trixbox CE v2.6.2

Complete setup guide for a small business

Document v1.1 - Last updated by Chris Sherwood, SureTeq, Inc.

trixbox CE v2.6.2 is an all-inclusive Asterisk PBX solution that comes on a bootable CD. It makes the process of bringing up a VoIP PBX solution a piece of cake. This document details, step by step, how to install and configure trixbox CE v2.6.2 for a small business. It includes information on how to set up extensions, incoming and outgoing phone calls, and other useful applications.

SureTeq is now an FtOCC authorized trixbox support vendor. We can provide remote support and troubleshooting, or we can design, implement and support your company's complete VoIP PBX solution.

Let SureTeq provide you with FtOCC certified trixbox Pro and/or trixbox CE **hourly support**! Hourly rates and support contracts are available. For pricing information and contact details, please contact us!

- SureTeq now has a blog/RSS feed! Subscribe to our blog for tips, tricks, news, and general information about all things Asterisk/trixbox.

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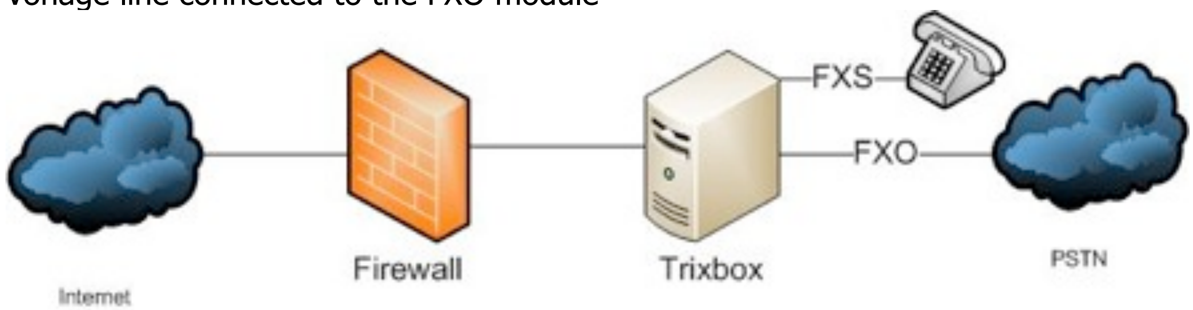
1.0 - Hardware platform used in the creation of this document

Installing on a Dell Dimension 9150
Intel Pentium D CPU (2.8GHz dual core)
2GB RAM
Zaptel card:

Digium TDM 400P with 1 FXO and 1 FXS module
Vonage line connected to the FXO module



1.1 - What's new in v2.6.2?

There are some additions and changes to trixbox CE v.2.6.1 (and now v2.6.2). The major changes are listed below (and here for 2.6.1, here for 2.6.2):

v2.6.2

New Linux Kernel - trixbox CE v2.6.2 now uses the v2.6.18-92 Linux kernel...what does this mean? More NIC support!

Fixed issues with bulk extensions

Improved CDR reports

Updated endpoint manager w/ all certified phones

v2.6.1

Setting up PSTN cards - to configure Digium and Openvox cards, users should now use the 'setup-pstn' script. For Rhino, first run 'setup-rhino,' reboot, and then run 'setup-pstn.' For Sangoma, install the wanpipe utils and driver, reboot, and then run 'setup-sangoma.'

netconfig is now gone - CentOS 5 has deprecated the netconfig script and it has been replaced with the 'system-config-network' command. This is only for those who configure their network settings via the command line.

Postfix - Due to popular demand, Postfix is now the default mail transport in trixbox CE. Postfix is quite simple to setup and manage. Along with this change, there is now a General Settings module which has basic mail settings for relaying and authentication. From the Linux CLI, there are now three new scripts:

setup-mail - walks you through setting up Postfix and will help set up a relay server.
install-sendmail - for those who prefer sendmail, this script will switch you.
install-postfix - if you are upgrading trixbox CE from a previous version, the upgrade does not automatically switch you to Postfix...but this script will.

SNOM support - the Endpoint Manager now supports all SNOM phones including the M3 DECT phone.

OpenVox support - Expanded support for OpenVox cards is now included.

Expanded Aastra support - Support in the Endpoint Manager was added for Aastra 5 and 9 series phones and the Aastra XML services are now installed by default.

Cohesive Interface - Fonality is making strides to provide a much more cohesive user interface that will look more professional and integrated. All references to FreePBX and Asterisk have been removed. The Asterisk menu is now called 'PBX' and the FreePBX menu is now called 'PBX Configuration.'

Bulk extensions - A new module is now included that will allow you to import a CSV file for bulk extension creation.

2.0 - Download and install trixbox CE v2.6.2

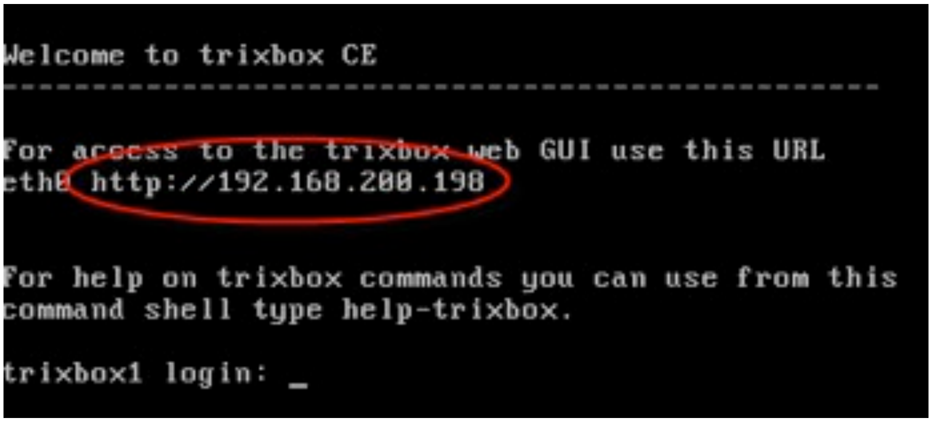
Download the trixbox v2.6.2 ISO file by going to http://trixbox.org/downloads. For this installation I am using v2.6.2.1 RC1, so I can not verify if anything will be different for the actual released version.
Burn the ISO to a CD and boot to it.
***** WARNING *****
This CD will completely destroy whatever data is on the computer you are booting to. Make sure this is what you want to do before proceeding.
When Trixbox splash screen opens, hit Enter. The installation process will start.
Choose a language and hit OK.
Select appropriate time zone and hit OK.
Type in your root password twice and hit OK. The stronger the better for the root password...most compromised Linux boxes are because of weak passwords!
System will now format the hard drive and install trixbox CE v2.6.2.
When the install finishes, the CD tray will open. Remove the CD (or else it will start over). System will reboot once, and then you will end up at the login prompt.
*** **NOTE:** If the system does reboot with the CD still in, you can simply remove the CD and press CTRL+ALT+DEL at the trixbox splash screen...it will continue normally upon the next reboot.
Once the install has finished, log in as root with the root password you entered above.

3.0 - Setting up your server

3.1 - Network configuration

GUI config:

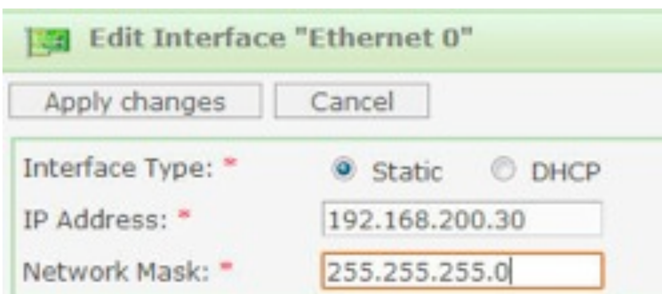
trixbox CE v2.6.2 has a GUI tool for configuring your network settings. Open up a browser and point it to your Trixbox IP (you should see the DHCP'd IP address at the 'Welcome to trixbox CE' screen or by logging in and running '`ifconfig`' in the Linux CLI).



Once you have the trixbox GUI open, switch into Admin mode by clicking 'User mode [switch].' Use logon 'maint / password' (we will change this default password soon). Then click on 'System' --> 'Network.'

*** **NOTE:** If you receive the registration screen, you can choose to register your trixbox. Registering your trixbox gives you a 5% discount at the trixbox store. You can always visit the registration later by choosing Settings --> Registration from the main page.

Click on 'Ethernet 0' (or your desired network card) to change the IP address settings. Click 'Apply changes' to save your changes. **If you changed your IP address, you will have to re-open your browser with the new IP address settings.**



To change other settings such as host name, DNS servers, or gateway, click on 'Edit Network Parameters' and make your desired changes. Click 'Save' when finished.

3.2 - Install Webmin

Webmin is a valuable tool used for the configuration of Linux-based servers. I install Webmin by default on all of my Linux boxes simply due to it's ease of use. Webmin installs an HTTP-based GUI which you can get to by using port 10000 from a browser. To install Webmin, you need to run the following commands in your Linux CLI:

```
cd /usr/src
wget http://prdownloads.sourceforge.net/webadmin/
webmin-1.441-1.noarch.rpm
rpm -ivh webmin-1.441-1.noarch.rpm
```

*****NOTE:** The version of webmin available at the time this document is being written is v1.441-1. You may have to adjust the lines above if a newer version is available.

Once the install has finished, you can get to your Webmin console by putting the following into a browser that exists on your LAN:

https://192.168.200.30:10000 (obviously, replace my IP with your trixbox IP, and note that this is HTTPS...not HTTP). The login is root and your root password.

4.0 - Zaptel Driver

Setting up your Zaptel driver has typically been a problem with Trixbox v1.1 and earlier, however now, the setup is done automatically during installation if you have a Digium analog port card. If you have a PRI card (or cards), you will have to set this up manually...I do not include documentation on how to do so in this document, however you can see sample setups by going to www.voip-info.org and searching for 'zaptel.conf' and 'zapata.conf.'

From command line do '`ztcfg –vvv`' – if there are no errors, you should get an output that says:

```
Zaptel Configuration
======================

Channel map:

Channel 01: FXO Kewlstart (Default) (Slaves: 01)
Channel 04: FXS Kewlstart (Default) (Slaves: 04)

2 channels configured.
```

Or similar based on your Zaptel hardware configuration. The above output is from my Digium TDM400 with 1 FXO and 1 FXS.

***** NOTE:** If there are no channels showing up when doing '`ztcfg -vvv`', try running '`setup-pstn`' at the Linux CLI.

If you are using Sangoma equipment, go to the package manager and install the wanpipe-modules and wanpipe-util packages. Once installed, run '`setup-sangoma`.'

If you are running Rhino equipment, run '`setup-rhino`' at the Linux CLI.

5.0 - General Trixbox Setup/Security

Now it is time to configure Asterisk/trixbox.

Change default passwords:

Before we do anything, let's change our passwords. We need to do this at the Linux CLI.

Update maint password by typing ‘`passwd-maint`’ at the command line. Enter the password twice.

PBX Settings manager password:

To change the PBX Settings manager pass, you need to edit two separate files and put in the new password.

```
nano /etc/asterisk/manager.conf
```

Find 'secret = amp111' under the [admin] section. Change 'amp111' to your new desired password. CTRL+X followed by 'Y' to save and exit.

Now, we need to edit the /etc/amportal.conf to use our new password.

```
nano /etc/amportal.conf
```

Find the line that says 'AMPMGRPASS=amp111' and change the 'amp111' to the new password you just set. CTRL+X followed by 'Y' to save and exit.

```
amportal restart
```

***** NOTE:** I have found out the hard way that PBX Settings does not like having an exclamation point (!) in the admin password. There may be other special characters that it doesn't like also.

MySQL passwords:

Update the MySQL asteriskuser password by doing the following at the Linux CLI:

```
mysql -u root -p
```

Enter in the default mysql root password '`passw0rd`' (with a zero).

```
mysql> use mysql;
mysql> update user set password=PASSWORD("new_pass_here") where
User='asteriskuser';
mysql> flush privileges;
mysql> quit
```

Replace 'new_pass_here' with your desired password.

Now, we need to edit the /etc/amportal.conf to use our new password.

```
nano /etc/amportal.conf
```

Find the line that says 'AMPDBPASS=amp109' and change the 'amp109' to the new password you just set. CTRL+X followed by 'Y' to save and exit.

*****NOTE:** NEW and improved confusion for v2.4 and above! In the amportal.conf, there are now TWO places to change the amp109 password...near the top of the file, and at the very end. The one at the top is commented out with a #, so changing that one doesn't really do anything...but to be consistent, make sure you change the password in both places.

You will also need to update the password in the /etc/asterisk/cdr_mysql.conf file for call detail records (cdr) and in /etc/asterisk/res_mysql (who knows why).

```
nano /etc/asterisk/cdr_mysql.conf
```

Find the line that says 'password=amp109' and change the 'amp109' to the new password you just set. CTRL+X followed by 'Y' to save and exit.

Once done, run these commands from the Linux CLI:

```
service mysqld restart
amportal restart
```

***** NOTE:** I have found out the hard way that PBX Config/MySQL does not like having an exclamation point (!) in the password. There may be other special characters that it doesn't like also.

It is also a good idea to change the default mysql root user password. To change the default mysql root password, do the following:

```
mysqladmin -u root -p password new_password_here
```

Replacing 'new_password_here' with your desired password. When you hit 'Enter' you'll be prompted for a password...enter in the default mysql root user password of 'passw0rd' (with a zero)

Now change the root mysql password (for Web Meetme) in the /etc/asterisk/cbmysql.conf:

```
nano /etc/asterisk/cbmysql.conf
```

Change the password= to your new mysql root password. CTRL+X followed by 'Y' to save and exit.

Now change the root mysql password (again for Web Meetme) in the /var/www/html/web-meetme/lib/database.php file:

```
nano /var/www/html/web-meetme/lib/database.php
```

change `$password = 'passw0rd';` to `$password = '(your new mysql root password)';`. CTRL+X followed by 'Y' to save and exit.

Next, change your asteriskuser password for mysql CDR's (Call Detail Records):

nano /var/www/html/maint/modules/cdrreport/config/database.php

Find the line that says:

```
'connection' => 'mysqli://asteriskuser:amp109@localhost/asteriskcdrdb',
```

Change the 'amp109' (in red above) to the asteriskuser password you set. CTRL+X followed by 'Y' to save and exit.

