call handling contexts in the **extensions_custom.conf** configuration file.

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

The purpose of these contexts is to restrict extensions belonging to each tenant to be confined to telephone lines belonging to that particular tenant when making external calls. See [How to restrict outgoing calls](#) in the Calls Handling chapter earlier in this document.

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

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

8 GET UNDER THE BONNET

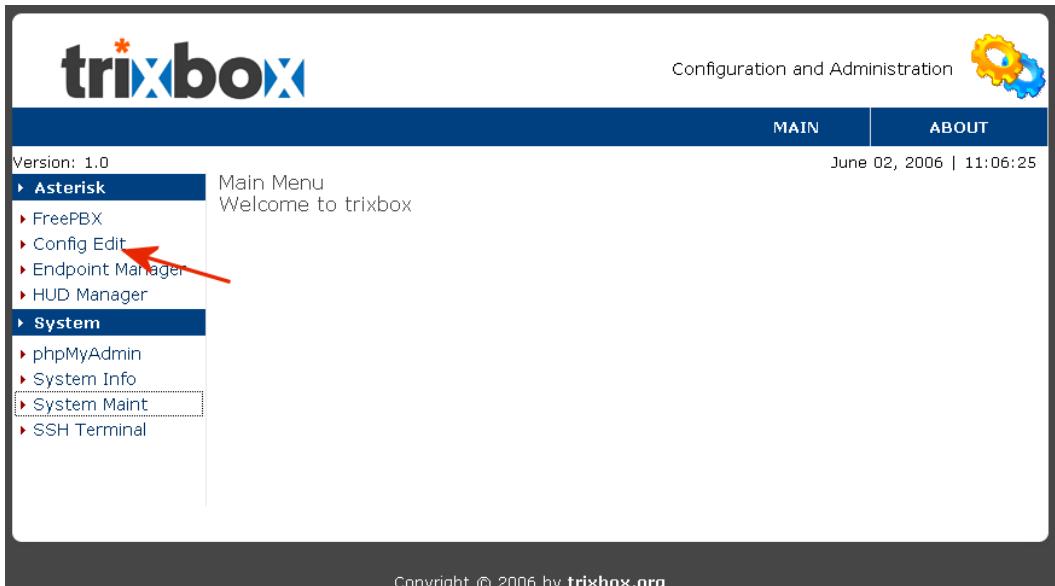
Note for home users

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 TRIXBOX 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 TRIXBOX 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.

phpconfig for Asterisk PBX				
/etc/asterisk	/var/www/html/panel	/etc	/tftpboot	Re-Read Configs
a2billing.conf agents.conf alarmreceiver.conf applications.conf asterisk.conf cdr_mysql.conf codecs.conf disa-1.conf				

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**, **features.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. We will look at them later on under [Interfacing Asterisk to PSTN](#)).

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 ; I add this in sip_custom.conf instead
allow=ulaw
allow=alaw
allow=gsm

; the 2 lines below I added them in sip_custom.conf although
; you can add them in sip.conf
defaultexpiry=600      ; **Engin users: include this if necessary
maxexpiry=3600        ; **Engin & BBP Global 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 dialled 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

; #, 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
```

Note:

*Every time you update freePBX, there is a good chance that changes or additions you have made in **sip.conf** will be overwritten. To avoid this, all additions I wish to make in **sip.conf** are made in **sip_custom.conf***

sip_custom.conf

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

defaultexpiry=600      ; **Engin users: include this if necessary
maxexpiry=3600         ; **Engin & BBP Global users: include this if necessary
```

8.1.2 iax.conf

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

```
[general]
delayreject=yes
jitterbuffer=yes ; I placed this line in my iax_custom.conf instead
mailboxdetail=yes ; I placed this line in my iax_custom.conf instead
```

Note:

*Every time you update freePBX, there is a good chance that changes or additions you have made in **iax.conf** will be overwritten. To avoid this, all additions I wish to make in **iax.conf** are made in **iax_custom.conf***

iax_custom.conf

```
; additions to iax.conf that would not get deleted if you
upgrade freepbx

;externip = <your fixed external IP> or (one or the other NOT both)
externhost = <your DynDNS name>
localnet = 192.168.1.0/255.255.255.0

language=au
delayreject = yes
jitterbuffer = yes
allow=g729
allow=ilbc
allow=g723
```

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 => el64.org
search => el64.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.

8.1.6 features.conf

The default features.conf included with trixbox is fairly minimal. It is recommended that you reflect the following in your features.conf file. The extra keywords will come in handy later on.

```

; Sample Parking configuration etc.
;
[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)

transferdigittimeout =>3 ; Number of seconds(in seconds)
courtesytone = beep ; to wait between digits when transferring a call
; Sound file to play to the parked caller
; when someone dials a parked call

xfersound = beep ; to indicate an attended transfer is complete
xferfailsound = beeperr ; to indicate a failed transfer
;adsipark = yes ; if you want ADSI parking announcements
;pickupexten = *8 ; Configure the pickup extension. Default is *8
featuredigittimeout = 800 ; Max time (ms) between digits for
; feature activation. Default is 500

[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

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.

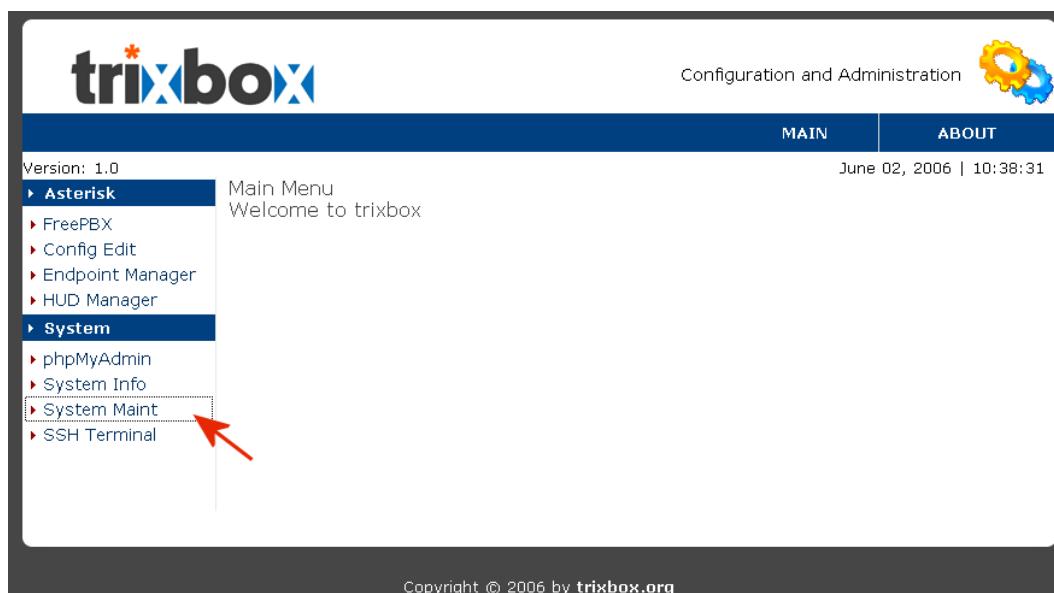
9.1 SYSTEM PROCESS STATUS

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.

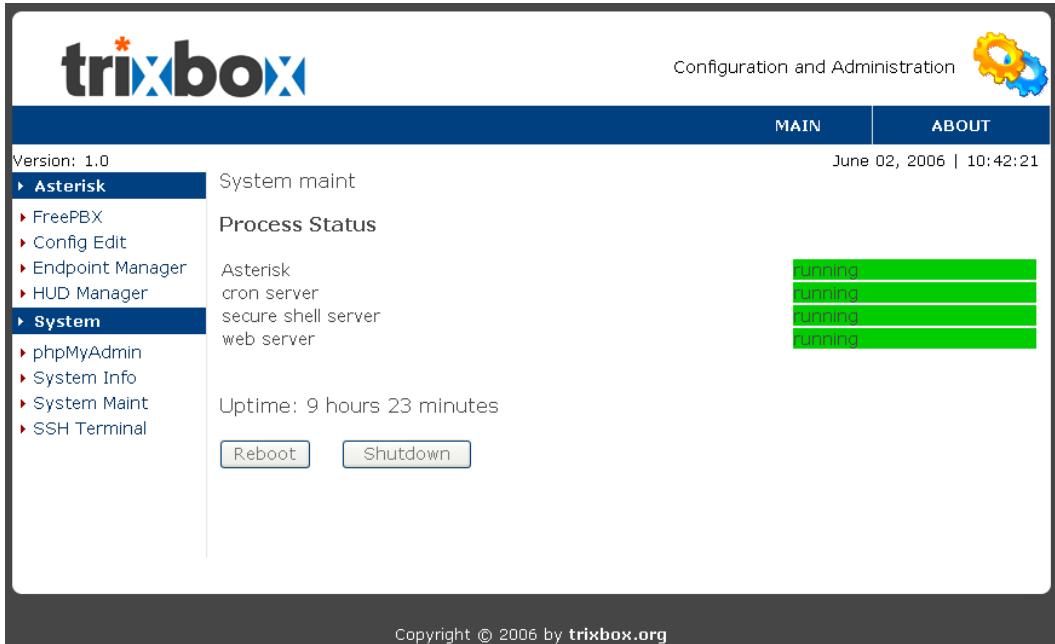
To do the initial checking, let us start with the System Maint option of TRIXBOX Configuration and Maintenance Menu.

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 devise, you should try to disable the related .conf files.

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

Next, 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	091334320sip	120	Request Sent
sip03.astraspip.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.sippone.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 Codec, Routings and Dialling rules errors.

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

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. This issue has been 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 your pots line in the **Caller ID field**
2. Enter **1** for **Maximum channels**
3. Set a dial rule you want for this trunk
4. Select an outbound dial prefix to select this trunk when dialling
5. Set the **Zap Identifier** to **1** (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
```

10.2 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.

```

; 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 I"
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 (*in later versions of Trixbox you don't need to do this*):

If exist, 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 boostringer=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 "boostringer=1 ,fxshonormode=1". Also see [Appendix E.3](#) (Users Suggestions)

10.3 CALLER ID (CID) USING ZAP DEVICE

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

The simple answers to this:

- In Australia, for you to get Caller ID is actually having to apply to your Telco to have it activated on your line – this is not activated by default by your Telco

(Optus or Telstra). If caller ID is not activated on your line, you will not get CID.

- If you already have caller ID activated and still don't get caller ID, it's time to have a look at the ZAP configuration file. Typically you may have to look at **zapata.conf** and/or **zapata-auto.conf**.

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

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

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

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

10.3.1 zapata.conf

```
usecallingpres=yes  
callwaitingcallerid=yes  
threeewaycalling=yes  
usecallerid=yes  
hidecallerid=no  
relaxdtmf=yes
```

10.3.2 zapata-auto.conf

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

```
useincommingcalleridonzaptransfer=yes  
adsi=yes  
sendcalleridafter=2 ;you may need to add this switch
```

After the above are done, restart trixbox:

```
Amportal restart
```

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

10.4 REBUILD ZAPTEL DRIVER

IMPORTANT: If you are installing Trixbox 1.2.x or upgrading from 1.1.x to 1.2.2, this section does not apply to you.

If you are using TRIBOX v1.0 you will need to 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.*)

Hint: To find out what zaptel modules you have, issue the following command:

```
rpm -qa | grep zaptel-modules
```

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:

```
amportal stop  
service zaptel stop (it helps to stop the zaptel service as well)  
cd /usr/src/  
wget http://ftp.digium.com/pub/zaptel/releases/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 (*depending on the version you have. At the time of writing it is 2.6.9-34.0.1.EL-1686*):

```
cd /usr/src/kernels/2.6.9-34.0.1.EL-i686/include/linux  
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](#))

Editor's 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 would 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.

10.4.1 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:

```
reboot
```

Your ZAP device will now be recognised – hopefully.

10.5 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.

10.5.1 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.

10.5.2 Change the settings

System tab

DHCP: No

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

NetMask: 255.255.255.0

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

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

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

Regional tab

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

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

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

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

Ring 1 Cadence: 60(1.5/3.4)

Ring 3 Cadence:

60(1.5/3.4,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2,.4/2)

CWT8 Cadence: 30(.2/.2,.2/4.4)

Hook Flash Timer Min: .07

Hook Flash Timer Max: .13

Delete all the Vertical Service Activation Codes.

FXS Port Impedance: 220+820 || 120nF

Line 1 tab

Proxy: *IP address of your Asterisk box* e.g.; 192.168.1.234

Register Expires: 60

Display Name: *Whatever*

User ID: *Asterisk extension number* e.g.; 200

Password: *password for that extension*

Silence Threshold: *medium*

DTMF Tx Method: INFO
Hook Flash Tx Method: INFO
Dial Plan: (*xx|000|001xxxxxxxxx.|0[23478]xxxxxxxx|09xxxxxxxx|1100
|122[135]|1222xxxxxxxx|12510[12]|12554|1[38]00xxxxxxxx|13[1-9]xxx
|1747xxxxxxxx|2xx|393xxxxxxxx|3xxxx.|[4689]xxxxxxxx|7777|899060xxxxx.) for
example

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

PSTN Line tab (method 1)

Proxy: IP address of your Asterisk box e.g.; 192.168.1.234
Register: no
Make Call Without Reg: yes
Ans Call Without Reg: yes
Display Name: No name
User ID: PSTN
Password: password
Silence Supp Enable: no
Echo Canc Enable: no
Echo Canc Adapt Enable: no
Echo Supp Enable: no
FAX CED Detect Enable: yes
FAX CNG Detect Enable: yes
FAX Passthru Codec: G711u
FAX Codec Symmetric: no
FAX Passthru Method: None
DTMF Tx Method: INFO
FAX Process NSE: no
Dial Plan 1: (S0<:0298765432>) for example - it could be your phone number
VoIP Caller Default DP: none
PSTN Ring Thru Line 1: no
PSTN CID For VoIP CID: yes
PSTN Answer Delay: 2
PSTN Ring Thru Delay: 3
PSTN Ring Timeout: 4
PSTN Hook Flash Len: .1
Disconnect Tone: 425@-30,425@-30;1(.375/.375/1+2)
FXO Port Impedance: 220+820||120nF
On-Hook Speed: 26ms (Australia)

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

Note:

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

PSTN Line tab (method 2)

Proxy: IP address of your Asterisk box e.g.; 192.168.1.234
Register: no
Make Call Without Reg: yes
Ans Call Without Reg: yes
Display Name: No name
User ID: PSTN
Password: leave blank
Silence Supp Enable: no

```

Echo Canc Enable: no
Echo Canc Adapt Enable: no
Echo Supp Enable: no
FAX CED Detect Enable: yes
FAX CNG Detect Enable: yes
FAX Passthru Codec: G711u
FAX Codec Symmetric: no
FAX Passthru Method: None
DTMF Tx Method: INFO
FAX Process NSE: no
Dial Plan 1: ($0<:s@YourAsteriskIP>) e.g. ($0<:s@192.168.0.101:5060>) or try w/o the port
designation
VoIP Caller Default DP: none
PSTN Ring Thru Line 1: no
PSTN CID For VoIP CID: yes
PSTN Answer Delay: 2
PSTN Ring Thru Delay: 3
PSTN Ring Timeout: 4
PSTN Hook Flash Len: .1
Disconnect Tone: 425@-30,425@-30;1(.390/.390/1+2)
FXO Port Impedance: 220+820||120nF
On-Hook Speed: 26ms (Australia)

```

Note:

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

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

User 1 tab

Default Ring: 3
Default CWT: 8

10.5.3 Add SIP Trunk

Then in freePBX, add a SIP trunk.

General Settings:

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

10.5.4 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 the later versions of **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 – I have resorted to this manual method because in the initial release of Trixbox, DISA did not work, as I would like it to. This has since been rectified

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

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

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

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

The example below 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 TRIXBOX 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. The top navigation bar includes links for Setup, Tools, Reports, Panel, and Recordings. A 'Setup' link is highlighted in orange. The main content area is titled 'DISA: PSTNVoIP (1)'. On the left is a sidebar with various configuration links. The main form contains fields for editing a DISA application:

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

A 'Submit Changes' button is at the bottom of the form. The footer indicates Version 2.1.1 on 192.168.1.100.

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 dialling the password followed by # followed by the telephone number.

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

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 sippone 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.

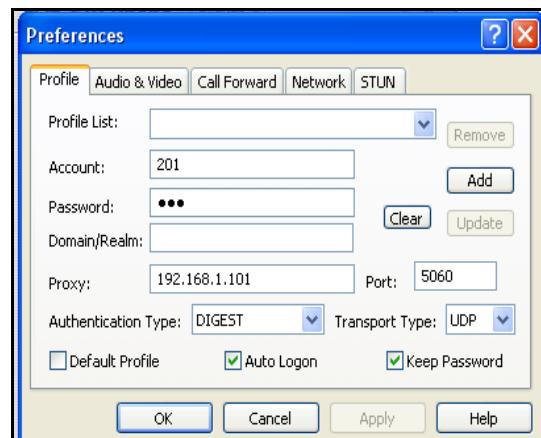
Profile Tab

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.

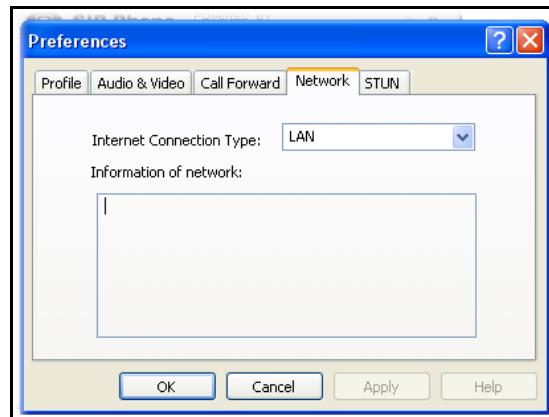
Click OK

Network Tab

Ensure that your Internet Connection Type is set to LAN.

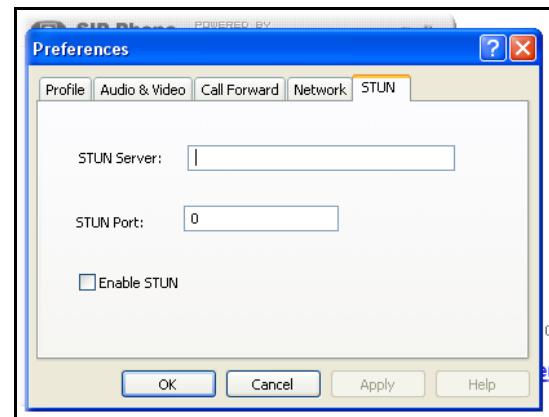
You may ignore the Information of Network field.

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 TRIXBOX.

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

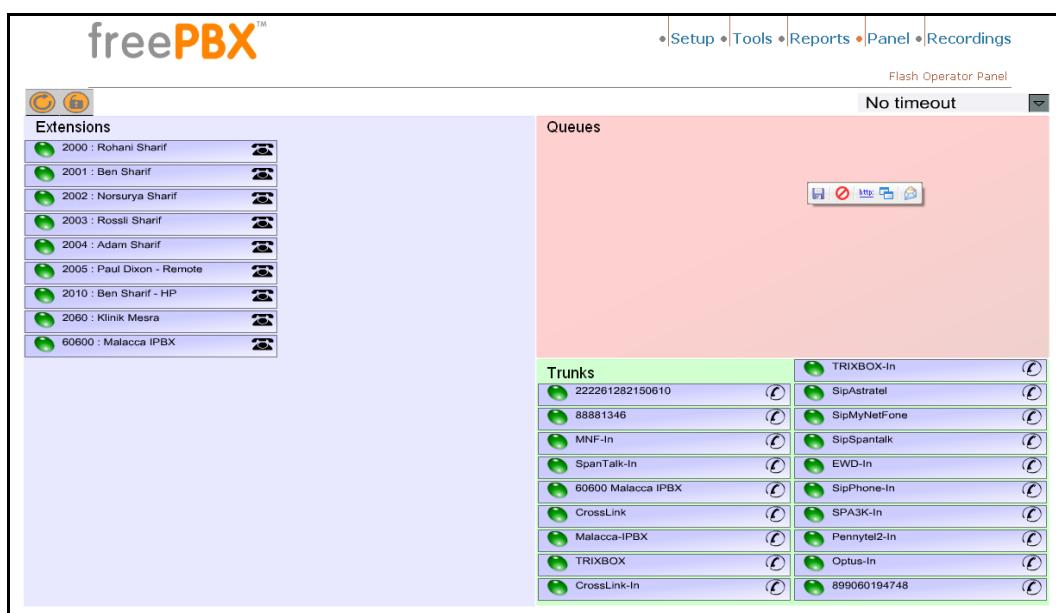
14.1 FLASH OPERATOR PANEL (FOP)

This is the standard Operator Panel that is already part of TRIXBOX – 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 TRIXBOX 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.asternic.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

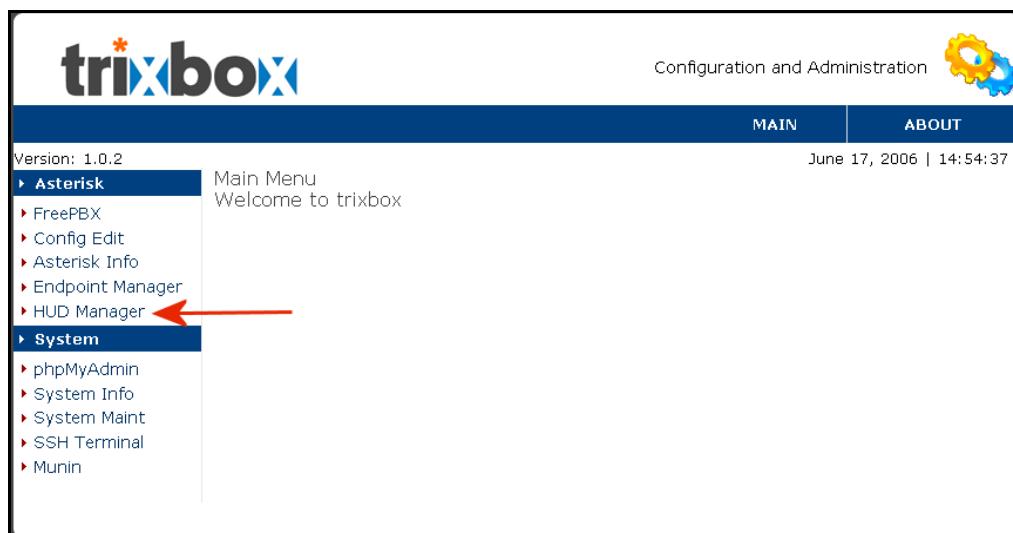
[HUDLite](#) from Fonality integrates very nicely with TRIXBOX and the HUD manager is already incorporated in TRIXBOX. It has most of the necessary functionalities (if not all) of a modern operator switch panel where you can tasks such as:

- 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.

Configuration and Administration

MAIN ABOUT

Version: 1.0.2

- ▶ Asterisk
 - ▶ FreePBX
 - ▶ Config Edit
 - ▶ Asterisk Info
 - ▶ Endpoint Manager
 - ▶ HUD Manager
- ▶ System
 - ▶ phpMyAdmin
 - ▶ System Info
 - ▶ System Maint
 - ▶ SSH Terminal
 - ▶ Munin

HUD ADMIN

Device:	Ext:	Name:	User Name:	Cell Phone:
SIP2000 [+]				X
	2000	Main Office	2000	X

[New device](#)

[+] manage device

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.

Configuration and Administration

MAIN ABOUT

Version: 1.0.2

- ▶ Asterisk
 - ▶ FreePBX
 - ▶ Config Edit
 - ▶ Asterisk Info
 - ▶ Endpoint Manager
 - ▶ HUD Manager
- ▶ System
 - ▶ phpMyAdmin
 - ▶ System Info
 - ▶ System Maint
 - ▶ SSH Terminal
 - ▶ Munin

HUD ADMIN

Device:	Ext:	Name:	User Name:	Cell Phone:
SIP2000 [+]				X
	2000	Main Office	2000	X

[New device](#)

[\[-\] manage device](#)

Device: SIP2001
 Ext: 2001
 Name: Ben Sharif
 Username: 2001
 Password:
 Email: bensharif@gmail.com
 IM:
 Cell Phone: 0400364780

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**.

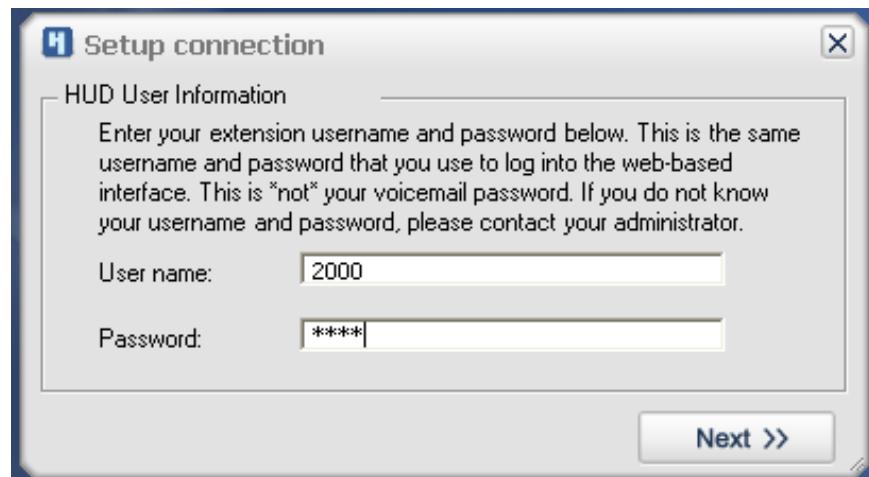
If you have more extensions to register, click on **New device** again and repeat this process as many extensions as you have.

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.

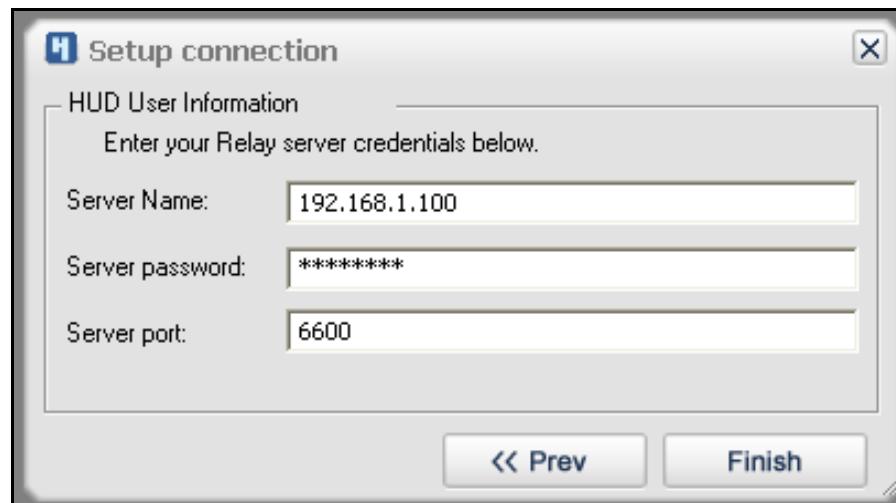
In my case I use my main office extension **2000**.

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 **features.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)

transferdigittimeout =>3 ; Number of seconds to wait between digits
courtesytone = beep ; Soundfile to play to the parked caller
; when someone dials a parked call

xfersound = beep ; to indicate an attended transfer is complete
xferfailsound = beeperr ; to indicate a failed transfer
;adsipark = yes ; if you want ADSI parking announcements
;pickupexten = *8 ; Configure the pickup extension. Default is *8
featuredigittimeout = 800 ; Max time (ms) between digits for
;feature activation. Default is 500

[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 dialling *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-isophone: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.

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 setup option.

meetme_additional.conf

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

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

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:

The screenshot shows the freePBX web interface with the following details:

- Conference Room Number:** 8200
- Conference Name:** General Conference
- User PIN:** 1234
- Admin PIN:** 1235
- Conference Options (Selected Values):**
 - join message: No_Marketing
 - leader wait: Yes
 - quiet mode: No
 - user count: Yes
 - user join/leave: Yes
 - music on hold: Yes
 - allow menu: No
- Buttons:** Submit Changes

- 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 (this has been rectified in the later ISOs).

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 mailbox. 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 needed 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:

```
dnl 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.

Note:

*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. In the later versions of Trixbox, this can be done through Config Edit.

```
; 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
operator=yes ; before saving it
nextaftercmd=yes ; Allow caller to press 0
                  ; 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 **Voice mail 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 USE POSTFIX INSTEAD OF SENDMAIL

Another alternative to email notification is to use Postfix instead of Sendmail because Sendmail is not as secure besides, using Postfix is a lot easier and it is more secure than Sendmail.

If you wish to go this path, it is strongly recommend you remove Sendmail and replace it with Postfix, with only 2 easy commands:

```
rpm -e --nodeps sendmail  
yum -y install postfix
```

Following this, edit your /etc/postfix/main.cf, (use nano and my smtp server is shown below, but you should be using yours) and add:

```
relayhost = smtp.ozemail.com.au
```

Replace **smtp.ozemail.com.au** with your own smtp server. E.g. if you belong to OPTUS, it should be:

```
relayhost = mail.optusnet.com.au
```

You should also edit /etc/asterisk/vm_general.inc and set the "serveremail" parameter to your real email address. This is to avoid messages with invalid from-addresses floating aimlessly in the Internet.

Then load Postfix.