Restart mysql:

```
service mysqld restart
```

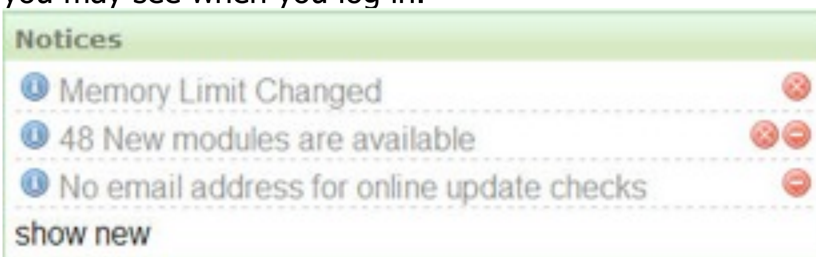
Update packages:
Update all Cent-OS packages by typing '`yum -y update`' at the command line.

6.0 - PBX Settings

The first thing we want to do is put Trixbox into Admin Mode. When you open up the main page in your browser, you will see 'User Mode [switch]' in the upper right-hand corner. Click on 'switch,' enter in your username and password (maint / maint password) and it will place you into Admin Mode.

By default, most modules needed for a small business are already installed so we don't need to install any at this time. We will cover the installation of additional modules below.

The first thing to do in PBX Settings is to clear out some of the informational messages you may see when you log in.



The 'Memory Limit Changed' message is telling us to increase the memory_limit setting in our php.ini file. To change this, go to the Linux CLI and type the following command:

```
nano -w /etc/php.ini
```

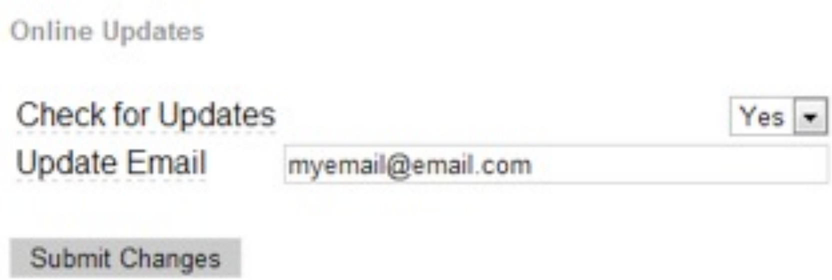
In nano, do a CTRL+W to search and type in 'memory_limit' (ENTER). Change 64M to 100M. CTRL+X followed by 'Y' to save and exit.

Once you have changed the /etc/php.ini setting, you can press the 'x' next to the informational message to clear it.

You can also click the 'x' next to 'X New modules are available' to clear that message...we won't be installing any new modules just yet.

To update modules (if necessary), click on 'Module Admin' in the left-hand menu. In Module Administration, click on 'Check for updates online' and look for any modules that need to be updated (or check the box next to 'Show only upgradable.' You can then click on individual modules and select 'Update', or click the 'Upgrade All' button. If you chose to do modules individually, you need to click 'Process' after you have selected the modules you wish to update. Then click on the orange 'Apply' bar to apply those updates.

The last message is telling us that there is 'No email address for online update checks.' Entering in your email address allows PBX Settings to notify you of any new module updates. Click on 'General Settings' in the left hand menu and scroll down to the bottom. Put your email address into the 'Update Email' field and click 'Submit Changes.' Now click on the orange 'Apply' bar and that nagging message on the PBX Settings home page goes away.



7.0 - Dial Plan

Before we start configuring, we need to come up with a dial plan. Basically, you want to map out exactly what you want your Trixbox to do prior to configuring it. This will help guide you to the appropriate configuration. Keep in mind that a Dial Plan is not something you configure in Asterisk...step away from the computer and pull out a piece of paper to design your Dial Plan.

I have 1 inbound PSTN line, and three extensions ready to be configured. I like to use 1xx extensions internally, and for inbound, I want to set up some hours of operation so that I'm not bothered by clients after 7pm. With this simple setup, my dial plan will look like this:

Inbound:

8:00am – 7:00pm – Go to main greeting (Thanks.wav). Greeting states that:

2 my extension x101 (then forwards to my cell phone)

3 my business partner's extension x104 (then forwards to his cell phone)

4 to reach first available representative (round robin between our two extensions)

7:00pm – 8:00am – Go to closed greeting (Closed.wav). Greeting states that our office is closed, and to please call back between 8am and 7pm PST.

Outbound:

All calls should be routed out the FXO port of my Digium TDM400 (my PSTN line). – I may change this later to include my IAX line if the PSTN line is busy, but let's keep it simple for now.

Internal extensions:

101 – Grandstream GXP2000 SIP phone

102 – XTen X-Lite softphone on my Windows computer

103 – Polycom IP430 SIP phone

104 – Aastra 480i SIP phone

111 – Cordless telephone that is connected to the FXS port of my Digium TDM400

That's it for my dial plan…very simple.

8.0 - Configure Internal Extensions

8.1 - Zaptel extensions

I will start by setting up my extensions. In PBX Settings, make sure you are on the 'Setup' tab and then click 'Extensions' in the 'Basic' section.

We'll start with my cordless phone (extension 111 – connected to my Digium TDM400 FXS port).

Select 'Generic ZAP device' from the drop-down box and click 'Submit.'
I used these settings:
User Extension: 111
Display name: Cordless
CID Num Alias: <blank>
SIP Alias: <blank>

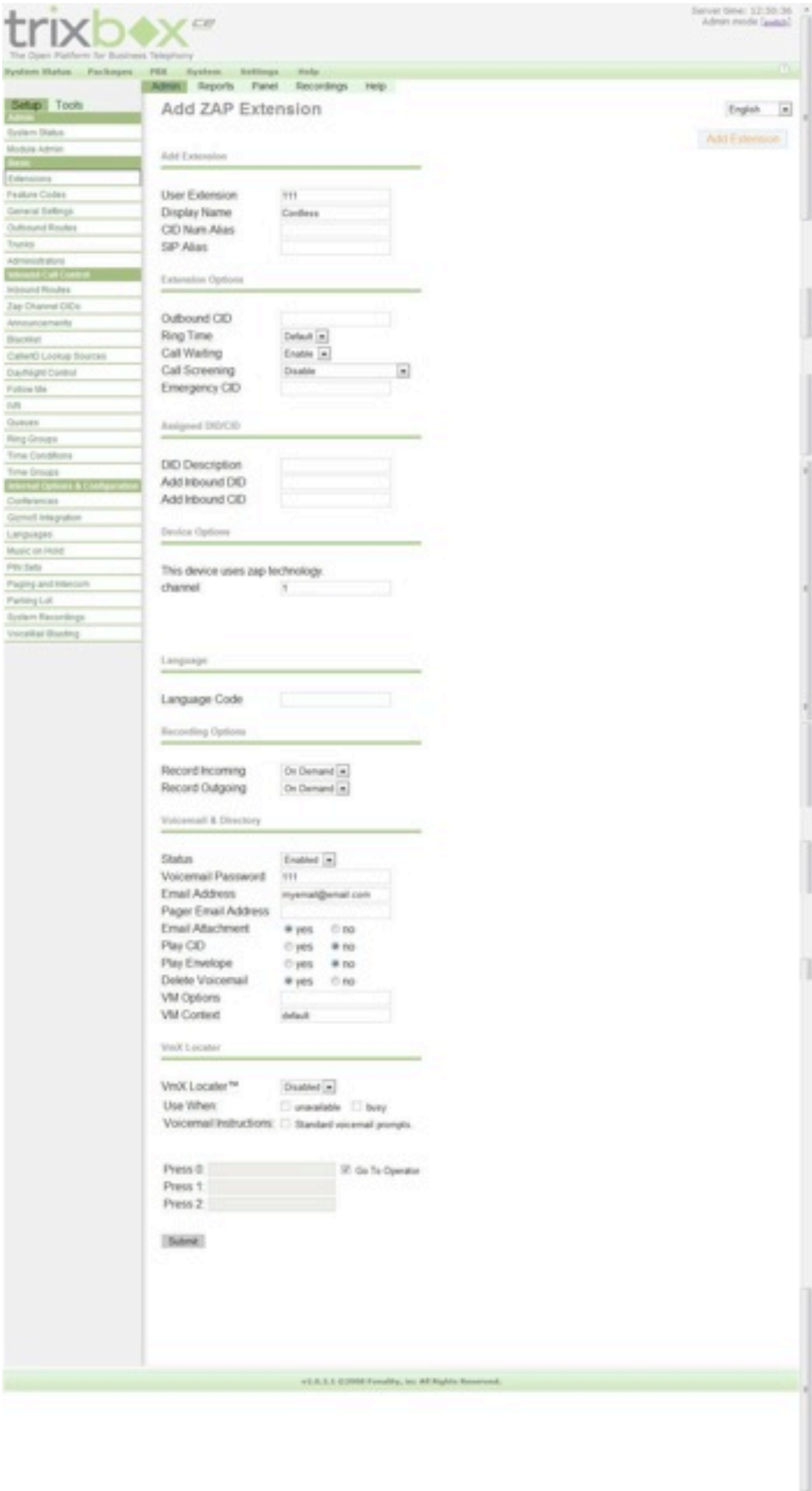
Outbound CID: <blank>
Ring Time: Default
Call Waiting: Enable
Call Screening: Disable
Emergency CID: <blank>

DID Description: <blank>
Add Inbound DID: <blank>
Add Inbound CID: <blank>

Channel: 1 (the FXS port of my Zaptel card)

Language Code: <blank>

Record Incoming: On Demand
Record Outgoing: On Demand
Status: Enabled
Voicemail password: 111 (same as extension number…keeping it simple)
Email address: (my email address)
Pager email address: <blank>
Email attachment: Yes (since I like getting the messages in my email)
Play CID: No
Play Envelope: No
Delete Voicemail: Yes (this way, the voice mails are delivered to my email inbox only…if set to no, once I delete the emails, I also have to go into the voicemail and delete the voicemail manually)
Vm options: <blank>
Vm context: default
VmX Locater: Disabled (for now)
Hit 'Submit'
Click the orange bar to apply the options.
I now get a dial tone on my cordless phone.


trixbox ce
Setup
Tools
Admin
Reports
Panel
Recordings
Help
Add ZAP Extension
English
User Extension
111
Display Name
Cordless
CID Num Alias
SIP Alias
Outbound CID
Ring Time
Default
Call Waiting
Enable
Call Screening
Disable
Emergency CID
DID Description
Add Inbound DID
Add Inbound CID
This device uses zap technology.
channel
1
Language Code
Record Incoming
On Demand
Record Outgoing
On Demand
Status
Enabled
Voicemail Password
111
Email Address
myemail@email.com
Pager Email Address
Email Attachment
yes
no
Play CID
yes
no
Play Envelope
yes
no
Delete Voicemail
yes
no
VM Options
VM Context
default
VmX Locater™
Disabled
Use When:
unavailable
busy
Voicemail Instructions:
Standard voicemail prompts.
Press 0:
Go To Operator
Press 1:
Press 2:
Submit

8.2 - Soft phone extension
Next, we'll set up a soft phone so that I can dial between internal extensions.
I use the X-Lite soft phone which can be downloaded from http://www.counterpath.com/13#Download.
In PBX Settings, click on 'Setup' and then 'Extensions' in the 'Basic' section.
Select 'Generic SIP device' from the drop-down box and click 'Submit.'
I used these settings:
User Extension: 102
Display name: Soft phone
CID Num Alias: <blank>
SIP Alias: <blank>

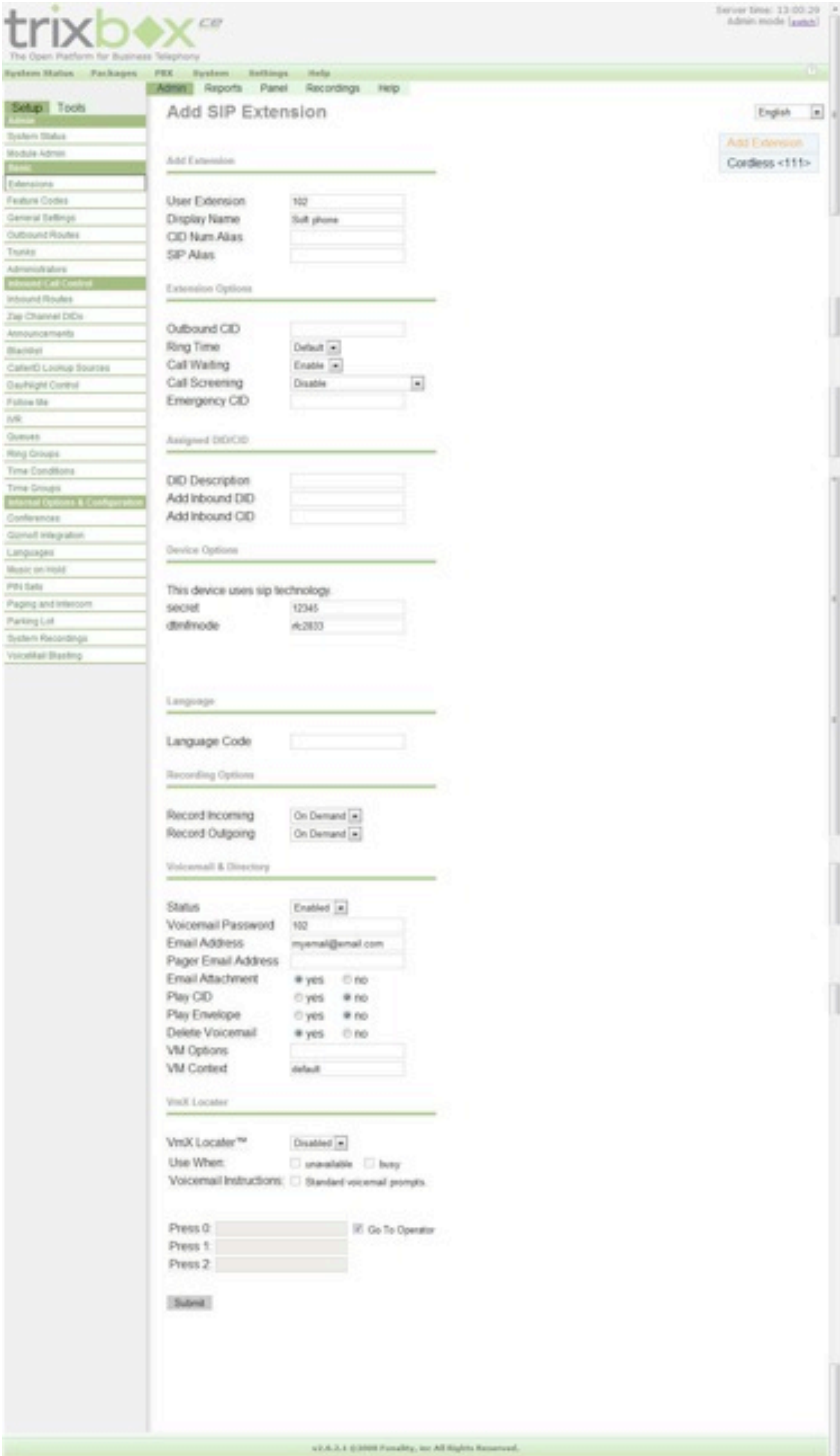
Outbound CID: <blank>
Ring Time: Default
Call Waiting: Enable
Call Screening: Disable
Emergency CID: <blank>

DID Description: <blank>
Add Inbound DID: <blank>
Add Inbound CID: <blank>

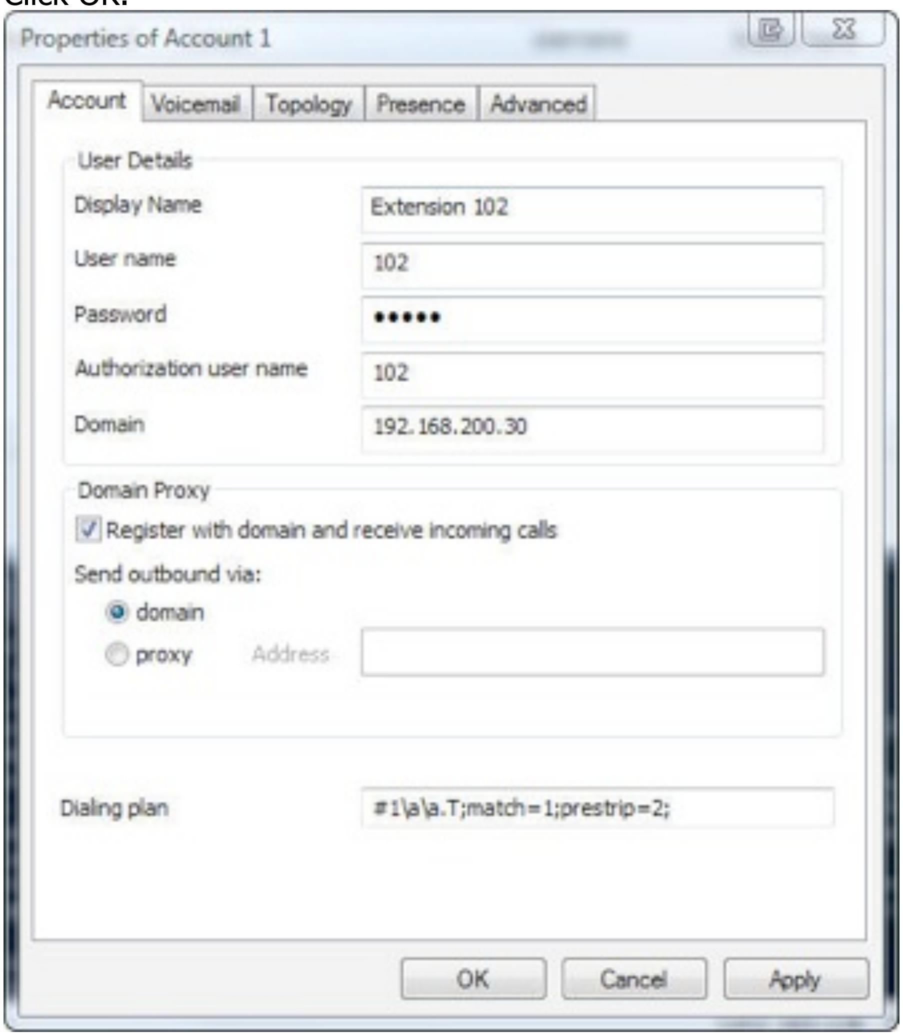
Secret: 12345 (can be whatever you want)
dtmfmode: rfc2833

Language Code: <blank>

Record Incoming: On Demand
Record Outgoing: On Demand
Status: Enabled
Voicemail password: 102 (same as extension number…keeping it simple)
Email address: (my email address)
Pager email address: <blank>
Email attachment: Yes (since I like getting the messages in my email)
Play CID: No
Play Envelope: No
Delete Voicemail: Yes (this way, the voice mails are delivered to my email inbox only…if set to no, once I delete the emails, I also have to go into the voicemail and delete the voicemail manually)
Vm options: <blank>
Vm context: default
VmX Locater: Disabled (for now)
Hit 'Submit'
Click the orange bar to apply the options.


trixbox ce
The Open Platform for Business Telephony
Admin
Reports
Panel
Recordings
Help
Setup
Tools
Add SIP Extension
English
Add Extension
Cordless <111>
User Extension
102
Display Name
Soft phone
CID Num Alias
SIP Alias
Outbound CID
Ring Time
Default
Call Waiting
Enable
Call Screening
Disable
Emergency CID
DID Description
Add Inbound DID
Add Inbound CID
This device uses sip technology.
secret
12345
dtmfmode
rfc2833
Language Code
Record Incoming
On Demand
Record Outgoing
On Demand
Status
Enabled
Voicemail Password
102
Email Address
myemail@email.com
Pager Email Address
Email Attachment
yes
no
Play CID
yes
no
Play Envelope
yes
no
Delete Voicemail
yes
no
VM Options
VM Context
default
VmX Locater™
Disabled
Use When:
Voicemail Instructions:
Press 0:
Go To Operator
Press 1:
Press 2:
Submit

Now to configure the X-Lite soft phone.
Upon first installing, the SIP configuration pops up automatically, otherwise, you can click on the down arrow at the top of the phone and choose 'SIP Account Settings.'
Click 'Add.'
Display Name: Extension 102 (can be whatever)
User Name: 102
Password: 12345 (my 'secret' from above)
Authorization user name: 102
Domain: 192.168.200.16 (your Trixbox IP address)
Domain Proxy
Register with domain and receive incoming calls (checked)
Domain selected
Click OK.

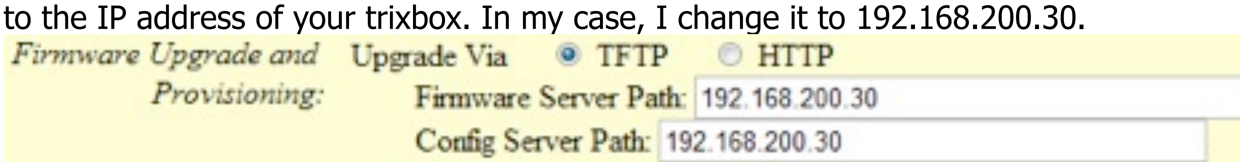


Click Close on the SIP Accounts window. Your soft phone should now register with your Trixbox.

I now try dialing my cordless phone x111 and it works! From the cordless, I dial my Soft Phone x102 and it works as well.

8.3 - Endpoint Manager / GrandStream GXP2000

I have a GrandStream GXP-2000 SIP phone on my desk and plugged into the network. It DHCP'd to IP 192.168.200.196 (I see that on the phone's display). First, I connect to the phone's web server at http://192.168.200.196 (the default password is 'admin') and click on 'Advanced Settings.' Under the 'Firmware Upgrade and Provisioning:' section, change HTTP to TFTP and then change 'Firmware Server Path' and 'Config Server Path' to the IP address of your trixbox. In my case, I change it to 192.168.200.30.



Click 'Update' and then click 'Reboot.'

I now need to create an extension for my GXP-2000. In PBX Settings, click on ‘Setup’ and then ‘Extensions.’

Select 'Generic SIP device' from the drop-down box and click 'Submit.'

I used these settings:

Extension number: 101

Display name: GXP-2000

CID Num Alias: <blank>

SIP Alias: <blank>

Outbound CID: <blank>
Ring Time: Default
Call Waiting: Enable
Call Screening: Disable
Emergency CID: <blank>

DID Description: <blank>
Add Inbound DID: <blank>
Add Inbound CID: <blank>

Secret: 12345 (can be whatever you want)
dtmfmode: rfc2833

Language Code: <blank>

Record Incoming: On Demand
Record Outgoing: On Demand
Status: Enabled
Voicemail password: 101 (same as extension number…keeping it simple)
Email address: (my email address)
Pager email address: <blank>
Email attachment: Yes (since I like getting the messages in my email)
Play CID: No
Play Envelope: No
Delete Voicemail: Yes (this way, the voice mails are delivered to my email inbox only…if set to no, once I delete the emails, I also have to go into the voicemail and delete the voicemail manually)
Vm options: <blank>
Vm context: default
VmX Locater: Disabled (for now)
Hit 'Submit'
Click the orange bar to apply the options.
Now let's set up the Endpoint Manager. Click 'PBX --> Endpoint Manager' in the trixbox GUI.

Notice the notes at bottom of the page? Since we are setting up a Grandstream GXP-2000 phone, you need to run the Grandstream Setup Tool in the Linux CLI first. Open up your SSH client to the Linux CLI and run:

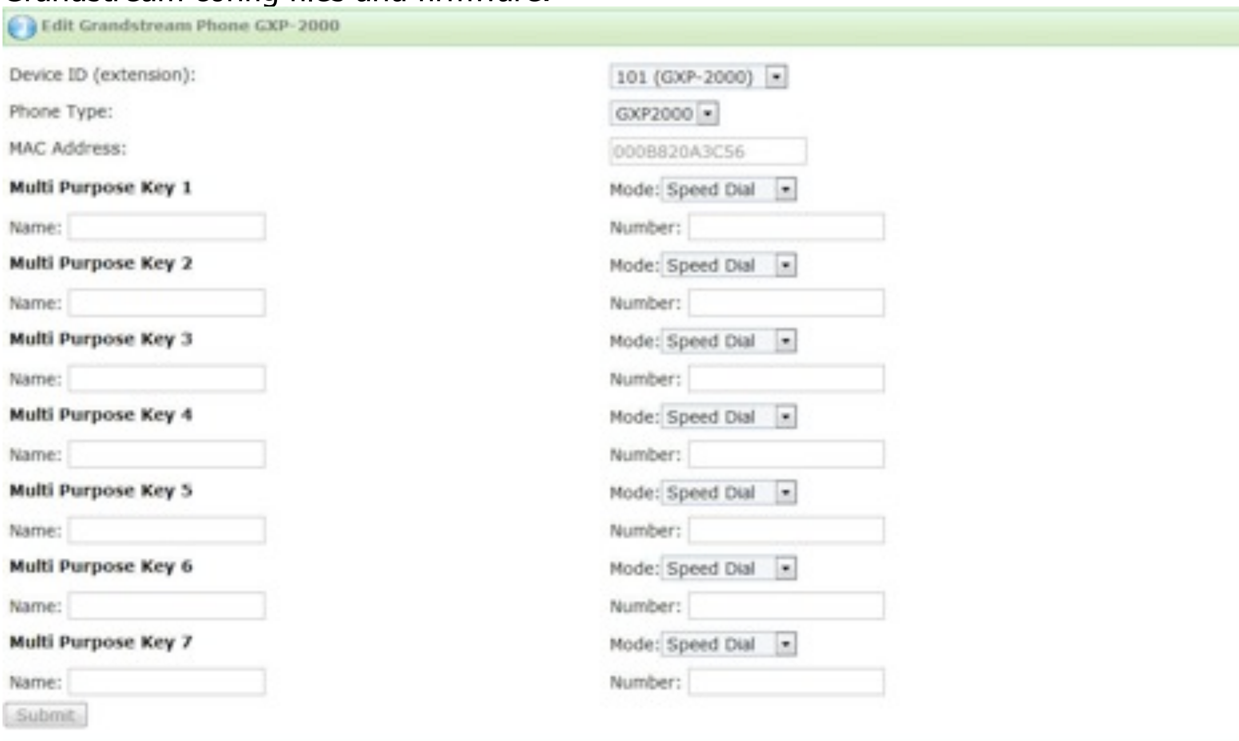
```
setup-grandstream
```

You are prompted to select which NIC you would like to use. I choose 'eth0' for my primary NIC (option 1). The program downloads and installs everything automatically.

Back in the Endpoint Manager, you should see your network already filled into the 'Map Devices' section. Click 'Go.' It should come up with a list of your phones. Click on the desired phone's MAC address and you are taken to the configuration screen.

For the Device ID, I select the GrandStream SIP extension I created above (101) and hit 'Submit.' My GrandStream phone is now configured (after a few auto-reboots).

***NOTE:** If you get an error stating that it could not create the config file, you need to run 'setup-grandstream' from the Linux CLI. This will download and install the latest Grandstream config files and firmware.



Now, each time you reboot your Grandstream phone, it will download its configuration files from your trixbox CE's TFTP server.

***** TIP:** To view the status of your TFTP server in real time to verify whether phones are connecting and successfull downloading files, run 'tail -f /var/log/atftpd.log' from the Linux CLI. CTRL+C to exit viewing the TFTP log file in real time.

For the GrandStream GXP-2000's however, there is one more bit of picky configuration that is optional. Once configured by the endpoint manager, I realized that there were a couple of annoying problems. First, the LCD backlight was always on, and did not turn off after a minute or so. Second, the time was not set correctly and appeared to be set to Eastern time (whereas I am on the West Coast). To fix this, I had to edit the GrandStream default settings file that resides on the TFTP server of my v2.6.2 trixbox.

```
cd /tftpboot
nano Grandstream_GXP2000_Default.txt
```

Towards the bottom, there is a section called 'End User Time Settings.' I changed the following:

```
# Time Zone. Offset in minutes to GMT
P64 = 240
```

(Not sure exactly how this works, even though my time zone is PST (or -8 GMT), 240 minutes set here made it accurate).

```
# Daylight Savings Time. 0 - no, 1 - yes
P75 = 1
```

(For the new DST rules)

```
# Optional rule. Daylight Saving rule.
P246 = 3,2,7,2,0;11,1,7,2,0;60
# LCD Backlight Always On. 0 - no, 1 - yes
P322 = 0

# Date Display Format. 0 - Year-Month-Day, 1 - Month-Day-Year, 2
- Day-Month-Year
P102 = 1

# Display Clock instead of Date. 0 - no, 1 - yes
P123 = 1
```

(That last two were simply personal preference...it displays the time large and the date small instead of vice versa and the date in the format MM-DD-YYYY).

Once changes have been made, press CTRL+X followed by Y to save and exit. Then, go back into the Endpoint Manager and click on 'Grandstream' followed by your Grandstream phone's MAC address and click 'Submit' at the bottom to re-generate the phone's config using the new default settings. Finally...reboot your phone one more time.

8.4 - Polycom IP430

The next phone I am going to configure is a Polycom IP430. I booted up the phone, and it DHCP'd an IP address of 192.168.200.195. The first thing we need to do is set the phone's TFTP server to the IP address of the trixbox. This has to be done through the phone's physical interface. Press the 'Menu' button and scroll down to 'Settings' and press the checkmark button. Then scroll down to 'Advanced' and press the checkmark button. You are now prompted to enter the Admin password which is '456' by default. Enter in the password and press the 'Enter' softkey.