```
service postfix reload
```

Postfix will now send all outgoing email to your ISP's mail server.

17.3 HOW TO RETRIEVE VOICEMAIL FROM EXTERNAL PHONE

To call in and retrieve your voicemail from external phone is as easy as simply calling your number and retrieve your voice mail. This can be done easily if you have IVR enabled in your new TRIXBOX system.

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

17.3.1 Method 1

1. Call my number and when the Digital Receptionist answers, I will dial my extension number prefixing it with an *. E.g. My extension number is **2001**, so when the DR answers me, I will dial ***2001**
2. I will then be asked 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.

17.3.2 Method 2

1. From freePBX Setup Screen, create a **Misc Destination** and give it a name of e.g. "**Check VM**"
2. Select Feature Code **Dial Voicemail (*98)** from the dropdown menu.
3. Submit the changes.

Next modify your IVR

1. **Increase option.** If you already have 7 options, this will be the 8th.
2. Create an unannounced IVR selection e.g. selection no: 8

3. For this option, select **Misc Destination** and from the drop down, select **“Check VM”**
4. Save and don't forget to click on the red bar.
5. When you call your PBX from an external phone, DR will naturally ask you to make a selection. Selection 8 will not be announced, however you should select 8.
6. DR will ask for a **mailbox number**
7. Enter your **extension number** (which is also your mailbox number)
8. DR will then ask for your password
9. And you will get your mailbox IVR prompt.

As I said, it is indeed simple.

17.4 HOW TO CHANGE VOICE MAIL GREETINGS

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

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 – you are done.

18 FAX TO E-MAIL

Before setting up Fax over IP, I suggest that we should read this: <http://www.softswitch.org/foip.html> and : <http://www.voip-info.org/wiki-Asterisk+fax>

While sending and/or 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 INSTALLING FAX FOR TRIXBOX

Unlike previous versions of Trixbox/AAH where it has to be installed separately, the later versions of Trixbox uses Asterisk version 1.2.2 where the developers of Asterisk have included NVfaxdetect in Asterisk which allows incoming faxes to be received over SIP, IAX & ZAP channel..

There has been mixed reactions on the ability of sending and receiving fax through Trixbox as such I would suggest that, if you are relying on fax, to use a dedicated fax machine and only use Trixbox for experimental purposes only (at this stage anyway).

Note: *Whatever it is worth, I have not been able to utilise this facility to my satisfaction.*

19 WEATHER REPORTS

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. 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 in your **extensions_custom.conf** with some minor changes. You will want the digital receptionist to know which weather report you are referring to when you make the choice. Give it a label – see example below (*the changes made are in bold*).

```
; #####  
; Forecast for Sydney Metro  
; #####  
[custom-weathermetro]  
exten => s,1,Answer  
exten => s,2,Playback(national-weather-service)  
exten => s,3,Playback(pls-hold-while-try)  
exten => s,4,Playback(/tmp/weather1)  
exten => s,5,Hangup  
  
; #####  
; Forecast for the whole of NSW  
; #####  
[custom-weathernsw]  
exten => s,1,Answer  
exten => s,2,Playback(national-weather-service)  
exten => s,3,Playback(pls-hold-while-try)  
exten => s,4,Playback(/tmp/weather2)  
exten => s,5,Hangup
```

When you create your menu option in Digital receptionist, e.g. “Press 6 for Sydney weather report, press 7 for the whole of NSW weather report”, select the custom app: and enter the appropriate custom procedures above. See example below:

Custom App custom-weathermetro,s,1

Custom App custom-weathernsw,s,1

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 IAX2 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: Ben - Remote

Extension Options

Outbound CID:

Record Incoming: On Demand

Record Outgoing: On Demand

Device Options

secret	205
notransfer	yes
context	from-internal
host	dynamic
type	friend
port	4569
qualify	no
disallow	
allow	
dial	IAX2/205
accountcode	

Voicemail & Directory: Enabled

voicemail password: 205

email address: bensharif@gmail.com

pager email address:

email attachment: yes no

Play CID: yes no

Play Envelope: yes no

Delete Vmail: yes no

vm options:

vm context: default

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: Ben - Remote

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"/>

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.freedsns.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 SETTING UP AN IAX SOFTPHONE

I use an IAX softphone called IDEFISK. You can get IDEFISK from here:
http://www.asteriskguru.com/tools/idefisk_beta.php

You can also use Virbiage softphone as it will handle IAX.

Setting up IDEFISK is very simple. All you have to do is download the file, extract it and save it in its own directory.

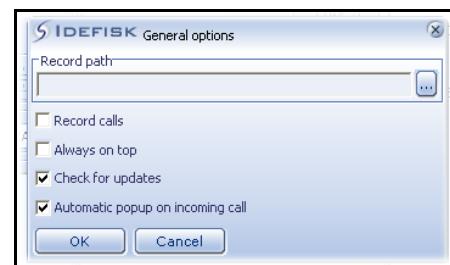
Click on the idefisk.exe and just configure it like the example below.



Once the softphone is active, to start configuring, simply place your 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 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 CALLS 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 recordings 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**

There are 2 ways you can record your calls:

22.1 RECORD ALL CALLS BY DEFAULT

To record all calls, on a specific extensions or all extensions, this function has to be enabled on the extension or extensions involved. This is done through the Extension Options Record Incoming and Record Outgoing field by selecting “Always” from the dropdown box for both fields.

22.2 RECORD CALLS ON DEMAND.

To record on demand, the caller or the receiver have to dial *1 when he/she need the recording to start (Asterisk v1.2). This function will only work if the feature is enabled.

To enable this feature, the following Dial Command Option must be enabled in freePBX **General settings Asterisk Dial command option** and **Asterisk Outbound Dial command option** fields by adding the following flag in the fields.

w: Allow the called user to start recording after pressing *1 (Asterisk v1.2)
W: Allow the calling user to start recording after pressing *1 (Asterisk v1.2)

Some users have noticed that the *1 need to be pressed very quickly for this feature to work. This can be remedied by adding the following line in the **features.conf** file under the general section.

```
featuredigittimeout = 1500 ;(the default is 500 which is too fast for an old guy like me
```

23 STUN SERVERS

STUN (Simple Traversal of UDP through NAT - Network Address Translation) is a protocol for assisting devices behind a NAT firewall or router with their packet routing operates on TCP and UDP port 3478.

I never have the need to use STUN. The list below is provided for those who have the need to use public STUN Servers. These are the servers that I am aware off - there may be others.

stun.xten.com	stun1.noc.ams-ix.net
stun.fwd.org	stun.voipbuster.com
stun01.sipphone.com	stun.voxgratia.org

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/assttapi/>

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.

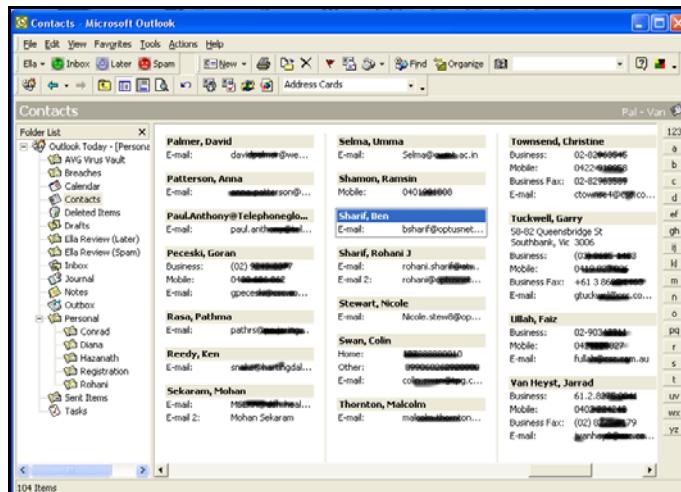
24.3 CONFIGURE OUTLOOK ADDRESS Book

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).

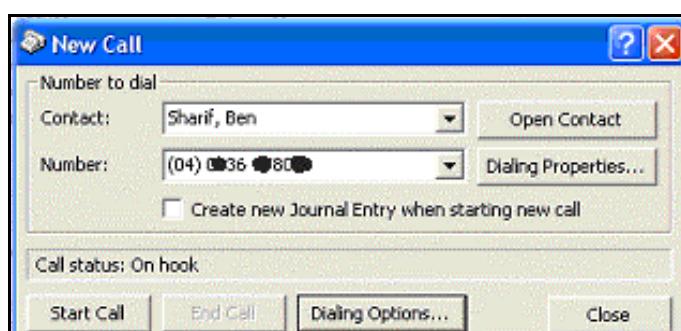
Click on the phone icon.



Click on the Telephone icon.

The New Call dialog box will be displayed:

Select Dialling Options.



The Dialling Option box will be displayed.

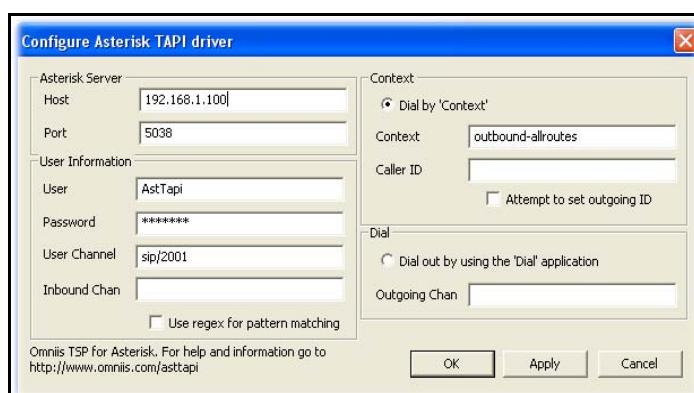
In the "connect using line" drop down box, select Asterisk.

Then click on Line properties.



In the **Host** field, enter the address of your asterisk server: e.g. **192.168.1.100**

In the Port Field: **5038**
User Field: **AstTapi**
Password: **AstTapi**



User Channel: (your extension – the extension you want Asterisk to ring before connecting you the number you are calling) e.g. **sip/2001**

Select dial by context radio button.

Enter **outbound-allroutes** in the context box

When done, click Apply and you are done.

You can now start making calls from Outlook using the click-to-dial facility.

Now... that was quite painless ☺

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 3003XX61212345680 (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:

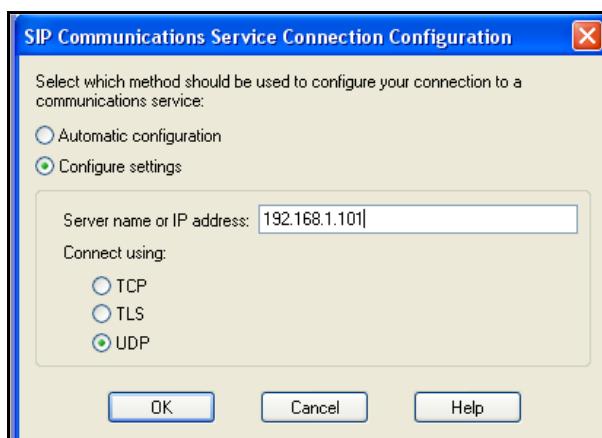
1. Start Windows Messenger from your PC.

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



2. Click on the [Advanced] button next to the Sign-in name.



- Check the [Configure settings] button
- Enter Asterisk IP address (e.g.192.168.1.101) in the [Server name of IP address] field
- Check the [UDP] button
- Click [OK] and [OK] again.

In order for Windows Messenger to work as a voice communication device, you must have a microphone and speaker(s) connected to your PC.

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

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 `xxxxyy` 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>
IAX Trunk		
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	91000:xxxxyy@xyz.com.au	90000:xxxxyy@def.com.au

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

	System 1	System 2
Extensions		
Phone Protocol	IAX	IAX
Extension Number	90000	91000
Extension Password	xxxxyy	xxxxyy
Fullscreen	Parramatta	Main Office
Voicemail & Directory	Disabled	Disabled

	System 1	System 2
Outbound Routing		
Route Name	Parramatta	MainOffice
Route Password	Leave Blank	Leave Blank
Dial Patterns	6XXX <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/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.

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.

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

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

	System 1	System 2
<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

27.3 METHOD 3 – FRIEND/FRIEND ARRANGEMENT

(By far the simplest method)

This method is by far the simplest method where the boxes are treated as friend to each other. Like Method 2, registration is not required and this method will also pass the Caller ID to the receiving party.

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

	System 1	System 2
<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

All the above examples assume that both Asterisk boxes have Public Fix IP address. If you have Dynamic IP addresses, you will need to register both the boxes with DynDns to obtain a valid DNS ID. If you are a part of a Corporate LAN, than you will have no need to worry about DynDns and what not.

Warning: You must provide for security, as this is pretty wide open.

Like all installation, you must provide for security. As different installation resorts to different types of security arrangement, I will leave that to the individual implementer to deal with the security issues.

27.4 THINKING OF MORE THAN 2 BOXES?

Just as a matter of interest, you can connect several boxes using this method. While I have connected 3 boxes successfully, I believe, the same principle can be applied to more boxes.

In my implementation I have box A, B and C (System 1, 2 and 3) in a star configuration. Box A is the master box. All the other boxes use box A as the main exchange.

A peers with B and C - B peers with A - And C peers with A.

Except for local traffic, all external and inter-office (inter-branch) traffic goes via Box A. – with the appropriate dial plan of course.

Both the above methods, while useable for a basic configuration, will not provide you with a complete solution. To provide a complete solution is beyond the scope of this document.

The following link will provide further reference for connecting two Asterisk boxes together
<http://www.voip-info.org/wiki/view/Asterisk++dual+servers>

If you require a complete solution tailored to your exact requirement, my advise to you is to hire a VOIP consultant.

27.5 REDIRECT INCOMING IN BOX A TO BOX B

There will be times when you will want to redirect certain calls made to one Trixbox box to another Trixbox box in another location. This can happen in a scenario where you are moving your mail office from your existing location to another location (or moving your sales office to another location away from your main office).

Personally I really wouldn't do this as it is simpler to get a redirection to the new number from the exchange.

In this example, lets say that all calls coming to your head office Trixbox PABX made on phone number **12345678** need to be sent to your Trixbox PABX in a different location.

Let us call your Head Office PABX Box-A and your remote PABX Box-B and calls coming in to Box-A to be sent to Box-B.

There are a number of ways that this can be achieved but most involved editing conf files and adding extra codes in the conf files. Unless you are quite comfortable with doing this, it can be quite daunting to the uninitiated.

The example I am illustrating here does not involve editing any conf file. All can be done through freePBX. It is a little long winded but you don't have to touch any of the conf files.

Step 1 – (Do this in Box-A)

i. Create an IAX Trunk

You will only need the following in the **Outgoing Setting** and leave everything else blank. Let us call this trunk **TransferTrunk**.

```
host=BoxB.com.au (your qualified DNS or the IP address)
type=peer
username=6000 (this number will be used later)
```

ii. Create and Outbound Route

After you have created the trunk, you will need to create an outbound route. Let us call this outbound route *TransferRoute*.

In the dial pattern, you will only require one entry:

6000

The trunk sequent will be **TransferTrunk** only.

iii. Create a new extension

Next create a new extension e.g. 500. You will be using this extension to unconditionally forward all calls made to this extension, to extension 6000 in BoxB, which we will create later.

To unconditionally forward calls made to this extension to extension 6000, you need to connect to extension 500 using a softphone and dial ***726000**. Any calls to this extension will be unconditionally forwarded to extension 9000.

iv. Create and inbound route

DID number is 12345678. This is your phone number of calls come in to Box-A from, assuming that all calls that come on this number are to be transferred to Box-B.

Set Destination to Core **500**.

That's all need to be done in Box-A

Step 2 – (Do this in Box B)

i. Create a Trunk

Trunk name: **TransferTrunk**. Leave the peer details blank.

User context: **Transfer-In**

You will only need the following in the **Incoming Setting** and leave everything else blank.

context=from-trunk

ii. Create a new IAX extension

Create a new IAX extension **6000** (this is the username defined in the outgoing setting in Step 1 above). Display name: **CallTransfers**.

Edit this extension context to read **from-trunk** (instead of from-internal).

iii. Direct all calls to IVR

Ensure that incoming route direct calls to the appropriate extension. In my case, all calls go to IVR.

That is all we need to do and from now on, calls that come in to Box-A on your phone number 12345678 will be directed to extension 500 of Box-A. However, since all calls to extension 500 has been unconditionally redirected to extension 6000, asterisk will forward the call to extension 6000 which your outbound route will pick up and send to Box-B.

When it gets to Box-B, it will be answered by the IVR in Box-B.

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 the entire prompt files there.

Add the following line in the iax.conf and sip.conf configuration file under [general]. In zapata.conf configuration file the line should be added under [channels].

`language=au ; (or fr, it or whichever language is installed)`

(You may use the Edit Configuration facility of 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).

28.3 CONVERTING WAV FILES

For those who wish to record their own sound prompts, the following may be of some assistance.

Trixbox will be able to play virtually any sound that it has the codec for e.g. wav(pcm), wav49, gsm, g711, g729 etc. However, the gsm format seems to be the common format used for the default voice prompts.

28.3.1 Converting WAV to GSM

Since the WAV format is the most common format that is being adopted when recording with a windows based PC, and most times, the windows based WAV format does not play well with asterisk, below is a method of converting the windows recorded WAV format to GSM using SOX.

Your result will be better if you record your sound files in mono, 16 bit, 8000 Hz.

After recording the WAV sound files, transfer the sound files to the TEMP directory of the Trixbox PC. In this example, let's call one of the sound files **hello.wav**.

- ❑ Login as root and change directory to the directory where you have transferred the sound files to e.g. `cd /tmp`
- ❑ At the prompt issue the following commands.
 - If your sound files were recorded in mono, 16 bit, 8000 Hz

```
sox hello.wav hello.gsm
```

- If your sound files were recorded in mono but NOT 16 bit, 8000 Hz

```
sox hello.wav -r 8000 hello.gsm resample -ql
```

- If your sound files were recorded in stereo, you will need the `-cl` switch.

```
sox hello.wav -r 8000 -cl hello.gsm resample -ql
```

- if your sound files were recorded in ADPCM wav files, to convert to standard wav file;

```
sox hello.wav -r 8000 -c1 -s -w hello-out.wav resample -ql
```

- Next, move the sound file to the sound directory where all your prompts are stored and you are done.

You may also convert a number of WAV files at once using the following command. In this example, lets assume that the files were all recorded in stereo;

```
for a in *.wav; do sox "$a" -r 8000 -c1 "`echo $a|sed -e s/wav//`gsm" resample -ql
```

28.3.2 Converting WAV to SLN

Starting from Asterisk version 1.2.0, the .sln (SLINEAR) format seems to be the format that is being adopted (the GSM format is still commonly being used), so don't worry about converting your existing prompts.

To convert wav file to sln, use the following command:

```
sox hello.wav -t raw -r 8000 -s -w -c1 hello.sln
```

Further reference for converting wav sound files can be found [here](#) at voip-info.org.

29 MOH (MUSIC ON HOLD)

Trixbox comes with a number of preinstalled music on hold selections, however you may want to add your own music to the existing selection or remove the default selection altogether, and only use your own selection instead.

29.1 CUSTOMISING MOH

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.

To customise your music to your favourite, you may need to create a new music category directory to hold your favourite music. Do this through the freePBX "On Hold Music" GUI interface. I call my category **favourites**.

Upload your favourite music to this directory. Again, do this through the freePBX GUI. Once finished uploading, click the "Enable Random Play" button and the red bar.

Make a little change to your musiconhold.conf file like the example below (this way you don't have to delete the music in your default directory).

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

The **musiconhold.conf** files in 1.2 is now set out differently as per the example of my file below:

```
musiconhold.conf
;
; Music on hold class definitions
; This is using the new 1.2 config file format, and will not work with
1.0
; based Asterisk systems
;
[default]
mode=files
;
; valid mode options:
; quietmp3 -- default
; mp3 -- loud
; mp3nb -- unbuffered
; quietmp3nb -- quiet unbuffered
; custom -- run a custom application
; files -- read files from a directory in any Asterisk supported format
;
directory=/var/lib/asterisk/mohmp3/favourite ;<- I pointed this to my favourite
```

```
random=yes  
#include musiconhold_additional.conf
```

The corresponding entries can be found in `musiconhold_additional.conf` – see below:

`musiconhold_additional.conf`

```
[acc_1]  
mode=files  
directory=/var/lib/asterisk/mohmp3/acc_1/  
random=yes
```

29.2 MOH FOR DIFFERENT TRUNKS

So you are hosting three different sections of your business on your Trixbox with three different Phone numbers (or trunks for that matter) and would like to have different music on hold for the different Music On Hold for each of the section.

While freePBX does not have a ready method of doing it from the GUI, we can trick the system into doing it with a simple creative modification. Here goes.

Create 3 separate music categories (Rock, country and classical) and populate them with MP3s or whatever that turns you on.

Create 3 custom context in **`extensions_custom.conf`**

```
[custom-department_1]  
exten => s,1,playback(pls-wait-connect-call)  
exten => s,n,SetMusicOnHold(country)  
exten => s,n,Dial(SIP/2001,60,r)  
exten => s,n,Macro(vm,2001); voice mail to extension if no answer  
exten => s,n,Hangup  
  
[custom-department_2]  
exten => s,1,playback(pls-wait-connect-call)  
exten => s,n,SetMusicOnHold(classical)  
exten => s,n,Dial(SIP/2002,60,r)  
exten => s,n,Macro(vm,2002); voice mail to extension if no answer  
exten => s,n,Hangup  
  
[custom-department_3]  
exten => s,1,playback(pls-wait-connect-call)  
exten => s,n,SetMusicOnHold(Rock)  
exten => s,n,Dial(SIP/2003,60,r)  
exten => s,n,Macro(vm,2003); voice mail to extension if no answer  
exten => s,n,Hangup
```

Assuming 2001,2002 and 2003 being the extensions of the various departments.

Incoming route for Trunk1 send to

Custom App custom-department_1,s,1

Incoming route for Trunk2 send to

Custom App custom-department_2,s,1

Incoming route for Trunk3 send to

Custom App custom-department_3,s,1

29.3 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:

```
cd /usr/bin  
wget http://www.mpg123.de/mpg123/mpg123-0.59r.tar.gz  
tar -zxvf mpg123-0.59r.tar.gz  
cd mpg123-0.59r  
make linux  
make install  
ln -s /usr/local/bin/mpg123 /usr/bin/mpg123
```

I have to make a directory for the streaming music. You can do this though the freePBX GUI or manually. In this example, I have shown the manual method.

```
su asterisk  
cd /var/lib/asterisk/mohmp3/  
mkdir stream
```

Next I need to create a zero length dummy MP3 file for this purpose in the **/var/lib/asterisk/mohmp3/stream** directory (this can be done manually).

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

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

Open the **musiconhold.conf** file using the Config edit option on your TRIXBOX and add the following lines on top of the file;

```
[classes]  
default => quietmp3:/var/lib/asterisk/mohmp3/stream,http://128.177.3.80:4064/
```

and leave everything else out (or commented out) like my example below.

The **http://128.177.3.80:4064/** (*SKY.FM – Country*) is where I get my streaming music from.

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

```
[classes]  
;default => quietmp3:/var/lib/asterisk/mohmp3/stream,http://radioio.sc.llnwd.net:8075/  
;default => quietmp3:/var/lib/asterisk/mohmp3/stream,http://202.6.74.107:8060/triplej.mp3  
default => quietmp3:/var/lib/asterisk/mohmp3/stream,http://128.177.3.80:4064/  
;  
;#include musiconhold_additional.conf
```

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.

The following are addresses where you may find some streaming music for your MOH.

181.kickin' country <http://64.236.34.106:80/stream/1075/>
Classical 96.3 FM <http://216.126.84.50:8000/>
SMOOTHJAZZ.COM <http://scfire-chi0l-2.stream.aol.com:80/stream/1005/>
Sky FM – Modern Jazz <http://205.188.215.227:8008/>
Energy 98 – Dance Hit <http://193.138.205.194:8014/>
Life FM – Adelaide Australia <http://203.16.214.174:8040/>
Groove Mix <http://63.249.3.194:8010/>
The 80s channel <http://64.236.34.106:80/stream/1040/>

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:

The screenshot shows the 'System Backup' configuration page in the freePBX interface. The 'Schedule Name:' field is set to 'Monthly'. Under 'Run Schedule', the 'Run Backup' dropdown is set to 'Monthly (on the 1st)'. The time is set to 00:00:00. The 'Days' section has 'All' selected. The 'Months' section has 'Selected' selected, showing January through December. The 'Weekdays' section has 'All' selected, showing Monday through Sunday. A 'Submit Changes' button is at the bottom.

- 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**. The backup files are identified by the backup names.

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 SUGARCRM CONFIGURATION

SugarCRM is a contact management software that comes bundled with Trixbox.

To set up SugarCRM, first, you need to open the SugarCRM application <http://<your.trixboxip.address>/crm> using the default username of **Admin** and the password of **password**.

For security reason you should change the Admin password. To do this, click on 'My Account' in the upper right-hand corner, then click on the 'Change Password' button underneath 'Users: Administrator (Admin) in the center-left of the screen.

Change it to a new password and confirm your new password and click 'Save.'

Now it's time to set up your contacts. I will start off setting up a couple of my internal extensions.

Click on 'My Account' again and then click the 'Edit' button.

Change 'Asterisk Phone Extension' to your Asterisk extension. My extension is 2001.

While you are at it, change your time zone and date format as well.

Click 'Save' to save that information.

Let's add another one.

Click on the contacts tab and then select 'Create Contact' from the left hand Shortcuts menu.

Add another extension, in my case I chose my daughter's extension 2002:

Firstname: Norsurya

Last name: Sharif

Home: 2002

Click 'Save' to save that information.

Add another and another if you want to, using the method above.

At this point, you may find that you are unable to make a phone call through SugarCRM. This is due to a little bug in the `popup_picker.php` (this bug may have been fixed by the time you read this, but at the time of writing, this bug exists).

To fix this bug, you need to edit `popup_picker.php` by doing the following:

From your Linux CLI, log in as root.

```
cd /var/www/html/crm/modules/Contacts
```

```
nano Popup_picker.php
```

Browse down to line 121 and change it from:

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

To

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

You should now be able to dial from SugarCRM to your other internal extensions and to the outside world.

Note: *You can add multiple users who will each have their own settings/contacts/etc.*

32 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.sedhacomputer.com/products/se-p1k.html>
http://www.welltech.com/product_e_06.htm
<http://www.waytec.com.au/download.htm>
<http://www.pc-telephone.com/free-phone.htm>
<http://www.sjlabs.com/usbphone/SJphoneDriverATCOM-AU-100.exe>

33 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 TRIXBOX 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 ☺

33.1 INSTALLING G729 AND G723 CODECs

If you have the licence for the g729 codec and found that the digium version of g729 is a little problematic and would like to try the free version, following is the method of installing it as described by David Klaverstyn – thanks David.

DISCLAIMER:

You might have to pay royalty fees to the G.729/723 patent holders for using their algorithm.

Before downloading the codec module you need to verify what your processor type is, as to get a compatible compiled codec.

From the Linux command type in:

```
# cat /proc/cpuinfo
```

Look specifically at the model name and the flags. The following is an example of the process that I am using.