Now select 'Admin Settings,' followed by 'Network Config.' Scroll down to 'Server Menu' and press the checkmark button.

For 'Server Type,' press the checkmark button and then press the right-arrow button until the screen reads 'Trivial FTP.' Now press the checkmark button again.

Now press the down-arrow button once to get to 'ServerAddr' and press the checkmark button again. You are now editing the TFTP IP address. Press the '1/A/a' softkey until you see (1/Ascii) at the top of the screen (meaning that we are going to be entering in numbers rather than letters). Punch in the IP address of your trixbox server and press the checkmark button once again. Use the '*' key for a period (.). Press the 'OK' softkey when finished followed by 'Exit,' 'Exit,' 'Select.' The phone will now reboot.

Next, we need to configure our extension. Per our Dial Plan, this phone will be extension 103. Go into 'PBX Settings' and click on 'Extensions.'

Select 'Generic SIP device' from the drop-down box and click 'Submit.'
I used these settings:
Extension number: 103
Display name: Polycom IP430

CID Num Alias: <blank>
SIP Alias: <blank>

Outbound CID: <blank>
Ring Time: Default
Call Waiting: Enable
Call Screening: Disable
Emergency CID: <blank>

DID Description: <blank>
Add Inbound DID: <blank>
Add Inbound CID: <blank>

Secret: 12345 (can be whatever you want)
dtmfmode: rfc2833

Language Code: <blank>

Record Incoming: On Demand
Record Outgoing: On Demand
Status: Enabled
Voicemail password: 103 (same as extension number…keeping it simple)
Email address: (my email address)
Pager email address: <blank>
Email attachment: Yes (since I like getting the messages in my email)
Play CID: No
Play Envelope: No
Delete Voicemail: Yes (this way, the voice mails are delivered to my email inbox only…if set to no, once I delete the emails, I also have to go into the voicemail and delete the voicemail manually)
Vm options: <blank>
Vm context: default
VmX Locater: Disabled (for now)
Hit 'Submit'
Click the orange bar to apply the options.

Before we go any further, we need to run '`setup-polycom`' in the Linux CLI to install the necessary Polycom files. Select eth0 as your network interface.

Now, go back to the Endpoint Manager and run 'Map Devices' again. You should see the Polycom's IP and MAC address show up in the list of available devices. Click on the MAC address and enter in the extension we created (x103) and select the phone type (IP 430). Click 'Submit.'

Reboot the phone, and it should now be configured at x103.

***NOTE: When I first configured this phone, it worked, but it kept rebooting over and over. It needed the latest Polycom firmware. I was able to install the latest Polycom firmware by running '`yum install firmware-polycom`' from the Linux CLI.

8.5 - Aastra 480i

Let's configure an Aastra 480i next. My Aastra 480i has DHCP'd an IP address of 192.168.200.194. Browse to the phone's IP address in your browser and enter in the default username and password of admin / 22222. Once in the phone's GUI interface, select 'Configuration Server.'

Set 'Download Protocol' to 'TFTP' and set the 'TFTP Server' setting to the IP address of the trixbox.

Click 'Save Settings' at the bottom and then reset the phone.

Now we need to create x104 for this phone.

Go into 'PBX Settings' and click on 'Extensions.'

Select 'Generic SIP device' from the drop-down box and click 'Submit.'
I used these settings:
Extension number: 104
Display name: Aastra 480i
CID Num Alias: <blank>
SIP Alias: <blank>

Outbound CID: <blank>
Ring Time: Default
Call Waiting: Enable
Call Screening: Disable
Emergency CID: <blank>

DID Description: <blank>
Add Inbound DID: <blank>
Add Inbound CID: <blank>

Secret: 12345 (can be whatever you want)
dtmfmode: rfc2833

Language Code: <blank>

Record Incoming: On Demand
Record Outgoing: On Demand
Status: Enabled
Voicemail password: 104 (same as extension number...keeping it simple)
Email address: (my email address)
Pager email address: <blank>
Email attachment: Yes (since I like getting the messages in my email)
Play CID: No

Play Envelope: No
Delete Voicemail: Yes (this way, the voice mails are delivered to my email inbox only…if set to no, once I delete the emails, I also have to go into the voicemail and delete the voicemail manually)
Vm options: <blank>
Vm context: default
VmX Locater: Disabled (for now)
Hit 'Submit'
Click the orange bar to apply the options.

Next, run '`setup-aastra`' from the Linux CLI and select eth0 as the network interface.

Now, go back to the Endpoint Manager and run 'Map Devices' again. You should see the Aastra's IP and MAC address show up in the list of available devices. Click on the MAC address and select the extension we created (x104) for the 'Device ID,' and then select the phone type (480i). Click 'Submit.'

Reboot the phone, and it should now be configured at x104.

8.6 - Voicemail options

One thing that I have found with the default voicemail configuration is that the sound files delivered to my email address have a very low volume level. If you are receiving voice mails via email to a cell phone, you can't even hear it. We can change the voicemail to email attachment format. There are two flavors of attachment...WAV and WAV49. It defaults to WAV49 to save file size, but let's change it to WAV. From the Linux CLI, do the following:

```
cd /etc/asterisk
nano vm_general.inc
```

Change

```
format=wav49|wav
```

to

```
format=wav|wav49
```

CTRL+X to exit and then Y when asked to save.

8.7 - Polycom central server configuration

Since the Polycom SoundPoint IP phone central server configuration is pretty involved, I have included it in a separate document. You can find detailed information for configuring the Polycom SountPoint IP phone central server at http://www.sureteq.com/asterisk/polycom.htm.

***Added note: My Polycom document is probably outdated for this version of trixbox CE. I'm sure you can get the gist of it though.

8.8 - A note on hardware phones

I get asked a lot what a 'good' phone is, so I thought I'd take a moment to give you some impressions on some of the more common hardware phones available.

GrandStream GXP-2000 - This is a very good entry level phone that retails for about $85 bucks. It has all of the basic functionality you would expect from an office phone including speaker phone, 4 line appearances, and buttons for transfer, conference, voice mail, and mute. It is pretty rugged and I have had nothing but good experiences with this phone. It is also very easy to configure via the endpoint manager.

Aastra phones - Aastra makes a really solid product in their entire IP phone line. They have a full range of phones from the low end (9143's - about $123.00 USD and the 51i - about $76.00 USD) to the high end (57i - about $205.00 USD). They are (in my opinion) the easiest phones to install and administer, and these are what I typically recommend to customers. The 57i can use the 536M (analog) or 560M (digital) side car expansion modules for receptionists. The 480i CT and 57i CT come with a cordless handset in addition to the phone base for users who need portability.

Polycom SoundPoint IP phones - The IP430's and IP501's are very good mid-level phones that retail for between $170 and $190 bucks each. The speaker phone is full duplex and is arguably the best quality speaker phone available. You can have from 2 to 6 line appearances depending on which phone model you choose. The IP601 has a very nice side-car available for receptionists which is fully programmable with line appearances. Plus, they are very easy to configure once you have set up their central server.

Cisco phones - The Cisco phones look great, but they are difficult to configure and are super expensive. I never recommend them.

9.0 - Outbound Routes

9.1 - Outbound route with default Zap trunk

Now that we have set up our internal extensions, let's focus on getting calls out. To do this, in PBX Settings, click on 'Setup' --> 'Outbound Routes.'

By default, Trixbox has already created a trunk out of my FXO port in the Digium TDM400 card (trunk Zap/g0), and has already created a route which makes the user dial 9 to get an outside line (0 9_outside). Lets click on the '0 9_outside' route on the right hand side of the screen and add a few more dial patterns for outbound dialing. The only dial pattern so far should be '9|.' You can leave this dial pattern if you want your users to dial 9 to get an outside line.

If you do NOT want to dial 9, or if you want to be more specific about your outbound routes, remove it and insert your own patterns. Under Dial Patterns, you should see

'Dial patterns wizards.' This allows you to insert pre-defined dial patterns for commonly dialed numbers. I picked the following patterns:
Local 7/10 digit
Long Distance
Toll Free
Information
Emergency
Click 'Submit Changes'
Click the orange bar at the top of the screen to apply the changes.
My Dial Pattern for route 9_outside now looks like this:

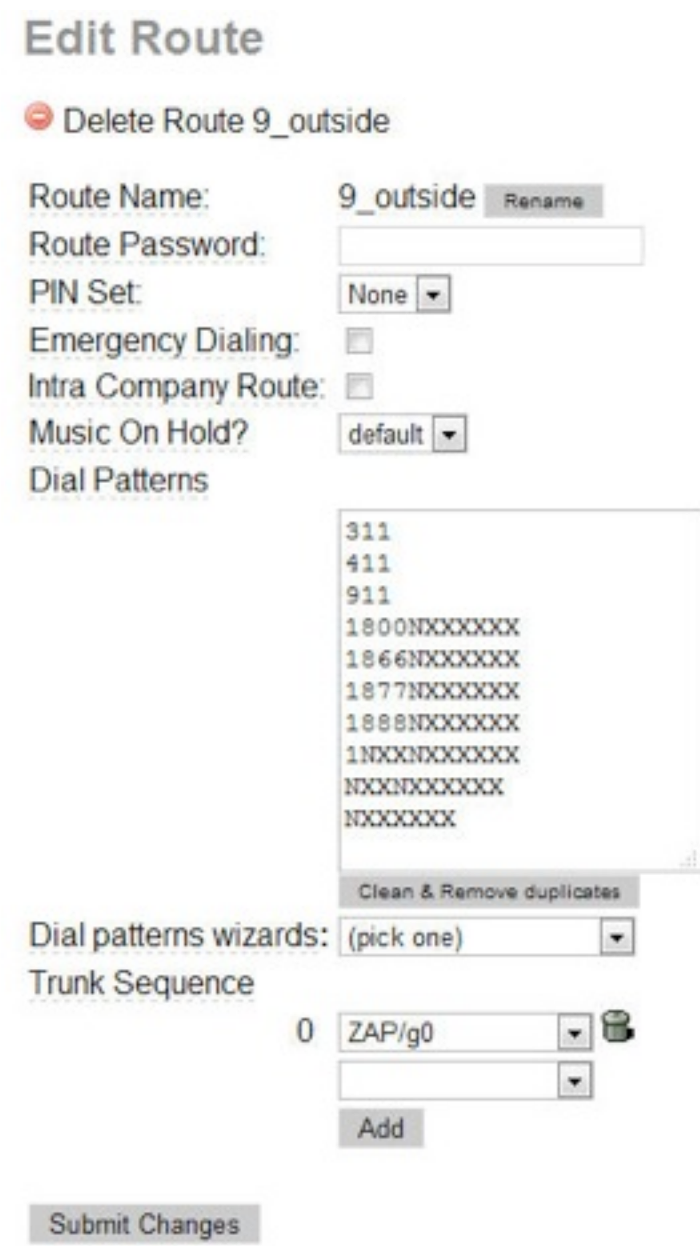
```
311
411
911
1800NXXXXXX
1866NXXXXXX
1877NXXXXXX
1888NXXXXXX
1NXXNXXXXXX
NXXNXXXXXX
NXXXXXX
```

I try dialing my cell phone with my cordless x111, and it works!

If you want to keep 9 as the number to dial to get an outside line, you can append it to the front of these dial strings with a pipe character. The pipe character tells Asterisk to strip off the 9 when sending the call to the outside world. This works for whatever prefix you want (ie. 8, 7, etc.). If I wanted to be specific about my dial plan (ie. not using 9|.), for instance, if I don't want anyone dialing Internationally (such as above), but I still want to use 9 as my prefix, I would change my dial pattern to look something like this:

```
9|311
9|411
9|911
9|1800NXXXXXX
9|1866NXXXXXX
9|1877NXXXXXX
9|1888NXXXXXX
9|1NXXNXXXXXX
9|NXXNXXXXXX
9|NXXXXXX
```

*****TIP:** Regarding emergency dialing (911), you may want to include BOTH 911 and 9|911 in your dial pattern so that in the event of an emergency, either pattern will work.



9.2 - VoIP Provider Setup

Since I only have a single Vonage line, if I want to be on more than one call at a time I'll have a problem. To fix this, I'll need to set up a VoIP provider to allow me to make calls through my broadband connection. trixbox has removed the VoIP Setup Wizard, but here is how to manually set up 2 providers previously included in the trixbox package.

*****NOTE:** Please note that I do not endorse either of the VoIP providers listed below...they are simply examples of two VoIP providers I know to work with trixbox CE. It is up to the individual installer to do the proper due diligence when selecting a VoIP provider.

9.3 - VoicePulse

To set up a VoicePulse trunk, first you need to create an account with VoicePulse by going to http://www.voicepulse.com and signing up for an account. Once your account is created, you will be able to log into the VoicePulse site and retrieve your login and password.

You will want to make note of your login and password as it will be used to set up your trunks. You can find your login and password by logging into the VoicePulse site and clicking 'Credentials' in the top bar.