```
Process:          0
vendor_id:       GenuineIntel
cpu family:      6
model:           8
model name:      Pentium III (Coppermine)
stepping:        3
cpu MHz:         648.110
cache size:      256 KB
fdiv_bug:        no
hlt_bug:         no
f00f_bug:        no
coma_bug:        no
fpu:             yes
fpu_exception:   yes
cpuid level:    2
wp:              yes
flags:            fpu vme de pse tsc msr pae mce cx8 mtrr pge mca
cmov pat pse36 mmx fxsr sse
bogomips:        1297.41
```

From this information we can see that it is a Pentium III processor and it is using **sse**. This is the 2 bits of required information that we need. If you don't see **sse** in your flag then the file you need must contain the words **no-sse**.

Now go to <http://kvin.lv/pub/Linux/Asterisk/> and select wither Asterisk version 1.2 or 1.4 depending on the version you are using.

The files are named as codec_g72x-???-pentiumx.so

So from the information above to obtain the G729 codec we need to get the file codec_g729-gcc-pentium3.so and for the G723 codec we need to get the file g723-gcc-pentium3.so

The gcc represents the way the module was compiled. This is your most likely choice. To view the different type of compiled version look at this file <http://kvin.lv/pub/Linux/Asterisk/README>

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

Change folders to /usr/lib/asterisk/modules

```
# cd /usr/lib/asterisk/modules
```

Download the required file(s)

For the G729 codec

For Pentium III

```
# wget http://kvin.lv/pub/Linux/Asterisk/built-for-asterisk-1.2/codec_g729-gcc-pentium3.so
```

For Pentium IV

```
# wget http://kvin.lv/pub/Linux/Asterisk/built-for-asterisk-1.2/codec_g729-gcc-pentium4.so
```

For the G723 codec

For Pentium III

```
# wget http://kvin.lv/pub/Linux/Asterisk/built-for-asterisk-1.2/codec_g723-gcc-pentium3.so
```

For Pentium IV

```
# http://kvin.lv/pub/Linux/Asterisk/built-for-asterisk-1.2/codec_g723-icc-pentium4.so
```

Restart asterisk and you are done.

If Asterisk does not start then you may need to try **icc** files instead of **gcc**.

34 PSTN TRUNK – SPECIAL HANDLING

34.1 DIAL 0 FOR OUTSIDE LINE

There will be times when a user would like to get an outside line something similar to a PABX by simply dialling a number e.g. 0 or 9.

I got around it by creating an entry in **extensions_custom.conf**.

In my case I use 0 as the prefix to get an outside line.

I have also pre-recorded a voice prompt to instruct the caller what to do after the second dial tone.

34.1.1 Using Zaptel Card Method

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

exten => 0,1,Playback(start-dial-after-dialtone)
exten => 0,2,Dial(ZAP/1/{EXTEN})
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.*

34.1.2 Using SPA3K Method

Using SPA3K will be a little tricky. I am describing how I got around it (assuming that you have set up the SPA3K as described in this document), however there maybe other method.

1. In the SPA Line 1 Tab, I have included the following entry in the dialplan:

```
<0:#><:@gw0> )
```

2. The following is what my dialplan looks like in SPA3K

```
(*xx|xx.|<0:#><:@gw0> )
```

3. In the PSTN Line Tab, I have the following:

- VOIP-PSTN Gateway Enabled: Yes
- VOIP Caller Auth Method: None
- VOIP Caller Default DP: None

I 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.

34.2 GIVING PRIORITY TO EMERGENCY NUMBERS (E.G. 000)

Emergencies, by definition should be given right of way. This applies to the right of using the telephone for emergency purposes above other uses.

Imagine what will happen if you want to use the PSTN line to call emergency 000 (in Australia) and someone else was hogging the line. The sensible way is for Asterisk to disconnect the line and allow you to make the emergency call instead.

The following method will provide you with the facility, however you need to do some editing to the extensions_custom.conf file in /etc/asterisk.

In my case I have pre-recorded a voice prompt called 'calling-emergency' which says, "You are trying to call emergency assistance. If you have dial this in error, please hang-up"

```
;#####
; Give priority to emergency number by disconnecting existing user
; if using the PSTN line first, otherwise simply call the emergency number.
;#####
exten => 000,1,Background(calling-emergency) ; voice warning
exten => 000,2,Wait(1) ; give caller a chance to hang up
exten => 000,3,Dial(Zap/1/000,,j) ; If ZAP 1 is busy, jump to n+101
exten => 000,4,Hangup() ; hang up after the call
exten => 000,104,Softhangup(Zap/1-1) ; hang up the line if being used
exten => 000,105,Goto(3) ; make call after disconnecting the other user
```

35 PORT FORWARDING AND QOS – ROUTERS

35.1 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

The screenshot shows the 'Port Range Forward' section of the router's web interface. The table lists five port forwarding rules:

Application	Start	End	Protocol	IP Address	Enable
IAX	4569	to	4569	Both	192.168.1.101
AMP*	8080	to	8080	Both	192.168.1.101
Azareus	50001	to	50010	Both	192.168.1.3
SIP	5060	to	5065	Both	192.168.1.101
RTP	10000	to	11000	UDP	192.168.1.101

A sidebar on the right provides a detailed explanation of Port Range Forwarding:

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.

35.2 QOS – ROUTERS

Some routers have QOS facility while some routers haven't. It is highly recommended that you enable this function if your router supports it.

There have been a number of discussions on the effectiveness of QOS on broadband utilising the public Internet circuit. It is beyond this guide to document the effectiveness or otherwise of enabling QOS on your router. For a start, there are just too many routers there for me to give any meaningful examples.

I use the Linksys™ WRT54G®. Taking the easy way out, I simply restrict the PCs (see below) that more often than not will be used as a P2P client or chatting on the MSN chat channels.

I restrict the PCs according to their MAC addresses.

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

My recommendation is, if your router has the function, use it. At the very least, it will prioritise traffic within your own LAN especially when you have sons and daughters (for that matter you too can be the culprit) who are always on the P2P, leeching files and music.

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

36 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:

```
amportal stop      (Remember to always stop asterisk first before doing this)
service zaptel stop (it helps to stop the zaptel service as well)
```

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 : php-pear              ##### [13/48]
Installing: kernel-devel        ##### [14/48]
Updating : sendmail              ##
Updating : sendmail              ##### [15/48] warning: /etc/mail/submit.cf created as /etc/mail/submit.cf.rpmnew
Updating : tldata                ##### [16/48]
```

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]
    0:adding group munin...
    0:adding user munin...
  Installing: munin               #####[ 6/7]
    0:group munin already present
    0: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.noarch 0:0.93-1.el4.rf perl-rrdtool.i386 0:1.0.50-3.2.el4.rf rrdtool.i386 0:1.0.50-3.2.el4.rf
Complete!
done.

Please reboot by typing reboot
[root@asterisk1 ~]#

```

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

reboot.

After reboot, log in to Asterisk and at the command prompt issue the following command to configure zaptel.

```
genzaptelconf
```

After that restart amportal.

Amportal restart.

Using your browser on another PC, 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 first section, 'Subscribe/Notify', displays the message: 'There are no registered dialplan hints'. The second section, 'Zaptel driver info', lists channel configurations. A red arrow points to the 'Language' column for the 'pseudo' channel, which is set to 'en'.

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

Verbosity is at least 1
Core debug is at least 1

Chances are, ZapTel will be found.

Note:

In Trixbox 1.2.2, there has been reports of problems associated with not being able to record messages, VM not working, wake up calls not working, etc

It seems like a patch from Asterisk that was needed for permissions was removed and, for some reason the extra sounds didn't get installed.

The following will fix all the reported 1.2.2 problems.

Go the command console on your trixbox and log in as root (you can also use the JavaSSH client in the GUI). At the command prompt, issue the following commands:

```
yum clean all
yum -y update asterisk
rpm -i --force /var/trixbox_load/rpms/asterisk-sounds-1.2.1-1.noarch.rpm
rm -rf /tmp/*
reboot

cd /usr/src
wget http://yum.trixbox.org/centos/4/devel/zaptel-1.2.9.1-4.1467.i686.rpm
wget http://yum.trixbox.org/centos/4/devel/zaptel-modules-2.6.9-34.0.2.EL-
1.2.9.1-4.1467.i686.rpm

rpm -i --force zaptel-1.2.9.1-4.1467.i686.rpm
rpm -i --force zaptel-modules-2.6.9-34.0.2.EL-1.2.9.1-4.1467.i686.rpm
reboot
```

This should solve the problem.

37 TOOLS

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

37.1 WEBMIN

Webmin is an invaluable web based tool for managing a Linux box. Webmin make it easy to configure application like SMTP mail, editing files etc (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  
-1.300-1.noarch.rpm
```

Install it with the following command through CLI:

```
rpm -Uvh webmin-1.300-1.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-1.300-1.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  
;  
rtpstart=10000 <-- change this to 10002(totally unnecessary.. ☺)
```

rtpend=20000

37.1.1 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 ☺

37.2 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>

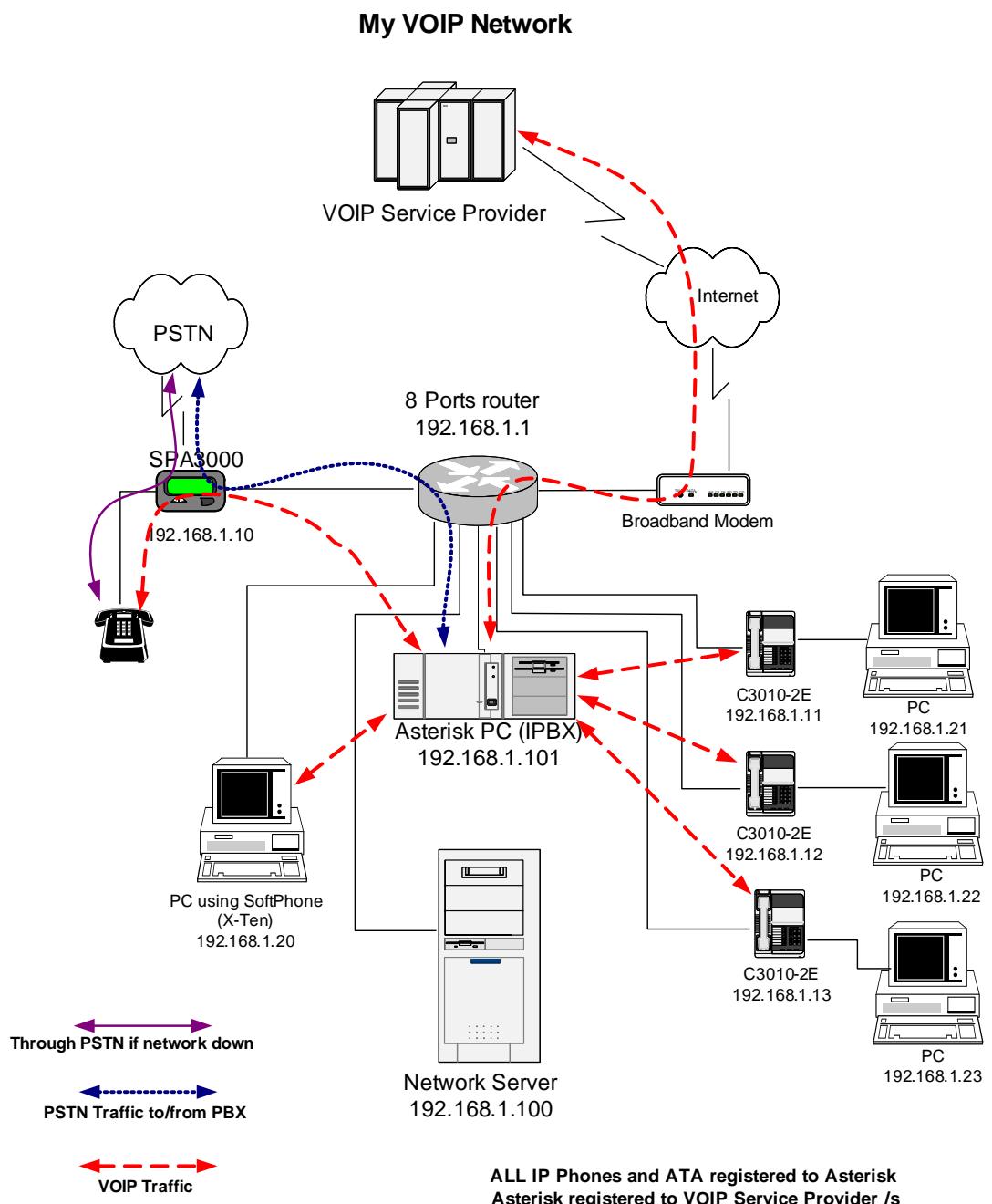
37.3 WINSSCP

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>

38 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.



39 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.

39.1 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 intercepting the codes thinking that it is its own feature codes.

This also applies to all other ATAs, FXS devices such as the various routers with built-in FXS ports and IP Phones that has their own feature codes built-in. Unless these feature codes are disabled, they are likely to clash with Asterisk's feature codes and as such Asterisk feature codes will not give you the desired results.

39.2 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**.

39.3 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.

39.4 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 section on [Calls Handling](#) in this guide:

39.5 SIPURA SPA3000 – NOT HANGING UP AFTER PSTN CALLS

For months I have to wrestle with the problem of SPA3K not hanging up after a PSTN call. Although the caller has hung up, SPA3K will hold the line off hook until I manually power cycle SPA3K.

The following settings may overcome the problem:

PSTN Line Tab

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) which seems to work well for Telstra, and replaced it with 425@-30,425@-30;1(.390/.390/1+2) which works better with Optus.

Only one catch with this: if you put somebody on hold for longer than the above settings, if they don't make any noise, it will hang up.

39.6 TRIXBOX 1.2 WILL NOT PLAY NICELY

Sometime, no matter what you do, Trixbox will not install correctly. A module or two may not compile etc. If this is the case and you are still having problem installing Trixbox 1.2, it probably is time for you to do some manual configuration as described in Rob Thomas' blog.

Here is the link and as I have personally done it this way, I can assure you that it will solve your problem – I hope :)

<http://www.freepbx.org/2006/09/28/un-trixbox-your-trixbox/#more-7>

APPENDIX A

A.1 ASTERISK FEATURE CODES

Asterisk feature codes are defined in the feature codes selection of the Setup Tab, illustrated below:

The screenshot shows the freePBX Feature Code Admin interface. The left sidebar has a 'Feature Codes' link highlighted with a red arrow. The top navigation bar has a 'Setup' link highlighted with a red arrow. The main area displays a table of feature codes with columns for 'Use Default?' and 'Feature Status'. The table includes sections for Call Forward, Call Waiting, Core, Gabcast, Info Services, Recordings, and Voicemail. A 'Submit Changes' button is at the bottom right.