First, we need to download the VoicePulse public encryption key. Go to your Linux CLI and do the following:

```
cd /var/lib/asterisk/keys
wget http://connect.voicepulse.com/keys/voicepulse20060419.pub
```

Now, go back to PBX Settings and click on Trunks --> Add IAX2 Trunk. Enter in the following information:

Outgoing CallerID: "Your Name" <Your number>
Never Override CallerID: unchecked
Maximum Channels: 4
Disable Trunk: unchecked
Monitor Trunk Failures: unchecked

Outgoing Dial Rules
Dial Rules: (Enter in your own dial rules)

`1NXXNXXXXXX`
`1818+NXXXXXX` (appends my local area code to 7-digit dialed numbers)

Or, if dialing 9, your Dial Rules may look like this:

`9|1NXXNXXXXXX`
`9|1818+NXXXXXX` (appends my local area code to 7-digit dialed numbers)

Outbound Dial Prefix: Leave blank

Outgoing Settings:
Trunk name: Voicepulse1
Peer Details: (put this in the text box):

```
allow=ulaw&alaw&gsm&ilbc&g726&adpcm
disallow=all
host=connect01.voicepulse.com
notransfer=yes
qualify=yes
secret=(your VoicePulse password)
type=peer
username=(your VoicePulse username)
```

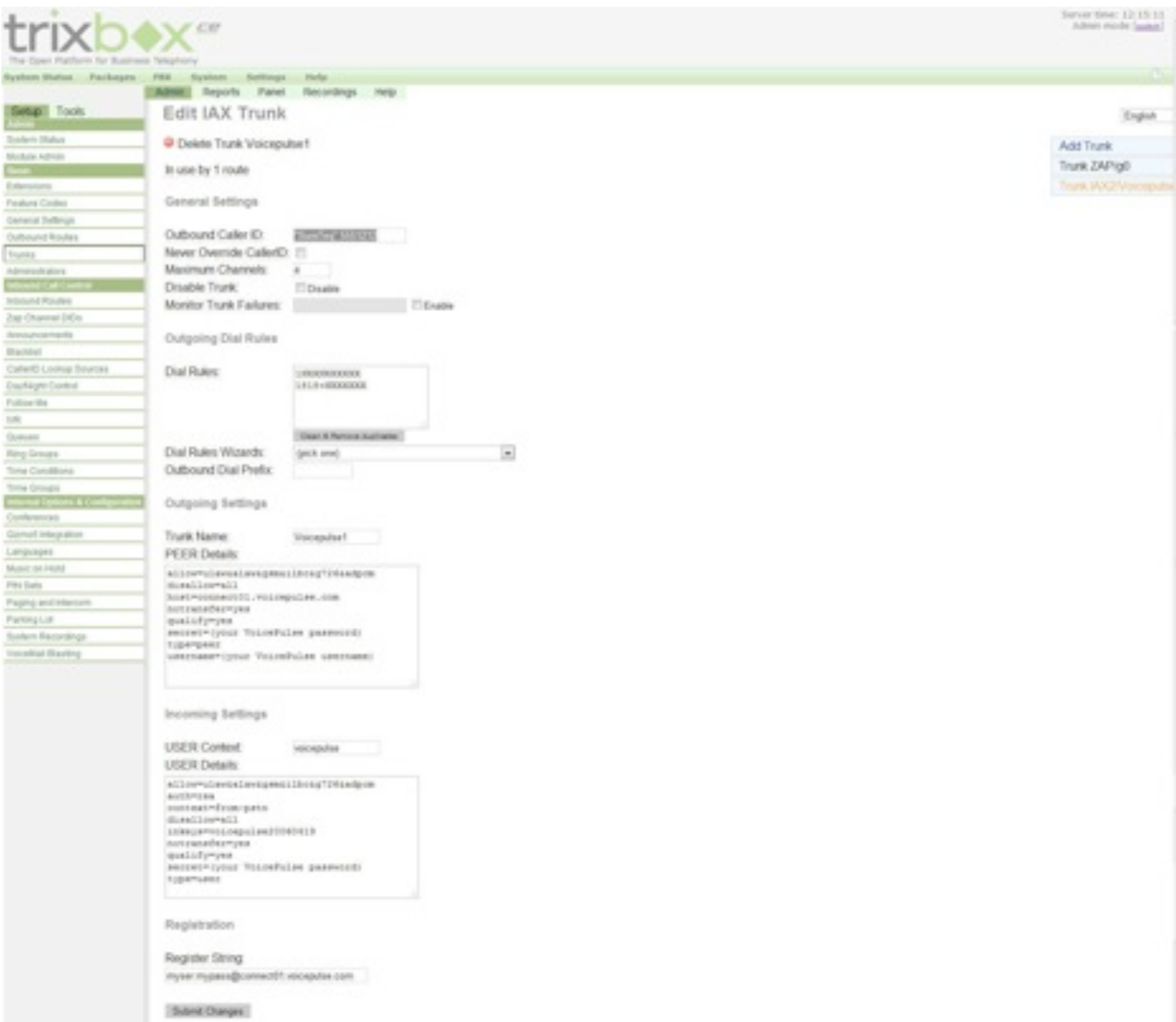
Incoming settings:
User context: voicepulse (this HAS to be voicepulse)
User details: (put this in the text box):

```
allow=ulaw&alaw&gsm&ilbc&g726&adpcm
auth=rsa
context=from-pstn
disallow=all
inkeys=voicepulse20060419
notransfer=yes
qualify=yes
secret=(your VoicePulse password)
type=user
```

Register String: (VoicePulse Login):(VoicePulse password)@connect01.voicepulse.com

So, for instance, if your Login and password are F8yyZ41fGG and J974YmmZ3d then your register string would be "F8yyZ41fGG:J974YmmZ3d@connect01.voicepulse.com".

Click Submit and then the orange bar to apply changes.



Repeat all of the steps above for your 2nd IAX2 trunk, but replace **Voicepulse1** with **Voicepulse2** and replace **connect01.voicepulse.com** with **connect02.voicepulse.com** everywhere you find them.

To verify whether or not these settings worked, go to your Asterisk CLI and type 'iax2 show registry' to display your IAX registry settings:

```
asterisk1*CLI> iax2 show registry
Host Username Perceived Refresh State
64.61.93.90:4569 (username) (your IP address):58403 60
Registered
64.61.93.87:4569 (username) (your IP address):58404 60
Registered
```

Next, we need to create an outbound route that uses these trunks. Click on 'Outbound Routes' and enter in the following info:

Route Name: VoicePulseOut
Route Password: <blank>
PIN Set: <no selection>
Emergency Dialing: unchecked
Intra Company Route: unchecked
Music On Hold: default

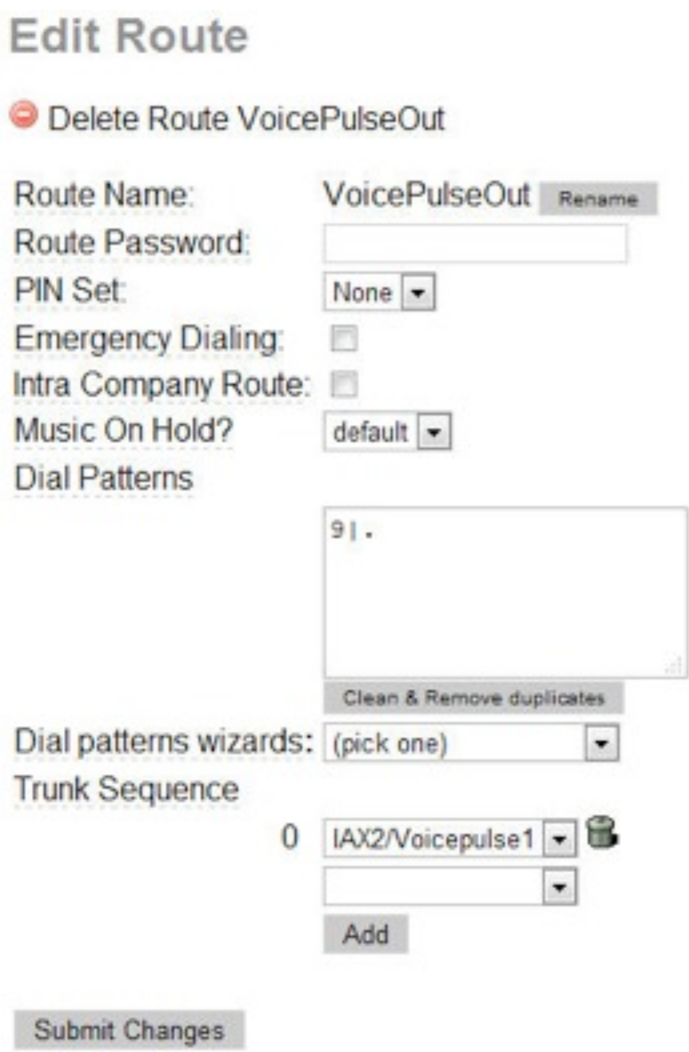
Dial Patterns:

`9|.` (or enter in desired dial pattern described above in chapter 9.1)

Trunk Sequence:

0: IAX2/Voicepulse1
1: IAX2/Voicepulse2

Click submit and then the orange bar to apply changes.



I set the VoicePulse trunks as my only outbound route and try dialing my cell phone in 10 digit format (91NXXNXXXXXX), and it works! 7 digit dialing also works because I added the '1818+NXXXXXX' to my Voicepulse trunk dialing pattern.

9.4 - VoIP Street

To set up VoIP Street, first, you need to sign up for an account. To do so, go to http://www.voipstreet.com and sign up for an account. Enter in your information and wait for your email response. Once you have your email, log in to the Voip Street customer portal and click on 'Menu' on the left-hand side of the screen. It will expand out. Click on the '+' sign next to 'Account Settings' and then click on 'Account Home.'

In the 'Quick Launch' menu, click on 'View/Add Devices.'

In the 'Add a New Device' section, enter in the following info:

Friendly name: (can be whatever...I called mine Outbound)
Protocol: IAX2 Protocol
Codec: ulaw (g711)

Click the 'Add New Device' button, then click 'OK' on the confirmation screen.

You should now see your new device in the 'Devices Currently Configured' section. Click on the 'Setup Help' link under 'Config' in your device. Note the information listed in the trixbox section and then go back to your trixbox VoIP Street provider page and fill in the following info:

Desired Caller ID Name: (Can be whatever you want)
Desired Caller ID Number: (Can be whatever you want)
Device Username: (Enter in the Device Username given to you by VoIP Street)
Device Password: (Enter in the Device Password given to you by VoIP Street)
Device Protocol: IAX
Device Codec: g711

Click 'Continue.'

Click the 'Create Trunk Now' button.

When the PBX Setting screen shows up, click the orange bar at the top of the screen. You should now have a new trunk called 'IAX2/voipstreet' listed in your list of trunks. Lets try to make a call with that trunk.

Click on 'Outbound Routes' and then the '9_outside' route.

Under 'Trunk Sequence' make IAX2/voipstreet the main trunk. (To be extra sure this is working, delete all other trunks listed...but remember to add them back after testing).

I try to dial my cell phone using 10 digit dialing (91NXXNXXXXXX) and it works...but when I try 7 digit dialing (9NXXXXXX), it says 'All Circuits are Busy.' So let's go back to the VoIP Street trunk and adjust the dial plan.

Click on 'Trunks' and then 'Trunk IAX2/voipstreet.' In the 'Dial Rules' section, add the following:

1NXXNXXXXXX
1818+NXXXXXX

This means to dial 10 digit calls normally, but to add my local area code (818) to 7 digit dialed numbers.

Or, if dialing 9, your Dial Rules may look like this:

`9|1NXXNXXXXXX`
`9|1818+NXXXXXX` (appends my local area code to 7-digit dialed numbers)

I try my 7 digit dialed call again, and this time it works perfectly!

10.0 - Incoming calls/IVR setup

Now, I need to tell trixbox what to do with incoming calls. When designing an IVR, it is always best to design forward, and program backward. What does that mean? Well...for instance, if you want trixbox CE to answer a call and play a recording...you need to have the recording first! So by programming backwards, you'll start with extensions, system recordings, queues, and then work your way to the front of the IVR where you will be referencing these endpoints.

10.1 - System recordings

First, let’s take care of the necessary recordings for my IVR.
I want two different messages…one for during business hours, and one for when we are closed. The scripts will look something like this:
Open – “Thank you for calling Sherwood’s Asterisk emporium. Please press 2 for Sherwood, press 3 for Jim, and press 4 for the next available associate. Thank you and have a wonderful day.”
Closed – “Thank you for calling Sherwood’s Asterisk emporium. Our office is now closed. Please call back between the hours of 8am and 7pm Pacific Standard time. Thank you.” (click).
To get these recorded first clear your throat and put on your best announcer voice.
Click on System Recordings (under Internal Options & Configuration).
You can either record your greetings as .WAV files in an external application (8-bit mono recordings work best over the phone lines), or straight into an extension. We’ll record using our extension.
Put your extension number into the extension field and click ‘Go.’
Pick up the extension that you put into the extension field and dial *77. You will hear a beep…start your recording. When you are done, hang up the extension. Type in a name for your recording in the 'Name this Recording' field (without the .wav extension) and click ‘Save.’
Repeat for as many recordings as you would like to have in trixbox CE.

***TIP: trixbox CE allows you to string together multiple recordings into a single recording. To do this, Add Recording (as shown above), and then click on your recording. You should see 'Files:' down at the bottom with 'custom/yourrecording' as the first file. To add additional files, select them from the drop down box. Any files you have recorded will show up in this box as 'custom/filename.'

10.2 - IVR

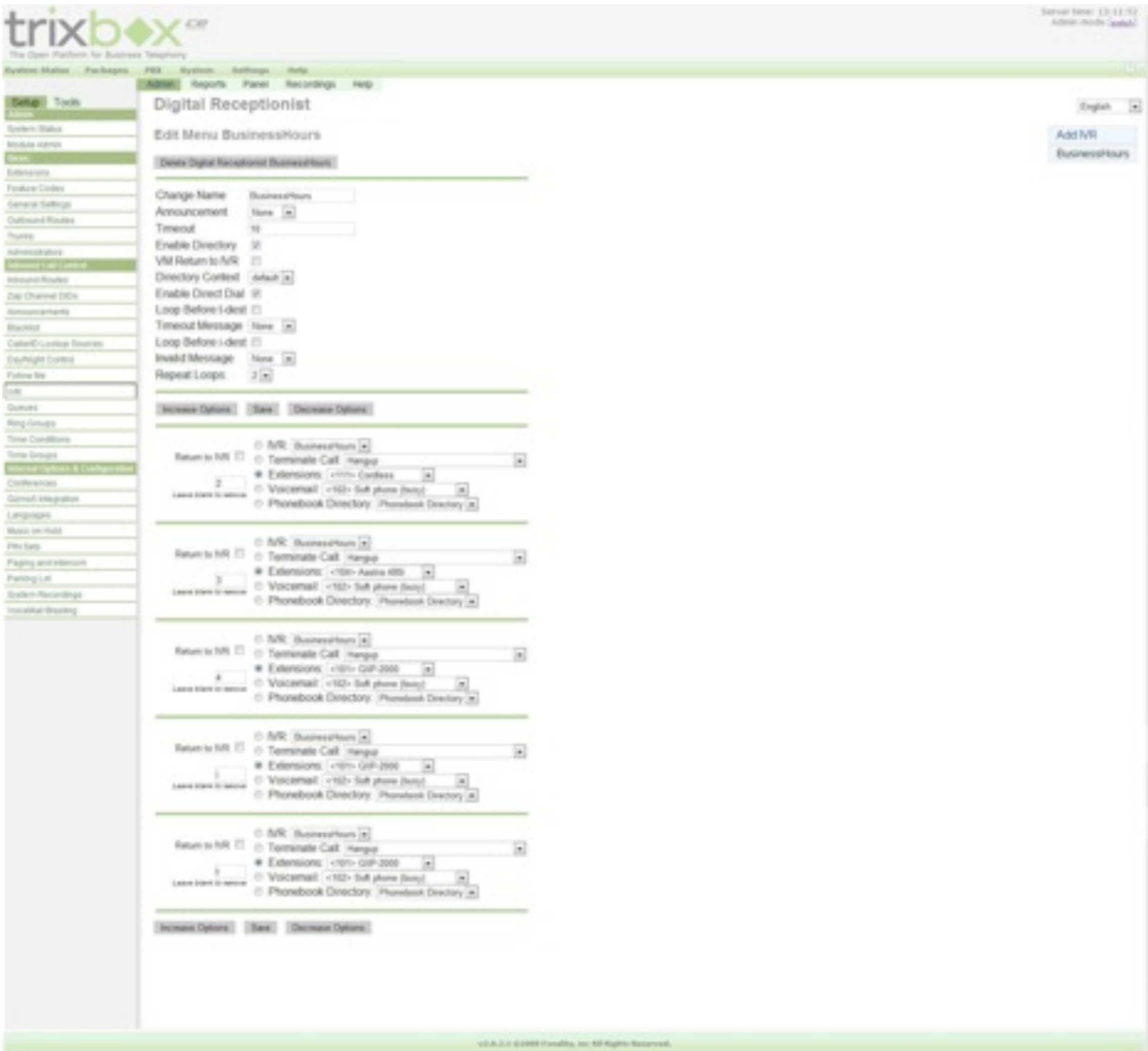
Click on IVR and then click Add IVR.
I used these settings for my initial IVR:
Change Name: BusinessHours
Announcement: Open (my pre-recorded 'Open' recording from above)
Timeout: 10
Enable Directory: checked
VM Return to IVR: unchecked
Directory Context: default
Enable Direct Dial: checked (this allows callers to dial extensions directly from the IVR)
Loop before t-dest: unchecked (this repeats the announcement before going to the 't' (timeout) destination if specified in the options below)
Timeout message: none (recording that plays to the caller instead of the 'Announcement' when looping back to the beginning if 'Loop before t-dest' is checked)
Loop before i-dest: unchecked (same as 'Loop before t-dest,' except this one is for if a caller enters in an invalid extension)
Invalid message: none (same as Timeout Message, but for invalid extension)
Repeat loops: 2 (default) (this sets how many times the IVR will loop for the 4 options above)

For my options, I will have 3, so I leave the Increase/Decrease default, but if you have more or less options, feel free to add/remove them.
Options
2 – Extensions – Cordless <111>
3 – Extensions – Aastra 480i <104>
4 – Extensions – GXP-2000 <101>

I'm also going to assume that the GXP-2000 (x101) is my receptionist for our purposes. I will also set the following IVR options (t for timeout and i for invalid extension):

t - Extensions - GXP-2000 <101>
i - Extensions - GXP-2000 <101>

Click 'Save'
Click on the orange bar at the top of the screen.
*** **NOTE:** I know I said that I was going to have option 4 be round robin between two extensions…I'm getting there, but I just want to get this set up and working with local extensions first.
*** **NOTE:** Why didn't I have option 1 in my IVR (I started with option 2)? Because my internal extensions are dial pattern 1xx. If I want to enable direct-dial, I can't have a number in my IVR that matches the start of my extension dial plan or else there will be a long delay while the system waits to see if I am going to input any more digits.)



Now to add my After Hours IVR. Click on ‘Add IVR’
Change Name: AfterHours
Announcement: Closed (my pre-recorded Closed recording from above)
Timeout: 10
Enable Directory: checked
VM Return to IVR: unchecked
Directory Context: default
Enable Direct Dial: checked
Loop Before t-dest: unchecked
Timeout Message: None
Loop Before i-dest: unchecked
Invalid Message: None
Repeat Loops: 2 (default)
Options:
2 – Extensions – Cordless <111>

3 – Extensions – Aastra 480i <104>
4 – Extensions – Grandstream GXP-2000 <101>
***** Note**: The options are exactly the same as my BusinessHours IVR except for the name and the announcement…why the heck did I do that? Well…because my Thanks recording lists the possible extensions (2,3,4), and my Closed recording does not… however I still want specific people to be able to call my extension after hours, so even though the after hours recording does not list the options…the options are still available...just not mentioned in my recording.

10.3 - Time Groups

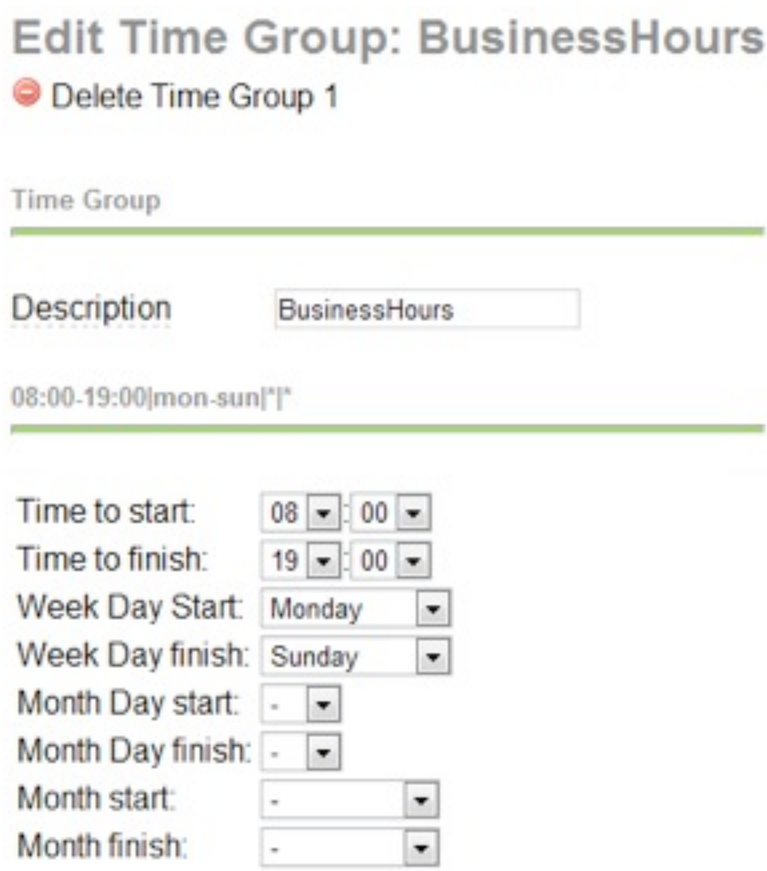
Setting up time groups and time conditions will be the next step. Click on Time Groups. I used these settings:

Description: BusinessHours

New Time
Time to start: 08:00
Time to finish: 19:00
Week Day Start: Monday
Week Day Finish: Sunday (we're open 7 days!)
Month Day start: blank
Month Day finish: blank
Month start: blank
Month finish: blank

Click 'Submit Changes'
Click the orange bar at the top of the screen to apply the settings.

***TIP: Having 'Month Day start/finish' and 'Month start/finish' allows you to set up schedules for specific days of the year such as holidays, and then play your default 'Closed' message or a custom 'Closed' message for that day. For instance, if you wanted to set up New Years Day, you would set 'Month Day start and finish' to 1, and then 'Month start and finish' to January.



10.4 - Time Conditions

Now that we have our Time Group set up, we can now create a Time Condition that uses that group. The Time Condition tells the system where to route calls based on the Time Group we created. Click on Setup --> Time Conditions to add a new Time Condition and use these settings:

Time Condition Name: Main (can be whatever)
Time Group: BusinessHours (the group we created in chapter 10.3 above)

Associate with: No Association (for now...this settings allows you to configure a feature code which will manually override the Time Group selected)

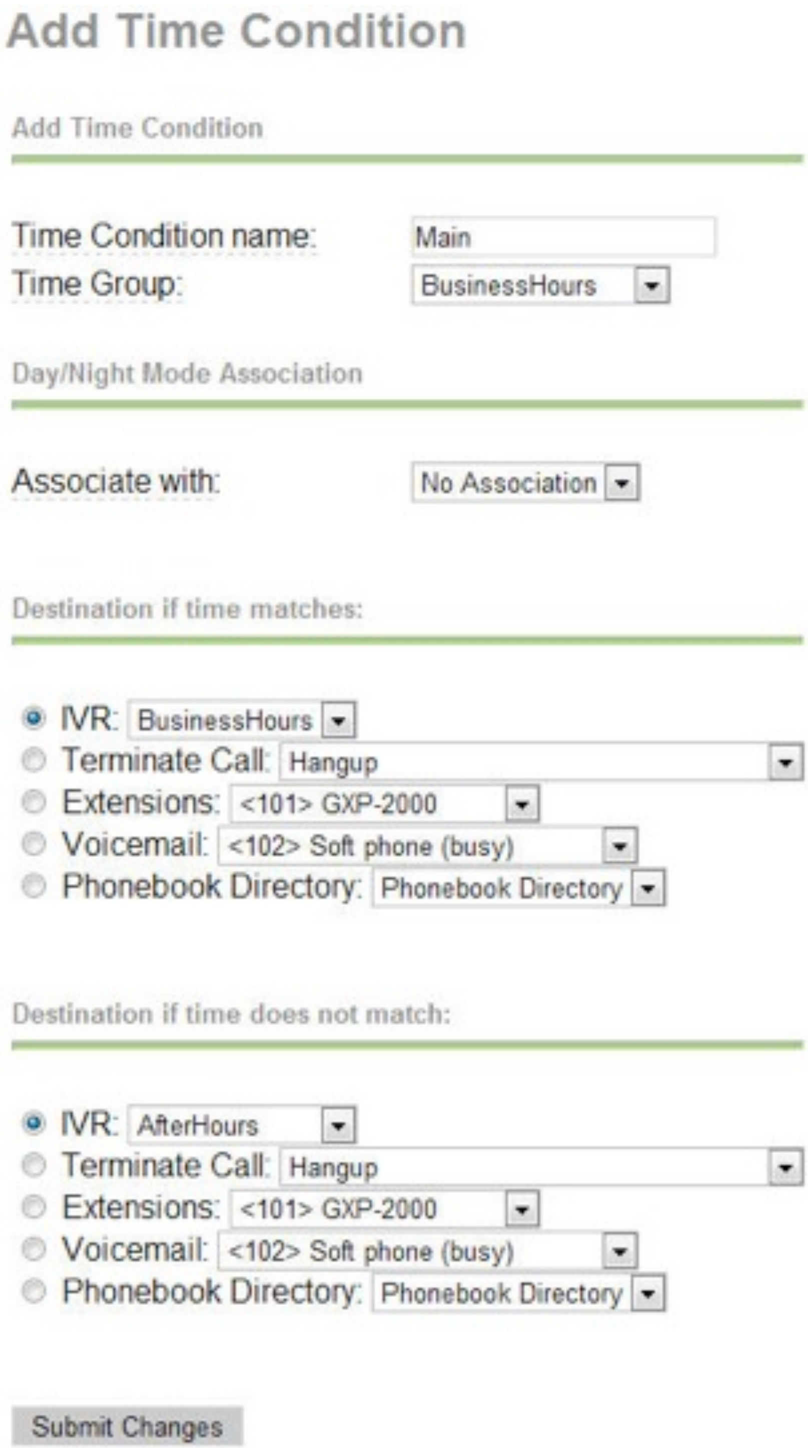
Destination if time matches:

IVR: BusinessHours

Destination if time does not match:

IVR: AfterHours

Click 'Submit Changes' followed by the orange bar.



10.5 - Inbound Route

Next, we want to add our incoming route. Click on 'Inbound Routes' under Inbound Call Control. You can route based on DID number or Caller ID Number, or if both of those are left blank, it will route all un-matched calls to your settings.

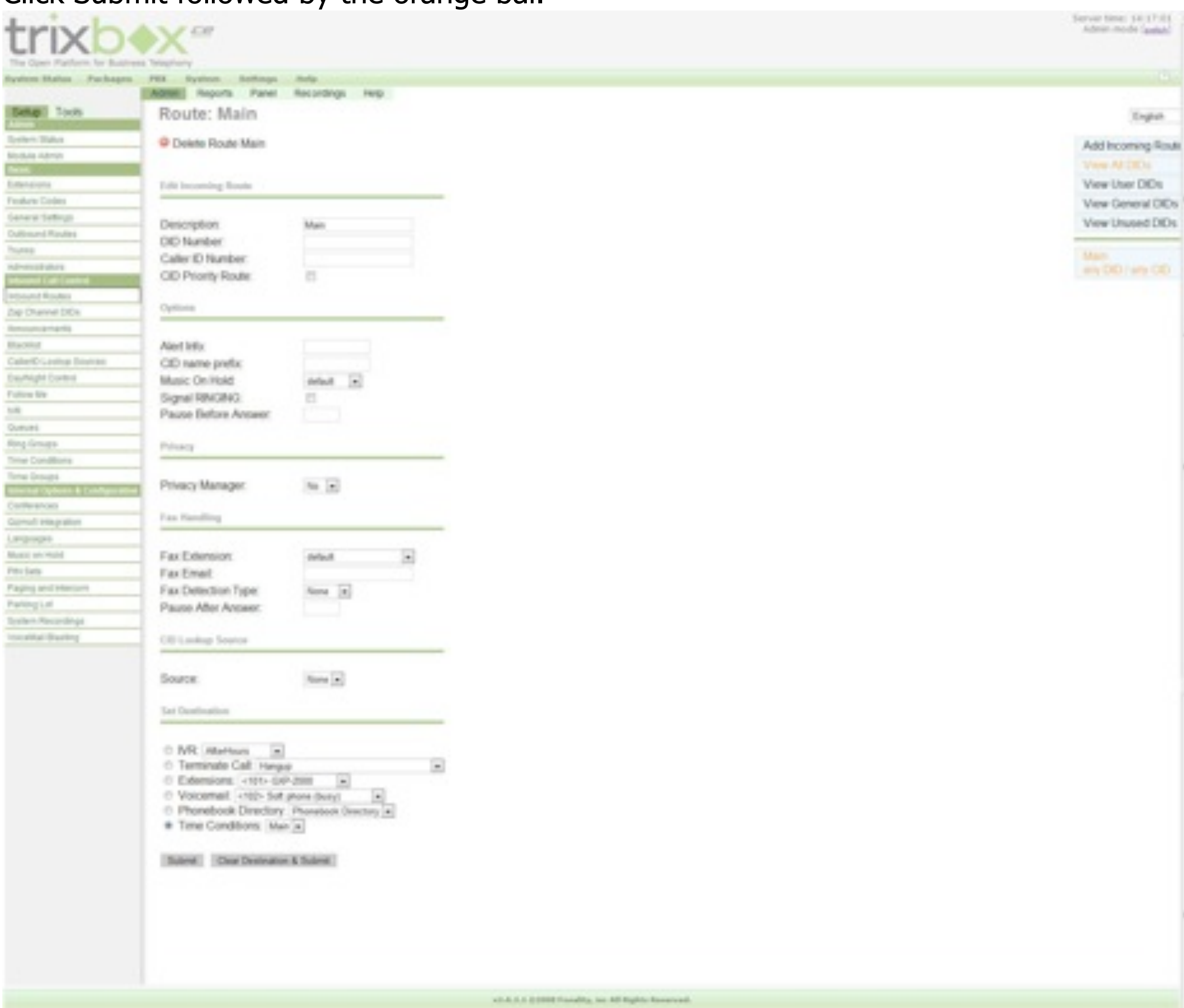
Description: Main

DID Number: <blank>

Caller ID Number: <blank>
CID Priority Route: unchecked
Alert Info: <blank>
CID name prefix: <blank>
Music On Hold?: Default
Signal RINGING: <Unchecked>
Pause Before Answer: <blank>
Privacy Manager: No (this prompts the caller to enter in their 10 digit phone number if they have their caller ID blocked)
Fax Extension: disabled (for now)
(Leave the rest of the fax stuff default)

Source: none

Set Destination:
Time Conditions: Incoming (the one we just created)
Click Submit followed by the orange bar.