#	Directory
*11	User Log on
*12	User Log off
*43	Echo Test
*52	Call Forward No Answer/Unavailable Activate
*53	Call Forward No Answer/Unavailable Deactivate
*60	Time
*61	Weather
*62	Schedule wakeup call
*65	Speak your Extension number (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
*74	Call Forward all prompting deactivate
*77	IVR Recording
*78	Enable Do-Not-Disturb
*79	Disable Do-Not-Disturb
*90	Call Forward on Busy
*91	Disable Call Forward on Busy
*92	Call Forward Busy Prompting deactivate
*97	Message Center (does no ask for extension)
*97 + 0	Change email greeting message.
*98	Enter Message Center
*99	Playback IVR Recording
*411	Directory (while in IVR – similar to #)
*422	Connect to Gabcast

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 TRIXBOX – **amportal stop**

To start TRIXBOX – **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 TRIXBOX and
amportal stop: Restart TRIXBOX.
debug channel: Enable debugging on a channel
dont include: Remove a specified include from context
help: Display help list, or specific help on a command
include context: Include context in other context
load: Load a dynamic module by name
logger reload: Reopen log files. Use after rotating the log files.
no debug channel: Disable debugging on a channel
pri debug span: Enables PRI debugging on a span
pri intense debug span: Enables REALLY INTENSE PRI debugging
pri no debug span: Disables PRI debugging on a span
remove extension: Remove a specified extension
remove ignorepat: Remove ignore pattern from context
remove indication: Remove the given indication from the country
save dialplan: Overwrites your current **extensions.conf** file with an exported version based on the current state of the dialplan. A backup copy of your old **extensions.conf** is not saved. The initial values of global variables defined in the **[globals]** category retain their previous initial values; the current values of global variables are not written into the new **extensions.conf**. Using "save dialplan" will result in losing any comments in your current **extensions.conf**.
set verbose: Set level of verboseness
show agents: Show status of agents
show applications: Shows registered applications
show application: Describe a specific application
show channel: Display information on a specific channel
show channels: Display information on channels
show codecs: Display information on codecs
show conferences: Show status of conferences
show dialplan: Show dialplan
show image formats: Displays image formats
show indications: Show a list of all country/indications
show locals: Show status of local channels
show manager command: Show manager commands
show manager connect: Show connected manager users
show parkedcalls: Lists parked calls

show queues: Show status of queues
show switches: Show alternative switches
show translation: Display translation matrix
show voicemail users: List defined voicemail boxes
show voicemail zones: List zone message formats
soft hangup: Request a hangup on a given channel

A.2.2 AGI Commands

show agi: Show AGI commands or specific help
dump agihtml: Dumps a list of agi command in html format

A.2.3 Database Handling

database del: Removes database key/value
database deltree: Removes database keytree/values
database get: Gets database value
database put: Adds/updates database value
database show: Shows database contents

A.2.4 IAX Channel Commands

iax2 debug: Enable IAX debugging
iax2 no debug: Disable IAX debugging
iax2 set jitter: Sets IAX jitter buffer
iax2 show cache: Display IAX cached dialplan
iax2 show channels: Show active IAX channels
iax2 show peers: Show defined IAX peers
iax2 show registry: Show IAX registration status
iax2 show stats: Display IAX statistics
iax2 show users: Show defined IAX users
iax2 trunk debug: Request IAX trunk debug
iax debug: Enable IAX debugging
iax no debug: Disable IAX debugging
iax set jitter: Sets IAX jitter buffer
iax show cache: Display IAX cached dialplan
iax show channels: Show active IAX channels
iax show peers: Show defined IAX peers
iax show registry: Show IAX registration status
iax show stats: Display IAX statistics
iax show users: Show defined IAX users
init keys: Initialize RSA key passcodes
show keys: Displays RSA key information

A.2.5 SIP Channel commands

sip debug: Enable SIP debugging
sip no debug: Disable SIP debugging
sip reload: Reload sip.conf (added after 0.7.1 on 2004-01-23)
sip show channels: Show active SIP channels
sip show channel: Show detailed SIP channel info
sip show inuse: List all inuse/limit
sip show peers: Show defined SIP peers (register clients)
sip show registry: Show SIP registration status (when Asterisk registers as a client to a SIP Proxy)
sip show users: Show defined SIP users

A.2.6 Server management

restart gracefully: Restart Asterisk gracefully
restart now: Restart Asterisk immediately

```

restart when convenient: Restart Asterisk at empty call volume
reload: Reload configuration
stop gracefully: Gracefully shut down Asterisk
stop now: Shut down Asterisk immediately
stop when convenient: Shut down Asterisk at empty call volume
extensions reload?: Reload extensions ONLY
unload: Unload a dynamic module by name
show modules: List modules and info about them
show uptime: Show uptime information
show version: Display Asterisk version info

```

A.3 ASTERISK SPECIAL EXTENSIONS

The following are some Asterisk Special Extensions that are useful to know when creating your Dial Plan or Special Custom Context.

- i** = invalid, jump here when an invalid number is dialled.
- t** = timeout, jump here when we timeout.
- h** = hang-up, jump here when a call 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
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

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

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

[fi]
ringcadence = 1000,4000
dial = 425
busy = 425/300,0/300
ring = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = 425/150,0/150,425/150,0/8000

dialrecall = 425/650,0/25
record = 1400/500,0/15000
info = 950/650,0/325,950/325,0/30,1400/1300,0/2600
stutter = 425/650,0/25

[fr]
ringcadence = 1500,3500
dial = 440
busy = 440/500,0/500
ring = 440/1500,0/3500
congestion = 440/250,0/250
callwait = 440/300,0/10000
dialrecall = !350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330
stutter =
!440/100,!0/100,!440/100,!0/100,!440/100,!0/100,!440/100,!0/100,!440/100,!0/100,440/10
0,!0/100,440

[gr]
ringcadence = 1000,4000
dial = 425/200,0/300,425/700,0/800
busy = 425/300,0/300
ring = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = 425/150,0/150,425/150,0/8000
dialrecall = 425/650,0/25
record = 1400/400,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000,0
stutter = 425/650,0/25

[hk]
ringcadence = 400,200,400,2000
dial = 350+440
busy = 480+620/500,0/500
ring = 440+480/400,0/200,440+480/400,0/2000
congestion = 480+620/250,0/250
callwaiting = 440/100,0/4000
dialrecall = 300/1500,0/500
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330

[hu]
ringcadence = 1250,3750
dial = 425
busy = 425/300,0/300
ring = 425/1250,0/3750
congestion = 425/300,0/300
callwaiting = 425/40,0/1960
dialrecall = 425+450
record = 1400/400,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000,0

stutter = 350+375+400

[il]

; Source: libtonezone.
description = Israel

ringcadence = 1000,3000
dial = 414
busy = 414/500,0/500
ring = 414/1000,0/3000
congestion = 414/250,0/250
callwaiting = 414/100,0/100,414/100,0/100,414/600,0/3000
dialrecall = !414/100,!0/100,!414/100,!0/100,!414/100,!0/100,414
record = 1400/500,0/15000
info = 1000/330,1400/330,1800/330,0/1000
stutter =
!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160
0,!0/160,!414/160,!0/160,!414/160,!0/160,!414/160,!0/160,414

[in]

; Source: libtonezone.
description = India

ringcadence = 400,200,400,2000
dial = 400*25
busy = 400/750,0/750
ring = 400*25/400,0/200,400*25/400,0/2000
congestion = 400/250,0/250
callwaiting = 400/200,0/100,400/200,0/7500
dialrecall = !350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330,0/1000
stutter =
!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440

[it]

ringcadence = 1000,4000
dial = 425/200,0/200,425/600,0/1000
busy = 425/500,0/500
ring = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = 425/400,0/100,425/250,0/100,425/150,0/14000
dialrecall = 470/400,425/400
record = 1400/400,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!0/1000,400/330,!1800/330,!0/1000,0
stutter = 470/400,425/400

[jp]

; Source: libtonezone.
description = Japan

ringcadence = 1000,2000
dial = 400
busy = 400/500,0/500
ring = 400+15/1000,0/2000

```
congestion = 400/500,0/500
callwaiting = 400+16/500,0/8000
dialrecall = !400/200,!0/200,!400/200,!0/200,!400/200,!0/200,400
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330,0
stutter =
!400/100,!0/100,!400/100,!0/100,!400/100,!0/100,!400/100,!0/100,!400/10
0,!0/100,400
```

```
[lt]
ringcadence = 1000,4000
dial = 425
busy = 425/350,0/350
ring = 425/1000,0/4000
congestion = 425/200,0/200
callwaiting = 425/150,0/150,425/150,0/4000
dialrecall = 425/500,0/50
record = 1400/500,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000,0
stutter =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/10
0,!0/100,425
```

```
[mx]
ringcadence = 2000,4000
dial = 425
busy = 425/250,0/250
ring = 425/1000,0/4000
congestion = 425/250,0/250
callwaiting = 425/200,0/600,425/200,0/10000
dialrecall = !350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440
record = 1400/500,0/15000
info = 950/330,0/30,1400/330,0/30,1800/330,0/1000
stutter =
!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!35
0+440/100,!0/100,!350+440/100,!0/100,350+440
```

```
[nl]
ringcadence = 1000,4000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/4000
congestion = 425/250,0/250
callwaiting = 425/500,0/9500
dialrecall = 425/500,0/50
record = 1400/500,0/15000
info = 950/330,1400/330,1800/330,0/1000
stutter = 425/500,0/50
```

```
[no]
ringcadence = 1000,4000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/4000
```

congestion = 425/200,0/200
callwaiting = 425/200,0/600,425/200,0/10000
dialrecall = 470/400,425/400
record = 1400/400,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000,0
stutter = 470/400,425/400

[nz]
ringcadence = 400,200,400,2000
dial = 400
busy = 400/250,0/250
ring = 400+450/400,0/200,400+450/400,0/2000
congestion = 400/375,0/375
callwaiting = !400/200,!0/3000,!400/200,!0/3000,!400/200,!0/3000,!400/200
dialrecall = !400/100!0/100,!400/100,!0/100,!400/100,!0/100,400
record = 1400/425,0/15000
info = 400/750,0/100,400/750,0/100,400/750,0/100,400/750,0/400
stutter =
!400/100!0/100,!400/100,!0/100,!400/100,!0/100,!400/100!0/100,!400/100,!0/100,!400/100,
!0/100,400

[pl]
ringcadence = 1000,4000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/4000
congestion = 425/500,0/500
callwaiting = 425/150,0/150,425/150,0/4000
dialrecall = 425/500,0/50
record = 1400/500,0/15000
info =
!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1400/330,!1800/330,!0/1000,!950/330,!1
400/330,!1800/330,!0/1000
stutter =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/10
0,!0/100,425

[pt]
ringcadence = 1000,5000
dial = 425
busy = 425/500,0/500
ring = 425/1000,0/5000
congestion = 425/200,0/200
callwaiting = 440/300,0/10000
dialrecall = 425/1000,0/200
record = 1400/500,0/15000
info = 950/330,1400/330,1800/330,0/1000
stutter =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/10
0,!0/100,425

[ru]
ringcadence = 800,3200
dial = 425

busy = 425/350,0/350
ring = 425/800,0/3200
congestion = 425/350,0/350
callwaiting = 425/200,0/5000
dialrecall = !350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440
record = 1400/500,0/15000
info = !950/330,!1400/330,!1800/330,0

[se]
ringcadence = 1000,5000
dial = 425
busy = 425/250,0/250
ring = 425/1000,0/5000
congestion = 425/250,0/750
callwaiting = 425/200,0/500,425/200,0/9100
dialrecall = !425/100,!0/100,!425/100,!0/100,!425/100,!0/100,425
record = 1400/500,0/15000
info =
!950/332,!0/24,!1400/332,!0/24,!1800/332,!0/2024,!950/332,!0/24,!1400/332,!0/24,!1800/3
32,!0/2024,!950/332,!0/24,!1400/332,!0/24,!1800/332,!0/2024,!950/332,!0/24,!1400/332,!0/
24,!1800/332,!0/2024,!950/332,!0/24,!1400/332,!0/24,!1800/332,0
stutter =
!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/100,!0/100,!425/10
0,!0/100,425

[sg]
ringcadence = 400,200,400,2000
dial = 425
ring = 425*24/400,0/200,425*24/400,0/2000 ; modulation should be 100%, not 90%
busy = 425/750,0/750
congestion = 425/250,0/250
callwaiting = 425*24/300,0/200,425*24/300,0/3200
stutter =
!425/200,!0/200,!425/600,!0/200,!425/200,!0/200,!425/600,!0/200,!425/200,!0/200,!425/60
0,!0/200,!425/200,!0/200,!425/600,!0/200,425
info = 950/330,1400/330,1800/330,0/1000 ; not currently in use acc. to reference
dialrecall = 425*24/500,0/500,425/500,0/2500 ; unspecified in IDA reference, use repeating
Holding Tone A,B
record = 1400/500,0/15000 ; unspecified in IDA reference, use 0.5s tone every 15s
nutone = 425/2500,0/500
intrusion = 425/250,0/2000
warning = 425/624,0/4376 ; end of period tone, warning
acceptance = 425/125,0/125
holdinga = !425*24/500,!0/500 ; followed by holdingb
holdingb = !425/500,!0/2500

[uk]
ringcadence = 400,200,400,2000
dial = 350+440
specialdial = 350+440/750,440/750
busy = 400/375,0/375
congestion = 400/400,0/350,400/225,0/525
specialcongestion = 400/200,1004/300
unobtainable = 400
ring = 400+450/400,0/200,400+450/400,0/2000
callwaiting = 400/100,0/4000
specialcallwaiting = 400/250,0/250,400/250,0/250,400/250,0/5000
creditexpired = 400/125,0/125
confirm = 1400
switching = 400/200,0/400,400/2000,0/400

```
info = 950/330,0/15,1400/330,0/15,1800/330,0/1000  
record = 1400/500,0/60000  
stutter = 350+440/750,440/750
```

```
[us]  
ringcadence = 2000,4000  
dial = 350+440  
busy = 480+620/500,0/500  
ring = 440+480/2000,0/4000  
congestion = 480+620/250,0/250  
callwaiting = 440/300,0/10000  
dialrecall = !350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,350+440  
record = 1400/500,0/15000  
info = !950/330,!1400/330,!1800/330,0  
stutter =  
!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!350+440/100,!0/100,!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
```

APPENDIX B

B.1 – AUSTRALIAN VSPs (TRUNKS)

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

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

B.1.1 BBPglobal

Peer Details

Trunk Name: bbpglobal

```
allow=g723&gsm
authuser=<user number>
disallow=all
fromdomain=sip2.bbpglobal.com
fromuser=<user number>
host= sip2.bbpglobal.com
insecure=very
secret=<password>
type=peer
username=<user number>
```

User Details

User Context: <user number>

```
context=from-trunk
fromuser=<user number>
insecure=very
secret=<password>
type=user
username=<user number>
Note: You may need to include the
following in your sip.conf
Maxexpiry=3600
Qualify=yes
```

Register String:

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

B.1.2 Engin BYO

Peer Details

Trunk Name: engin

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

User 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=xxxxxx
```

User Details

User Context: xxxxxx

```
auth=md5
context=from-trunk
host=iax.faktortel.com.au
qualify=3000
secret=<password>
type=friend
username=xxxxxx
```

Register String:

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

Faktortel supports the following codecs: ulaw, alaw, ilbc, gsm, g729. The following is the link on the discussions for setting up Faktortel on TRIXBOX.

<http://www.faktortel.com.au/support-trixbox-settings.shtml>

B.1.4 iPrimus

Primus is a strange animal. To get it to work with Trixbox requires me to do a bit of detective work. With my partner in crime, Colin, we set out to play detective because we need to obtain the correct credential to get Trixbox to register and to be able to make and receive calls through Trixbox. Primus Telecom has refused to assist.

To get the necessary credentials, we cheated a little as the method was published in the whirlpool forum <http://forums.whirlpool.net.au/forum-replies.cfm?t=604015>

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

https://spconfig.voip.iprimus.net.au/SoftphoneConfigurationServer/Default.aspx?Action=LOG_IN&username=username&password=password&computerid=db7b351e57af&MAC=your_NIC_MAC_address

Replace the bold type with your Talk Broadband username and password given to you by iPrimus. You can get your NIC MAC address by running **ipconfig /all**.

When you submit that address, the following data will be displayed in your browser.