At this point, we can simulate an incoming call by dialing 7777 from one of our extensions. You should hear one of the two recordings we did for the IVR.

11.0 - On Hold Music

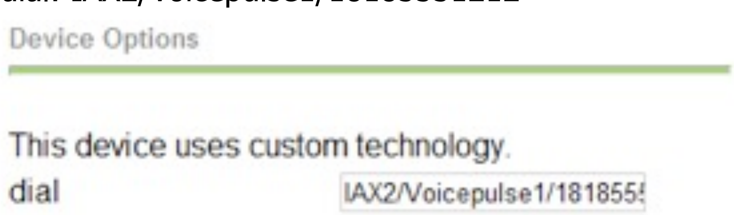
Next, we will set up our hold music.
Click on ‘Music On Hold’ under 'Internal Options & Configuration.' I typically delete all of the included wav’s and choose to upload my own. You have to always have 1 song available for hold music, so you’ll have to upload at least one song prior to deleting all of the included music.
Click on 'Choose File' and browse for a .wav or .mp3 file that you would like to be played as your hold music, and click ‘Upload.’ When a box pops up telling you that you need to wait for the page to reload, click 'OK.'
Once the page has reloaded, click on the orange bar at the top of the screen to apply the settings.
You can repeat this for as many songs as you’d like to upload.
One nice thing about Asterisk is that it only plays hold music while someone is actually on hold. So, if someone hears 30 seconds of an on-hold song, and then is taken off of hold, the song will pick up where it left off the next time someone is put on hold.
***TIP: You can create multiple Music Categories with different sets of hold music. This is helpful for situations where you want to perhaps play advertisements while people are waiting on hold in your Sales queue, or perhaps you want to give some tech tips when someone is waiting on hold for Tech Support. You can choose your Music on Hold Category per incoming route or per queue.

12.0 - Follow Me

Follow Me allows you to create a route (almost like a personal IVR) that calls will take to try to find you if you do not answer your extension. If I want it to be able to follow to my cell phone however, I first need to create a custom extension that dials my cell phone number.

12.1 - Cell phone extension

Click on 'Extensions' and then select 'Other (Custom) Device' from the drop-down box and click 'Submit'. Enter the following information:
User Extension: 112
Display Name: Cell phone
(leave all of the rest default until you get to 'dial.')
dial: (SIP or IAX2)/(trunk name)/18185551212 (my cell number on the end there)
So, for instance, if you were trying to dial out of your Voicepulse IAX2 trunk, you would fill in:
dial: IAX2/Voicepulse1/18185551212



Leave everything else default and leave voicemail disabled.

Click 'Submit' and then click the orange bar at the top to apply the changes to Asterisk.

12.2 - Follow Me

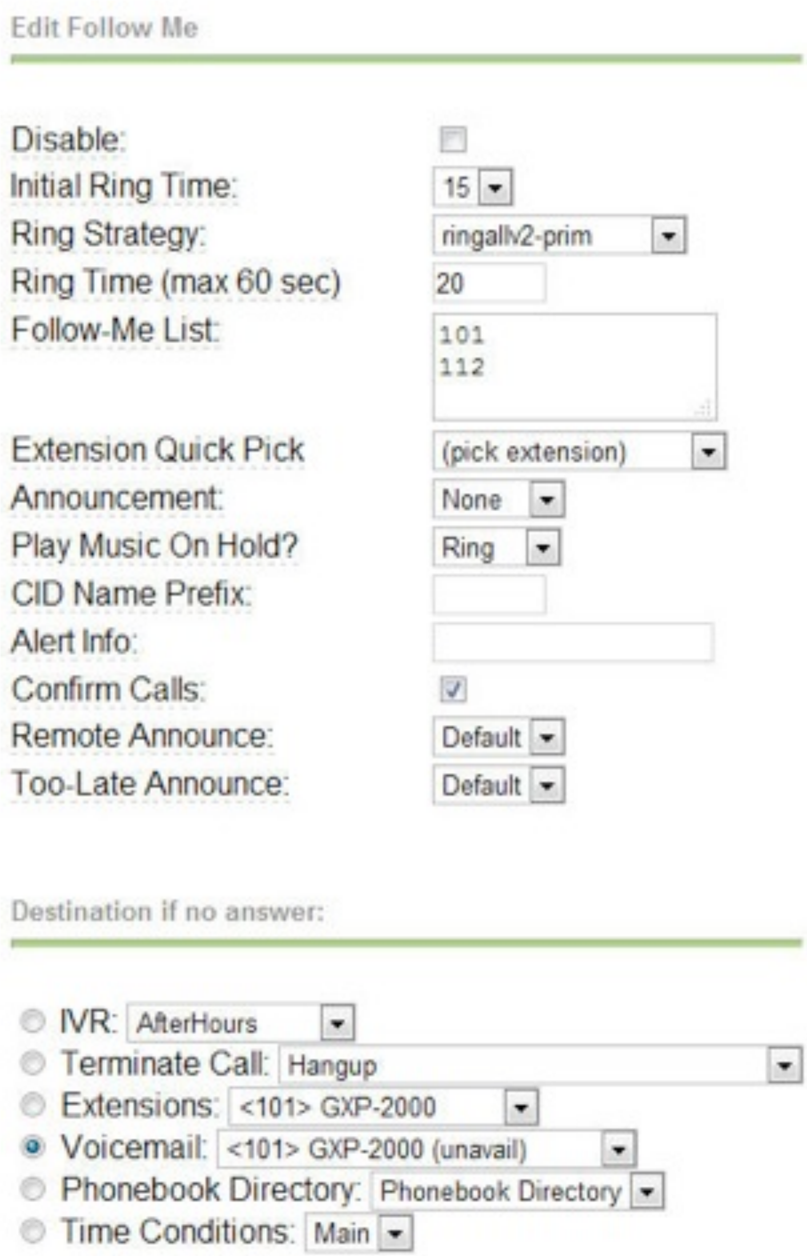
Now that we have our cell phone set up, let's configure follow me to first ring my extension (x101) and then try my cell phone (x112). Click on 'Follow Me' under 'Inbound Call Control,' and then click on the extension you want to set Follow Me settings on (in my case, x101). Use these settings:

Disable: unchecked (you can check this later if you want to temporarily enable/disable your Follow Me settings)
Initial Ring Time: 15 (about 3 rings)
Ring Strategy: ringallv2-prim (see below for more details about the various ring strategies)
Ring Time: 20
Follow Me List: 101 (already there by default, but add 112 on the next line)
112
Announcement: None
Play Music on Hold?: Ring (you can leave it ringing which is pretty normal, or you can opt for no sounds, or one of your Music on Hold Categories)
CID Name Prefix: <blank> (you can put something in here like 'Office:' and then your caller ID on your cell phone will say 'Office: <orig CID>'
Alert Info: <blank>
Confirm Calls: checked (this requires you to press 1 to answer a call that has been forwarded...check this box if you want the call to NOT go to your cell phone voicemail)
Remote Announce: Default (optional announcement when a call is going through Follow Me extensions)
Too-Late Announce: Default (optional announcement if you tried to answer a call going through Follow Me extensions, but someone else already grabbed it)

Destination if no answer: Voicemail - x101 (unavail) (my main trixbox voicemail)

Click 'Submit Changes' followed by the orange bar.

I chose 'ringallv2-prim' as my Ring Strategy. This means that first, it will try ringing my extension, and then it will ring both my extension and my cell phone extension. The '-prim' means that if I'm already talking on my main extension (x101), then trixbox will not use the Follow Me settings (cause it assumes that I'm at my desk and on the phone). Other strategies are hunt (round robin), memoryhunt (like round robin, but if it rang ext 1 and 2, it will pick up with ext 3 in the list on the next call). You can also choose to do firstavailable (ring the first available channel) or firstnotonphone (ring the first available channel which is not currently off hook). You can choose the best ring strategy for your situation.



13.0 Simple Queue setup

I would now like to set up a new queue which routes inbound callers to two of the extensions I created in chapter 8. For my queue, I will use x101, x102, and x103, and I would like to set them up to be a ring-all queue.
Click on Queues on the left hand side of PBX Settings.
I used the following information to set this up:
Queue number (extension for the queue): 150
Queue name: Sales
Queue password: <blank> (since I am going to use static agents, there is no need to have the agents log into the queue)

CID name prefix: <blank> (if you have multiple queues, you may want to take advantage of this...this prepends the caller ID with some text, for instance if you put 'Sales:' in here, the caller ID would show on the agent's phone as 'Sales:18185551212' so that the agent knows which queue the call is coming from)
Wait Time Prefix: No (this prepends the time that call has been waiting in the queue to the CID)
Alert Info: <blank>
Static agents: 101
102
103
Agent announcement: <none> (An optional recording played to the agent before getting the call)
Join Announcement: <none> (An optional recording played to the caller upon joining the queue...this is good if you have a DID pointing straight to the queue, and you want to announce something to the callers)
Music on Hold Class: Inherit
Ring tone instead of MOH: <unchecked> (who wants to hear ringing while they're on hold for 10 minutes??)
Max wait time: Unlimited
Max callers: 0 (unlimited)
Join empty: Yes
Leave when empty: No
Ring Strategy: ringall
Agent timeout: 15 seconds (how long an agent's phone can ring before it times out and moves to the next available agent)
Retry: 5 seconds
Wrap-up-time: 0 seconds (how long an agent has after a queue call before they can receive a new queue call)
Call recording: No
Event when called: No
Member status: No
Skip Busy Agents: Yes (if they're already on the phone, I don't want them to receive queue calls)
Queue weight: 0 (set this higher for higher priority queues...calls in higher priority calls will go through to agents first if the agents are members of multiple queues)
Autofill: unchecked (this will send multiple calls through to agents if multiple agents are available...if unchecked, it connects each call individually before going to the next queued call...this is helpful to check in higher-volume call centers)
Agent Regex Filter: <blank>
Caller Position Announcements
Frequency: 2 minutes (announce the caller's position and/or hold time every 2 mins)
Announce position: Yes
Announce Hold Time: Yes

IVR Break Out Menu: None (you can point this to another IVR if, for instance, you would like to give people on hold in the queue the option to press 8 to leave a message

instead of waiting).
Repeat Frequency: 0 seconds

Fail Over Destination:
Voicemail: 101 (unavail) (my receptionist extension...calls will go to this extension if they fail out of the queue for any reason)
Click 'Submit Changes' followed by the orange bar.
You should now be able to click on 'Panel' and see your new Queue. If you dial 150 from an internal extension, the call will ring x101, x102, and x103.

At this point, I may want to go back to my IVR and set option 4 to ring this queue.

14.0 - Panel configuration

Now that we have some good extensions, trunks, and queues in our trixbox, we should take a look at the FOP (Flash Operator Panel) or simply Panel. Click on 'Panel' at the top of the trixbox CE GUI, and you will be taken to the Panel. Once there, you will see your extensions, queues, trunks, conference bridges, and the parking lot all laid out nicely. If you double-click on any of the green buttons, you will be prompted for a password.

First, lets set that password to something we can use. (The default password is: passw0rd).

Go to your CLI and type the following:

```
cd /etc
nano amportal.conf
```

About 50 lines down, you'll see `FOPPASSWORD=passw0rd.` Change the 'passw0rd' to whatever password you would like, and hit `CTRL+X` and then `Y and ENTER` to exit nano.

Now, restart the panel with `amportal restart` from the CLI, and you're good to go.

There are a few useful things you can do with the FOP. For instance, if a call is in session, you can click on the RED button to disconnect the call. If you would like to call an extension, you can drag the phone icon located to the right of the extension number to another extension. This initiates the call to both phones. To transfer a call, drag the phone icon from the extension you want to disconnect to the new extension, and the call is transferred. For more information on the FOP, please see http://www.aussievoip.com/wiki/TB-FOP.

15.0 - HUDlite configuration

HUDlite and HUD Server are another set of tools for visually monitoring and controlling what your extensions are doing. The HUD product is put out by Fonality, the owners of the trixbox project...we can expect to see some exciting things happen with HUD/

trixbox integration. It has been announced that soon you will be able to purchase the full version of HUD 3.0 for trixbox CE! It has some amazingly powerful features, and it blows the balls off of the FOP.

For Trixbox v2.6.2, HUD Server and HUD Admin are not installed by default. To install them, we need to first go to the package manager.

Click on 'Packages' and see a list of available packages to install in trixbox. To make things easier, you may want to first go to 'Settings --> Repositories' and de-select everything but the 'trixbox Stable' package...this makes the list a lot shorter.

Scroll through the list of packages until you find 'tbm-hudadmin.noarch.' Check the box to the right of that package and then scroll down to the bottom and click 'Install.'

When prompted about whether or not you are sure you want to install the selected packages, click 'OK.' Your browser will now hang while it installs or upgrades the requested package(s).

Once it has completed, it will display a summary of the package install/upgrade transaction. If you scroll to the bottom and see 'Complete!,' then you have successfully installed or upgraded.

The HUDlite Admin now no longer needs us to put extensions in manually, however it does assign a random password to each of your extensions which should probably be changed to something easier to remember. Mouseover 'Asterisk' and select 'HUDlite Admin.' You should see all of your created extensions. Change the password for the extensions you want to and click 'Submit.'

*****NOTE:** HUDlite is not currently compatible with IAX or ZAP extensions.

Time to download and install HUDlite for another computer. They have versions for Windows, Mac, and Linux, but my document will focus on Windows since that is what I have installed. You can get HUDlite at www.hudlite.org. Download the latest version and install onto your PC using all defaults.

When you first open HUDlite, you will be given a setup wizard. On the first screen, enter the following information:

Username: (the HUD Login shown in HUDlite Admin)
Password: (the password shown in HUDlite Admin)
Next ->

Server name: (the IP address of your Trixbox server)
Password: (the HUD-Server password - by default it is 'password')
Server port: 6600
Next ->

Outlook Addin (choose to install the Outlook plug in now...you can always do it later if you want)
Install or Later -> (if you choose Install, a separate installation routine is opened...just take all defaults. You will have to close Outlook to install the addin.)