```
[Data] Success=1 [Settings] proxies:proxy0:display_name="02*****"
proxies:proxy0:username="AU_*****" proxies:proxy0:password="*****"
system:auto_update:config_server_url="https://spconfig.voip.iprimus.net.au/Soft
phoneConfigurationServer/Default.aspx?Action=CONFIG&Username=username&Version=1
&MAC=$MAC$&computerid=$computerid$" audio:headset:audio_in_agc_enabled="0"
audio:speaker_phone:audio_in_agc_enabled="0" codecs:speex_fec:enabled="0"
codecs:speex_wb:other_bandwidths="24000,26000,28800,32800,40000,44000"
codecs:speex_wb:starting_bandwidth="44000"
codecs:speex_wb:usable_bandwidths="20800,24800,28800,36800,44000"
codecs:speex_wb_fec:enabled="0" proxies:proxy0:contacts_etag=""
proxies:proxy0:contacts_last_upload_failed="0" proxies:proxy0:contacts_mod="0"
proxies:proxy0:default_session_refresh_interval_in_seconds="60"
proxies:proxy0:digit_map="#1\`a\`T;match=1;prestrip=2;"
proxies:proxy0:domain="asnjl1a2.bw.iprimus.net"
proxies:proxy0:enable_session_timer="0" proxies:proxy0:enabled="1"
proxies:proxy0:ice_enabled="1"
proxies:proxy0:maximum_time_before_reregistration_in_seconds="1800"
proxies:proxy0:minimum_time_before_reregistration_in_seconds="20"
proxies:proxy0:override_outbound_proxy="1"
proxies:proxy0:override_sip_listen_port="0"
proxies:proxy0:override_stun_server="1" proxies:proxy0:port_range_enable="0"
proxies:proxy0:proxy="sbc01a.syd.voip.iprimus.net.au:15060"
proxies:proxy0:register="1" proxies:proxy0:reregister_in_seconds="3600"
```

Note the bold highlights. These are the data that you need for your trunk setting. Armed with the data above, you may now set your trunk setting.

Peer Details

Trunk Name: iPrimus

```
allow=alaw&ulaw
Disallow=all
fromdomain=asnjl2.bw.iprimus.net
fromuser=AU_XXXXXXX
host=asnjl2.bw.iprimus.net
insecure=very
outboundproxy=sbc01a.syd.voip.iprimus.net.au
port=15060
secret=YYYYYYYY
type=peer
username=AU_XXXXXXX
```

User Details

User Context: **02*******

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

Register String:

AU_XXXXXXX@asnjl2.bw.iprimus.net:YYYYYYYY:AU_XXXXXXX@iprimus/02*****

B.1.5 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.1.6 MyNetFone

Peer Details

Trunk Name: MyFonesip

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

User Details

User Context: 091xxxxx

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

Register String:

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

For further information on MyNetFone, you may refer to the following link:

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

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

B.1.7 Nehos (IAX)**Peer Details**

Trunk Name: Nehos

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

User Details

User Context: 661XXXX

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

Register String:

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

B.1.8 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.1.9 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.1.10 TelephoneGlobal

Peer Details

Trunk Name: Teleglobal

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

User Details

User Context: 2222xxxxxxxxxx

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

Register String:

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

B.2 – NEW ZEALAND VSPs (TRUNKS)

B.2.1 iTalk

Peer Details

Trunk Name: italk

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

User Details

User Context:

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

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

Register String:

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

(Source Ref: Steve Biddle)

B.2.2 Double Zero 64 (IAX)

Peer Details

Trunk Name: 0064

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

User Details

User Context:: Your_Customer_ID

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

Register String:

```
Your_Customer_ID: <your password>@voip.0064.co.nz
```

B.3 – OTHER VS^PS (TRUNKS)

B.3.1 Broadvoice

Peer Details

Trunk Name: Broadvoice

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

User Details

User Context: <usernumber>

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

Register String:

```
<usernumber>@sip.broadvoice.com:<password>:<usernumber>@sip.broadvoice.com/<usernumber>
```

B.3.2 FWD aka Pulver(IAX)

Peer Details

Trunk Name: fwd

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

User Details

User Context: 65xxxx

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

Register String:

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

B.3.3 SipBroker

Peer Details

Trunk Name: sipbroker-out

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

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

You do not need to register either.

```
type=peer
```

If you are using SipBroker, you must ensure that your sip_nat.conf is modified to the following:
nat=yes
externip=<Fix Ip address> ; if you have fix IP **or**
externhost=<your DNS hostname> ; if you are using Dynamic IP
localnet=192.168.1.0/255.255.255.0

B.3.4 [SipPhone](#)

Peer Details

Trunk Name: sippone

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

User Details

User Context: 1747xxxxxxxx

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

Register String:

```
1747xxxxxxxx:<password>@proxy01.sippone.com/1747xxxxxxxx
```

B.3.5 [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.4 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.4.1 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

APPENDIX C

Details for Asterisk Sound Files can be obtained from the following links:

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

C.1 VOICE FOR ASTERISK MESSAGES

Brian, a Whirlpool Enthusiast, posted the following tip in the Whirlpool forum. Brian has asked me to include his discovery in this guide.

The following is an extract of Brian's exploits – verbatim ☺

<Quote>

I have just been setting up the Digital Receptionist and I wanted to use a professional voice for messages. I thought I might detail my exploits here for those that might be considering the same thing.

First you need the original recording either from a real professional or a "text to voice" utility. I went with the Realspeak demo

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

The "British Daniel" voice was deep and professional. Check this out

<http://www.cbservers.com.au/whirlpool.wav>

Now the demo has a 100-character limit so you may need to do 2 or 3 files and join them together with a program like Accoustica MP3 mixer.

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

Save it as a wav file.

It was after this part that I had a lot of problems. Asterisk is very funny about sound files. I tried using all sorts of programs to get the file to the right specs but it failed to work every time. I eventually used "**sox**" an open source command line editing tool and it worked first go (using advice from VoIP-info & digium forums) <http://sox.sourceforge.net/> then at the command line prompt use this syntax:

```
sox winwave.wav -r 8000 -c 1 -s -w linwave.wav
```

you may then upload your message to asterisk.

Some further reading;

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

<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 set-up 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' CONTRIBUTIONS)

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+4XXXXXXX  
02+6XXXXXXX  
02+80XXXXXX  
02+81XXXXXX  
02+82XXXXXX  
02+83XXXXXX  
02+84XXXXXX  
02+85XXXXXX  
02+86XXXXXX  
02+87XXXXXX  
02+89XXXXXX  
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+9XXXXXXX  
0+NXXXXXXX  
0011+1NXXNXXXXXX  
0011+44ZXXXXXXXXX  
0011+49NXXXXXXX.
```

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 20000. 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 boot up. 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 that lead to echo.
4. Set the Impedance on your lines.
 - PSTN -> "Port Impedance" = $220+820||120nF$ as a starting point.
 - Regional -> FXS Port Impedance = " $220+820||115nF$ " as a starting point.
5. In the PSTN tab set –
 - "Tip/Ring Voltage Adjust: = 3.1V" and
 - "Operational Loop Current Min = 16mA".

Doesn't seem to affect echo, but I believe that these are the correct numbers for Australia.

6. Turn down the jitter buffers!
 - "PSTN -> Network Jitter Level: = low",
 - "PSTN -> Jitter Buffer Adjustment: = disable".

This reduces the delay across your SPA-3000.

- "LINE1 -> Network Jitter Level: = low",
- "LINE1 -> Jitter Buffer Adjustment: = up and down".

If you are using a poor quality VoIP service as well as the PSTN then you could change the

- "LINE1 -> Network Jitter Level: = medium".

7. Set the preferred codec for the PSTN to be g711a and lock it in.
 - "PSTN -> Preferred Codec = g711a",
 - "PSTN -> Use Pref Codec Only = yes".

Obviously adjust this if you're accessing your PSTN line via VoIP from a remote network. Set the LINE1 to allow g711a as well as whatever else you 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 you're probably sending too 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 too much, the SPA sorts does a double negative and turns it way up. I believe the range of this variable is about -127 -> 127 (from my testing). Turn it down, down, down, down until the person can still hear you but reduced echo.

Note: if you enable "Echo Supp Enable" then you will negate these parameters. It seems that the Sipura echo suppression is actually just an automatic gain control. It's really annoying - leave it off.

11. Make a test call to someone with a known good phone out via the SPA's PSTN line or get someone to call in to the PSTN line. Best if it's just a boring old Telstra phone hard wired to a socket on the wall. Don't call a mobile!

If the remote party is hearing echo, it could be that your phone is so loud that it's feeding back into the microphone. Turn down the PSTN To SPA Gain until you can comfortably 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 mismatch 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 canned 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 reappears, 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
fxscls=2
fxscls=3
fxscls=4

# Global data

loadzone = au
defaultzone = au
```

zapata_additional.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  
threeewaycalling=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
```

E.7 RAID1, LVM (LOGICAL VOLUMES) - HOWTO

(Submitted by a user: Rehan)

Introduction

According to Digium recommendations Asterisk boxes should be IDE based. This is counter to what one would expect. However if you are using Zaptel drivers then a fast

hard disk or hardware RAID might ‘steal’ interrupts from the Zaptel drivers. Additionally many hardware RAID solutions set themselves up on the PCI bus with timings that can delay voice traffic to your voice cards.

(See note 3 at the end if you want to use scsi or sata drives)

For this particular application (A PBX) we are not too concerned about disk performance. Much of the software will run in memory on a 300Mb Ram machine (Trixbox default install consumes no more than 200Mb including the file cache). Fast hard discs improve performance in systems that are heavily dependant on a swap file. Linux will keep everything in memory + a reasonable file cache as much as possible. CPU performance and making sure you have sufficient memory to run the PBX are the key performance indicators of a healthy PBX.

We wouldn’t want to put an Asterisk box in production without some kind of resiliency to survive continuous use over perhaps many years. To use an IDE disk based system we can use software RAID to provide some protection and to make sure the PBX stays up and running. The Raid subsystem can also email you if a drive fails.

The following procedure describes setting up a software RAID during installation on all partitions of a Trixbox. This applies to a vanilla install from the Distribution CDROM.

We will be using RAID and also Logical Volume Management. Logical Volume Management gives us the ability to dynamically resize drives and dynamically add disk space on a live system without having to delete or create new partitions. So if your requirements change in the future you can easily accommodate them.

The issues:

1. Implementing RAID and logical volumes post-install is complicated, dangerous and not recommended.
2. RAID (hardware or software) and LVM are desirable features. IDE is still the cheapest storage platform. This is especially true of Asterisk installs as many PBX’s use low-end hardware.
3. The default install of Trixbox does not allow any options to install RAID or LVM
4. The text mode Anaconda install does not allow manual configuration of Logical volumes however a kickstart file does.
5. A kickstart install of RAID will always (!!!!) produce file system corruption if the software RAID is not allowed to complete its first sync before the first reboot.
6. High end hardware RAID is still relatively expensive. Linux drivers are not always available or if available, not always stable.

Point 5 will many times dissuade people from using software RAID and LVM. As long as we can accommodate the first sync then we will have no problems.

Ok, lets get started:

- 1) We’ll need to set-up the drives in a particular order:

Hardware configuration:

HD 1 connected to primary IDE as master.
HD 2 connected to secondary IDE as master.
CDROM connected to secondary IDE as slave.

The hard disks don't need to be exactly the same size. Actually using this method the disks don't need to be the same size at all. I would recommend that the smallest disk be at least 5Gigs. The Raid will configure itself to the smallest disk and the larger disk will match up to the smaller disk's partitions with any extra space left unused.

Although I worked this for IDE drives it can be used on any type of disk subsystem - SCSI, sata etc with some modification for the drive labels.

2) Extract/Edit the CDROM:

Windows:

I used UltraISO on windows to modify the CDROM. Open the ISO with UltraISO and copy the ks.cfg file somewhere. Remove the read only flag so you can edit it. It is very important you edit the file with an editor that knows how to do Linux linefeeds. I discovered and used Editpad pro for this. The lite (demo) version will suit the needs we have. If your editor is not 100% compatible with Linux linefeeds after you reboot you will get errors which have directory names with strange characters (/var/trixbox_load^M/ or /var/trixbox_load/R where the /R is actually in the directory name of load not a separate directory)

Linux: (incomplete)

Extract the CDROM to a temporary Directory. Make the file ks.cfg as non-read only. Edit and create the files as below. Use the following command to rebuild the ISO.

```
mkisofs -o modified-trixbox.iso -b isolinux/isolinux.bin -c isolinux/boot.cat -no-emul-boot -boot-load-size 4 -boot-info-table
```

3) Get the Installation configuration file as follows:

We need to create a file that will tell the install program what to do. A copy of the existing file is located in the root of the CDROM. It is called ks.cfg. Make a copy of this file and call it ksRAIDlvm.cfg

4) Edit the ksRAIDlvm.cfg as follows

Delete the line where it says reboot. We don't want it to automatically reboot at the end. ! This is a critical step!

Where it says:

```
#Partition clearing information
clearpart --all -initlabel
part /boot --fstype ext3 --size=100
part / --fstype ext3 --size=1024 -grow
part swap --size=256 --grow --maxsize=768
```

Replace with

```
#Partition clearing information
clearpart --all -initlabel
part raid.01 --size=100 --ondisk=hda -asprimary
part raid.02 --size=100 --ondisk=hdc -asprimary
part swap --size=256 --ondisk=hda
part swap --size=256 --ondisk=hdc
part raid.11 --size=1 --grow --ondisk=hda
```

```
part raid.12 --size=1 --grow --ondisk=hdc
raid /boot --level=1 --device=md0 raid.01 raid.02
raid pv.01 --level=1 --device=md1 raid.11 raid.12
volgroup vg01 pv.01
logvol / --vgname=vg01 --size=10000 --name=root
```

Very important: the `--asprimary` switch must only appear on the `raid.01` and `raid.02` lines.

This will give you a 100MB RAID 1 /boot partition, two 256Mb swap partitions (change the 256 to a desirable size or replace with `--recommended`) and the rest of the disk will be a logical volume (over laying a RAID1) containing the / (or root) filesystem. All partitions will be formatted as Ext3 as normal. If you want to create other partitions to support other uses you can change the `raid.11` and `raid.12` lines to read `--size=xxxx` (in megabytes) and remove the `--grow` option. This will fix the size of the partition you create. After the installation completes you can then go ahead and create more RAID1 partitions and Logical volumes.

Where it says (in the section starting - %post)

```
mount /dev/hda /mnt/source
```

change to

```
mount /dev/md1 /mnt/source
```

Add the following lines to the end of the `%post%` section. This will update the boot information to make both drives bootable. Remember to update the hdx numbers to match your drive configuration.

```
/sbin/grub --batch <<EOT 1>/dev/null 2>/dev/null
root (hd0,0)
setup (hd0)
root (hd1,0)
setup (hd1)
quit
EOT
```

Save the file.

Add it to the root directory of the ISO and save the ISO under a different name.

Burn the new ISO to a CD. Boot your soon to be asterisk machine from this CD and use the following command to start the install.

```
linux ks=cdrom:/ksRAID1lvm.cfg
```

The install process will continue as before however it will create the RAID as we have set-up.

Begin the install. Let it run through to the end. When you see the REBOOT screen STOP. Don't do anything here. We need to wait for the RAIDs to complete their first sync. For IDE disks the only way I have found of reliably checking this is to wait for the drive light to go out. This can take some time depending on the speed of the disks and their size. Go and have a coffee. If you have no other indication here just wait about half an hour/45 mins to be sure.

Once you are sure the disks are in sync. Reboot.

You are done!

Checking and maintaining:

Login as root using the password set-up during the earlier phase.

You can check the state of the RAID by using the following command:

```
cat /proc/mdstat
```

You should have two raids called md0 and md1. In square brackets for each it should say [2/2]. Your disks are ready and Trixbox is installed. The block sizes may be different if you are not using identical disks.

If it doesn't say [2/2] but the machine has restarted ok and it says [2/1] then one or both your raids have not synced properly or your disk is faulty (bad blocks). To manually sync the raids type some commands (for simplicity all the commands are included below. If a partition is already in a Synced RAID the command will fail with an error (in this case the error means success!). This you can ignore and move onto the next command):

```
mdadm -a /dev/md0 /dev/hda1  
mdadm -a /dev/md0 /dev/hdc1  
mdadm -a /dev/md1 /dev/hda3  
mdadm -a /dev/md1 /dev/hdc3
```

use

```
watch cat /proc/mdstat
```

to monitor the raid re-build. Once it is complete you can issue the

reboot

command and restart the machine. Check your raids again with

```
cat /proc/mdstat
```

They should be ok now.

If the machine has not started ok (errors on the startup screen) then the chances are the RAID was not completely built before the restart. The best idea here is to re-install this time waiting longer before rebooting. It is difficult to know which files are not complete.

Notes:

1) In a VMWARE machine everything works fine. (To test this you need to create a second hard drive when creating the virtual machine then do the install booting either off the new CDROM or the new ISO).

2) For anyone who wants to revert (or vert) an existing drive to Raid, here is a good link:

http://linux-sxs.org/hardware/raid_for_idiots.html

Just ignore the last bit and use grub instead (Lilo is too old!!!!).

The following commands achieve the same ends:

```
grub (enter)
```

```
root (hdx0,0)  
setup (hdx0)
```

```
root (hdx1,0)
setup (hdx1)
```

(ctrl-z to exit)

3) This procedure will work for SCSI and SATA drives as well. Bearing in mind the caveats Digium have about using IDE many people have successfully installed RAID 1 on SCSI and SATA drives. To accomplish this replace the HDx with a suitable replacement (usually hd0 is replaced with sd0 and hd1 with sd1).

If you have any issues booting from the RAID they can usually be fixed by booting from the Trixbox CD and starting the Rescue Environment and running some commands:

Boot from the Trixbox CD. At the prompt type:

```
Linux rescue
```

Issue:

```
grub (enter)
```

```
root (hdx0,0)
setup (hdx0)
root (hdx1,0)
setup (hdx1)
```

(ctrl-z to exit)

```
reboot
```

We continue to use the textmode install because it reduces the chances of the install failing. For example if you have a Rage XL video card then using a graphical install, the format of the drives will fail (RAID or no RAID). In textmode logical volumes can only be created in the kickstart file.

If your disk controller needs to have special drivers then the install must be started using the following command:

```
Linux ks=cdrom:/ksRAIDlvm.cfg dd
```

At some point it will ask you for a driver disk. This can usually be obtained from the manufacturers website.

APPENDIX F (PUBLICATIONS AND REFERENCES)

F.1 DUNDI – A GUIDE TO A SIMPLE SETUP

You will find a guide to a simple DUNDi set-up at the GoldfishNet Blog:
<http://blog.thegoldfish.net/dundi-tutorial-for-asteriskhome/>

F.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.

F.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>

F.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

NOTES

WHAT VERSION OF ASTERISK IS INSTALLED?

To get this information, at the Linux command prompt, type the following:

```
rpm -qa | grep asterisk
```

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**

Login as **admin** using the default password of **mypassword**.

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>

DELETE CALL RECORDS FROM CDR

There will be time when you need to delete the CDR records from your call details record. One example is, those calls you made while testing 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 the 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 asked to 'confirm truncate table cdr',
 - If you click Yes at this point the content 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

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 substituted **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 ☺

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

0 | NXXXXXXX

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

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

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.

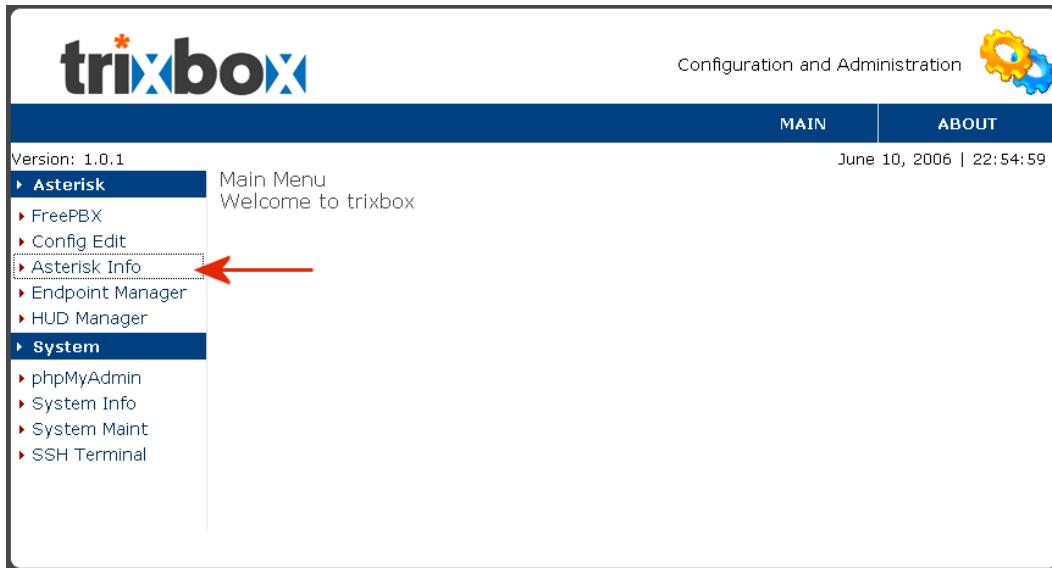
In your Trunk Outgoing Dial Rule, you will need the following pattern.

61+NXXXXXXX

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 TRIXBOX 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 TRIXBOX (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 TRIXBOX. 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 TRIXBOX) in the zip into the **var/www/html/admin/modules** folder of TRIXBOX (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 Asterisk 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.

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 ☺

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 (this has been rectified in the later ISOs).

Note: *This issue has been rectified in later ISOs.*

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.

Note: *This issue has been rectified in later ISOs.*

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.

FreePBX version 2.1.3 does not seem to have this problem.

FREEPBX 2.1.2 CAUSED ASTERISK NOT TO LOAD

Note: *FreePBX 2.1.3 has since been released to overcome this problem and other CDR related problems.*

This is not really a Trixbox bug but rather, it is an issue with upgrading to freePBX 2.1.2.

To upgrade from the existing version of freePBX to the new version 2.1.2, it is recommended (as per the documentation site) to use the following command.

```
cd /usr/src/  
svn co https://svn.sourceforge.net/svnroot/amportal/freepbx/tags/2.1.2 freepbx-2.1.2  
cd freepbx-2.1.2  
.install_amp
```

The result is, you will have the latest freePBX, however, it may also result in your asterisk not starting.

I was lucky enough to speak to Kerry_G on the IRC support and was told that it was caused by app_trunkisavail.so call, which is a common error.

Kerry provided the following workaround:

Log into the system and edit /etc/asterisk/modules.conf

```
nano /etc/asterisk/modules.conf
```

Add the following line:

```
noload => app_trunkisavail.so
```

Hit Ctrl-o, then Ctrl-x to save and exit. Then restart amportal.

```
amportal restart
```

The other option is to just delete the module. This is the option recommended by Rob Thomas, our freePBX Lead Developer.

```
rm /usr/lib/asterisk/modules/app_trunkisavail.so
```

in the [global] section add the following (at the bottom of the file):

```
chan_zap.so => yes
```

Then reboot Trixbox.

Your modules.conf file should look like this.

modules.conf

```
;  
; Asterisk configuration file  
; Module Loader configuration file  
;  
[modules]  
autoload=yes  
;  
; If you want, load the GTK console right away.  
; Don't load the KDE console since  
; it's not as sophisticated right now.  
;  
;  
;noload = app_trunkisavail.so ; no need to do this as app_trunkisavail.so is deleted  
noload => pbx_gtkconsole.so  
;load => pbx_gtkconsole.so  
noload => pbx_kdeconsole.so  
;  
; Intercom application is obsoleted by  
; chan_oss. Don't load it.  
;  
noload => app_intercom.so  
;  
; DON'T load the chan_modem.so, as they are obsolete in * 1.2  
  
noload => chan_modem.so  
noload => chan_modem_aopen.so  
noload => chan_modem_bestdata.so  
noload => chan_modem_i4l.so  
  
;  
; Ensure that format_* modules are loaded before res_musiconhold  
;load => format_ogg_vorbis.so  
load => format_wav.so  
load => format_pcm.so  
load => format_au.so  
;  
; This isn't part of 'asterisk' iteslf, it's part of asterisk  
; addons. If this isn't installed, asterisk will fail to start.  
;  
; But it does need to go here for native MOH to work using mp3's.  
load => format_mp3.so  
load => res_musiconhold.so
```

```

;
; Load either OSS or ALSA, not both
; By default, load OSS only (automatically) and do not load ALSA
;
noload => chan_alsa.so
noload => chan_oss.so
;
; Module names listed in "global" section will have symbols
globally
; exported to modules loaded after them.
;
[global]
chan_zap.so => yes ; you may need to add this switch

```

TRIXBOX 1.2 DISTRIBUTION MAJOR BUG

Every time Asterisk is reloaded due to creating a new extension or trunks or whatever else one does on freePBX (including editing .conf) files will result in Asterisk loosing all SIP connections, extensions as well as trunks. As such you will not be able to make or receive calls.

When this happens, you are required to restart amportal from the console (or SSH) using the following command.

```
amportal restart
```

It seems that this is related to the Asterisk distribution 1.11.x used by freePBX and Trixbox within the Trixbox version 1.2 distribution ISO. This has been fixed in the later revision release of Trixbox or Asterisk.

ASTERISK RE-LOAD ISSUE

After making changes to Trunks, IVR or DISA etc, the changes do not translate to the system. This is due to an inherent bug in the Asterisk binary in Trixbox 1.2.

If this happens, run the following at the command prompt to force the reload:

```
asterisk -rx restart now
```

If the problem is related to Asterisk, this should get you going otherwise, it may be some other gremlin that has gotten into the system.

ARE YOU STILL HAVING PROBLEM?

If you are still having problem installing Trixbox 1.2, it probably is time for you to do some manual configuration as described in Rob Thomas' blog.

Here is the link and as I have personally done it this way, I can assure you that it will solve your problem – I hope ☺

<http://www.freepbx.org/2006/09/28/un-trixbox-your-trixbox/#more-7>

HANGS ON SHUTDOWN

Using X100P card caused Trixbox to hang on shutdown in Trixbox 2.0beta2. This happened to me on 2 different make X100P. This does not happen with TDM400 card.

To avoid this hanging anomaly amportal stop must be executed before a reboot or shutdown.

The following is the hack to stop the panic from occurring:

Log in as root:

Go to the /etc/rc.d directory.

```
cd /etc/rc.d
```

Rename all of the K92zaptel files to K95zaptel so that they are invoked after the K94asterisk scripts:

```
mv rc0.d/K92zaptel rc0.d/K95zaptel  
mv rc1.d/K92zaptel rc1.d/K95zaptel  
mv rc2.d/K92zaptel rc2.d/K95zaptel  
mv rc4.d/K92zaptel rc4.d/K95zaptel  
mv rc6.d/K92zaptel rc6.d/K95zaptel
```

```
shutdown -r now
```

The system will now shutdown or reboot without a panic.

CHEAT SHEET

amportal restart	To restart Trixbox after configuration change without rebooting. However certain changes will need rebooting.
asterisk -rvvvv	To get to asterisk CLI
service asterisk restart	To restart asterisk if required when something goes wrong
service network restart	To restart network service if connection goes down and did not come up again even though internet has been restored.
sip reload	To reload SIP. This command is to be run from Asterisk CLI
Enable Call waiting by default	Edit ENABLECW=yes in amportal.conf

EXPLOIT ALERTS

According to Computerworld, a vulnerability in the Asterisk PBX server that enables an attacker to gain complete control of a PBX system has been discovered by an Australian and New Zealand security outfit.

This was also highlighted by Ward Mundy of Nerd Vittle who said that he exploit allows an attacker to spoof caller-IDs, sniff voice calls on the network and take complete control of the system. This vulnerability has been identified in all versions of Asterisk included in the Trixbox and Asterisk@Home releases except the new Asterisk 1.4 (currently still under development).

An attacker who can connect to the Asterisk server SCCP “Skinny” port (by default 2000/tcp) can attack the vulnerable function prior to registering as a configured Skinny phone, permitting pre-authentication remote compromise and remote code execution as the root user.

To overcome this, simply block port 2000 tcp traffic through your firewall. Otherwise, If your server is in the DMZ or connected to the Internet directly, then you should disable the chan_skinny module. To do this, edit /etc/asterisk/modules.conf and add the following line in the [modules] context.

```
noload => chan_skinny.so
```

Then restart Asterisk.

```
amportal restart
```

Note: *This module is primarily used to talk to Cisco phones in their native language, i.e. non-SIP.*

LIVE CASE STUDY

(Placeholder for live case study in respect of Medical Centre Installation. To be included at a later date pending approval where some details has to be edited out).

BIBLIOGRAPHY

The real people behind this guide (Real Names used or aliases if real names unknown): I have also made references to a number of other online publications and forums whenever I need confirmations and further clarifications. Some of the information in this document were sourced from these people and places.

Andrew Gillis	<i>The man who started it all – The Mahaguru</i>
Asterisk Guru	http://www.asteriskguru.com
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 can't 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>
voip-info.org	<i>A site I often refer to when I require further clarification.</i>
VoipShop	<i>Source for Australian voice and others</i>
Voxilla.com	http://voxilla.com
Ward Mundy	<i>My hero http://mundy.org my place of salvation ☺.</i>

CHANGES MADE SINCE LAST UPDATE

Included indications for the following countries:

India
Israel
Japan

[Solution](#) to asterisk not loading after upgrading to freePBX 2.1.2

Instruction to install the [free codecs](#).

Instruction for [Multi-Language IVR](#)

[Multi-Tenants](#) application