When the wizard finishes, HUDlite should now connect to your trixbox and show you the extensions you set up:

HUDlite tells you a few things when calls are in progress with colors:

Green - extension is on an outside call
Purple - extension is on an intra-office call
Orange - extension is on a queue call
Gray - extension is unavailable

You can drag and drop your extension (upper-left corner extension 101) to another extension to call that extension (will dial your phone first and then call the 2nd extension once you have picked up).

You can also drag and drop your call to the 'on hold' section in the upper left to put a call on hold (although this didn't seem to work too well for me). You can also drag and drop a call into the cassette tape icon of another extension to send that call directly to the voicemail of that extension.

To transfer a call to an employee's cell phone, just drag and drop to the cell phone icon (cell phone number must be set up in HUDlite Admin first).

HUDlite also allows you to Barge/Monitor a call in progress by clicking the 'B' next to a call that is in progress. This will ring your extension and when you pick it up, you will hear the conversation that is going on between the two other extensions, or to the outside world...press the 'B' again, and you can Barge the call. Muahahaha...big brother in the house.

16.0 - Tools

16.1 - Callback

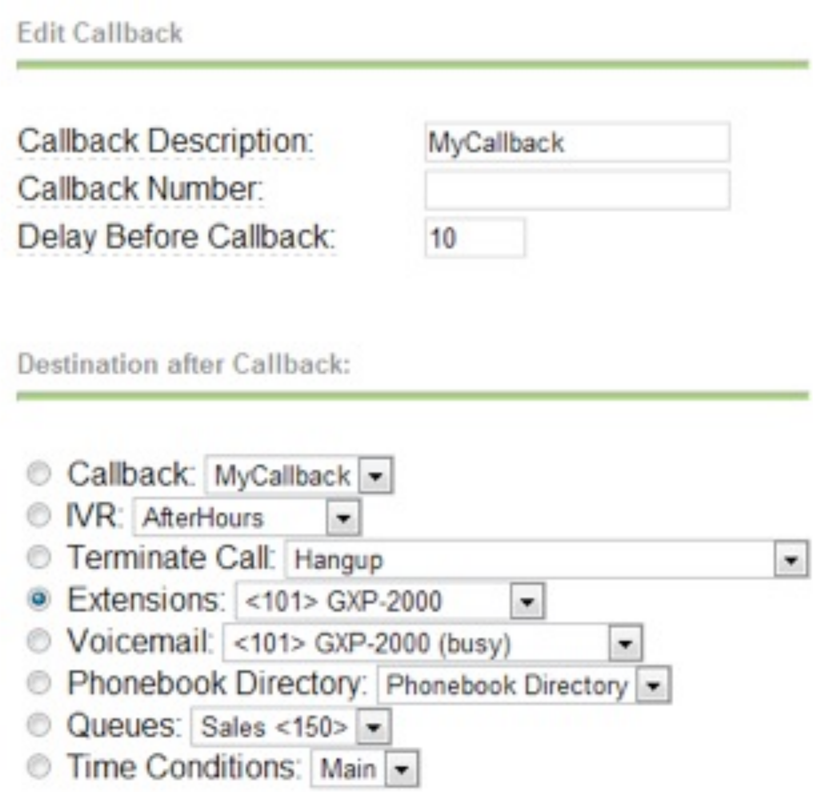
Callback allows a user to dial into the system, hang up, and then have the system call them back directing them to a specific location. This is helpful if for instance, you are using a cell phone to dial in, but do not want to pay for the minutes used. The system can call your cell phone back at the cheaper SIP/IAX rates of your trunks.

The first step in setting up Callback is that the module needs to be enabled. In PBX Settings, click on Module Admin. Then click on 'Check for Updates Online.' Once connected, click on 'Callback' to expand it, and then and click 'Download and Install'. Scroll down to the bottom of the screen and click the 'Process' button. Once it has downloaded and installed, click 'Return' to go back to the Module Admin.

Now, click on 'Setup' in the top toolbar and then 'Callback' on the left under 'Internal Options & Configuration.' Enter in the following information:

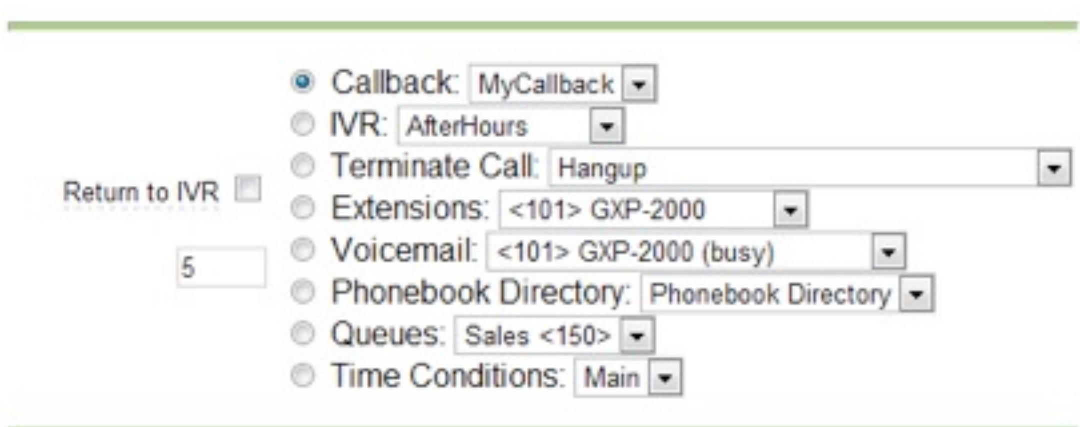
Callback Description: (whatever you want)
Callback Number: (enter in a specific number to call back, or leave blank to use the caller's caller ID)
Delay Before Callback: (the amount of time the system waits before calling the caller back...I put in 10 for 10 seconds)

Destination after Callback: (select your destination - I chose to have it ring one of my extensions x101 for testing purposes)



Now you need a way to access your Callback. I will do this through my main IVR as a hidden selection (by hidden, I mean that it will be an option in the menu, but not spoken to the callers in the announcement...I don't want EVERYONE to use this feature). :)

Click on 'IVR' and then click on your desired IVR (I'm using the 'BusinessHours' IVR). Click the 'Increase Options' button to create a new option...I will make Callback option 5. Then click the radio button for 'Callback' and choose the Callback that you just set up.



Click 'Save' and then the orange bar to apply changes.

Now, let's test it out. I call my phone number with my cell phone, and then press 5 when the announcement comes on. And the stupid computer hangs up on me!! Just kidding...that is what it is supposed to do. I can see that it has all of the options set properly, and the system now waits 10 seconds before dialing me back.

16.2 - DISA

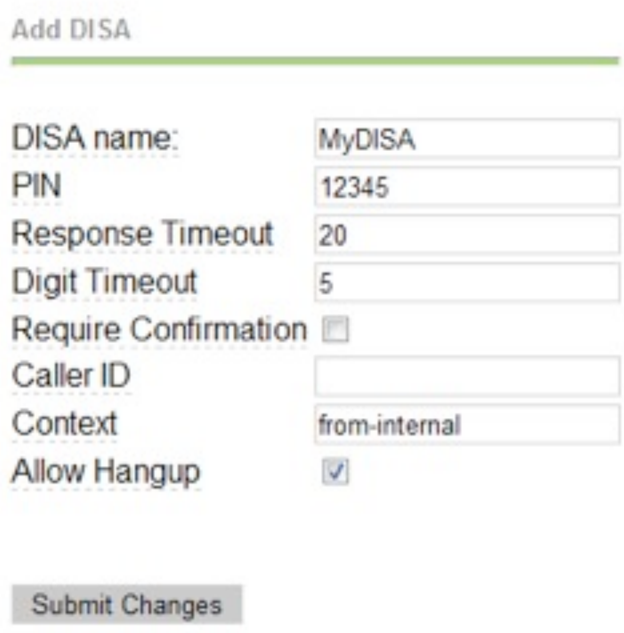
DISA stands for 'Direct Inward System Access.' It gives you the ability to have an option on an IVR that gives you a dial tone which, in turn, allows you to dial out from the trixbox as if you were using an internal extension. It is very handy for remote users who don't want to use their own money for long distance calls, or who want to appear to be calling out from within the office.

The first step in setting up DISA is that the module needs to be enabled. In PBX Settings, click on Tools --> Module Admin. Then click on 'Check for Updates Online.' Once connected, click on 'DISA' to expand it, and then and click 'Download and Install'. Scroll down to the bottom of the screen and click the 'Process' button. Once it has downloaded and installed, click 'Return' to go back to the Module Admin.

Now, click on 'Setup' in the top toolbar and then 'DISA' on the left under 'Internal Options & Configuration.' Enter in the following information:

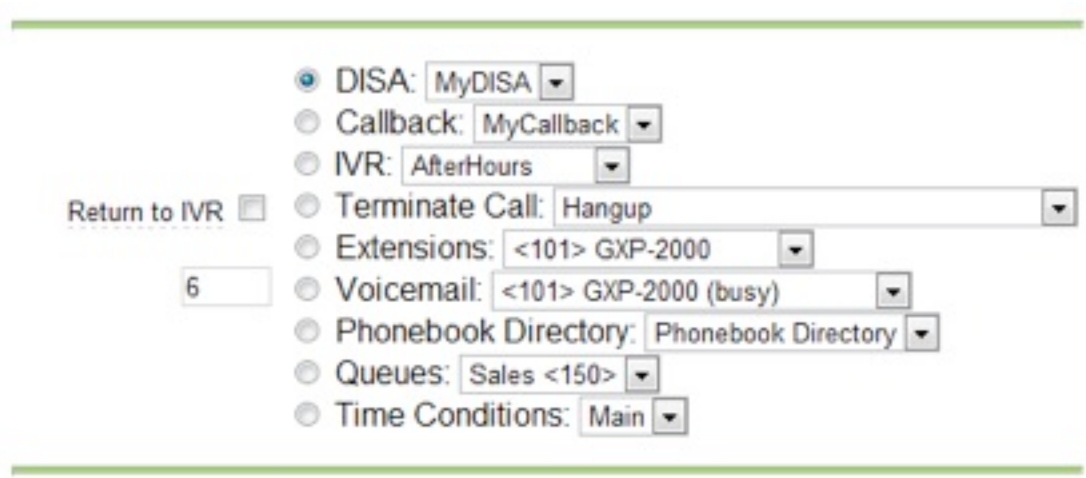
DISA name: (can be whatever)
PIN: (whatever DISA password you want...I used 12345)
Response Timeout: 20 (the amount of time in seconds the DISA user has to dial a phone number)
Digit Timeout: 5 (the maximum amount of time between digits dialed)
Require Confirmation: (unchecked)
Caller ID: <blank>
Context: from-internal
Allow Hangup: checked (this will allow DISA users to press ** to hang up a call and get dial tone again...great for making multiple DISA calls in a row)

Click Submit followed by the orange bar.



Now you need a way to access your DISA. I will do this through my main IVR as a hidden selection (by hidden, I mean that it will be an option in the menu, but not spoken to the callers in the announcement).

Click on 'IVR' and then click on your desired IVR (I'm using the 'BusinessHours' IVR). Click the 'Increase Options' button to create a new option...I will make DISA option 6. Then click the radio button for 'DISA' and choose the DISA that you just set up.



Now call in, press 6 and you should get a dial tone. Dial a new number and your system will call out.

*****NOTE:** Since the DISA context is set to 'from-internal,' you need to dial numbers using DISA exactly as you would if you were dialing from an extension attached to the PBX. By that I mean...if your dial plan requires you to dial 9 to get out...you need to dial 9 to use DISA as well.

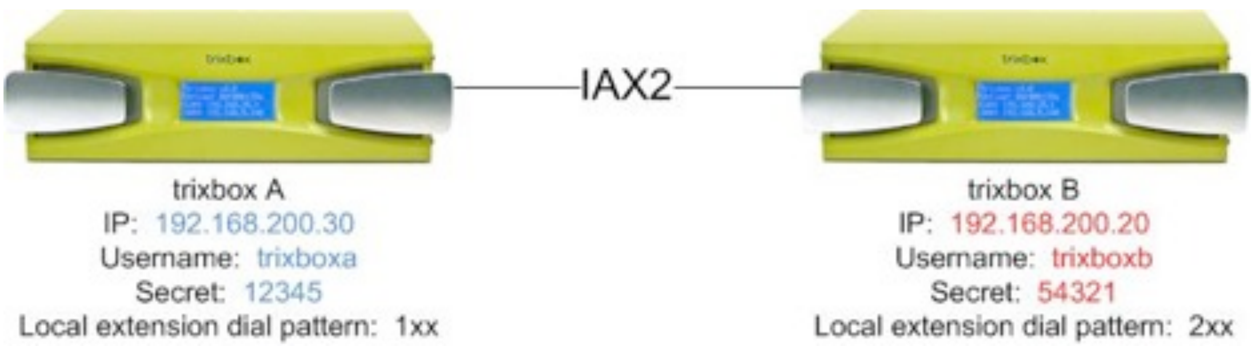
16.3 - Trixbox to Trixbox IAX2 trunk

This section will explain how to connect two trixbox PBX systems together. It assumes that you have 2 trixbox v2.6.2 systems, but this should work with other versions of trixbox as well.

This setup can be confusing, so it would be best to get all of our information gathered up before starting. Fill in your own trixbox information for the following fields:

trixbox A
Hostname or IP: 192.168.200.30
Username: trixboxa
Secret: 12345
Local extension dial pattern: 1xx

trixbox B
Hostname or IP: 192.168.200.20
Username: trixboxb
Secret: 54321
Local extension dial pattern: 2xx



*****NOTE:** This document assumes that the two trixbox systems are on the same local subnet, however if they aren't, you will want to make sure that UDP port 4569 is open in the firewall on both sides to the trixbox systems, or that you are using some sort of VPN.

*****NOTE:** I have used blue and red to distinguish between the two systems...the color coding has nothing to do with the configuration...it just makes it less confusing.

Let's start by configuring trixbox A. Click on Trunks --> Add IAX2 Trunk and enter in the following information:

Outbound Caller ID: (can be whatever)
Never Override CallerID: unchecked
Maximum Channels: <blank>
Disable Trunk: unchecked
Monitor Trunk Failures: unchecked

Dial Rules: 2xx (the dial pattern for the trixbox B extensions)
Outbound Dial Prefix: <blank>

Outgoing Settings (these are the settings needed to create a trunk TO trixboxa)
Trunk Name: TotrixboxB

Peer Details:

```
host=192.168.200.20
username=trixboxb
secret=54321
type=peer
trunk=yes
```

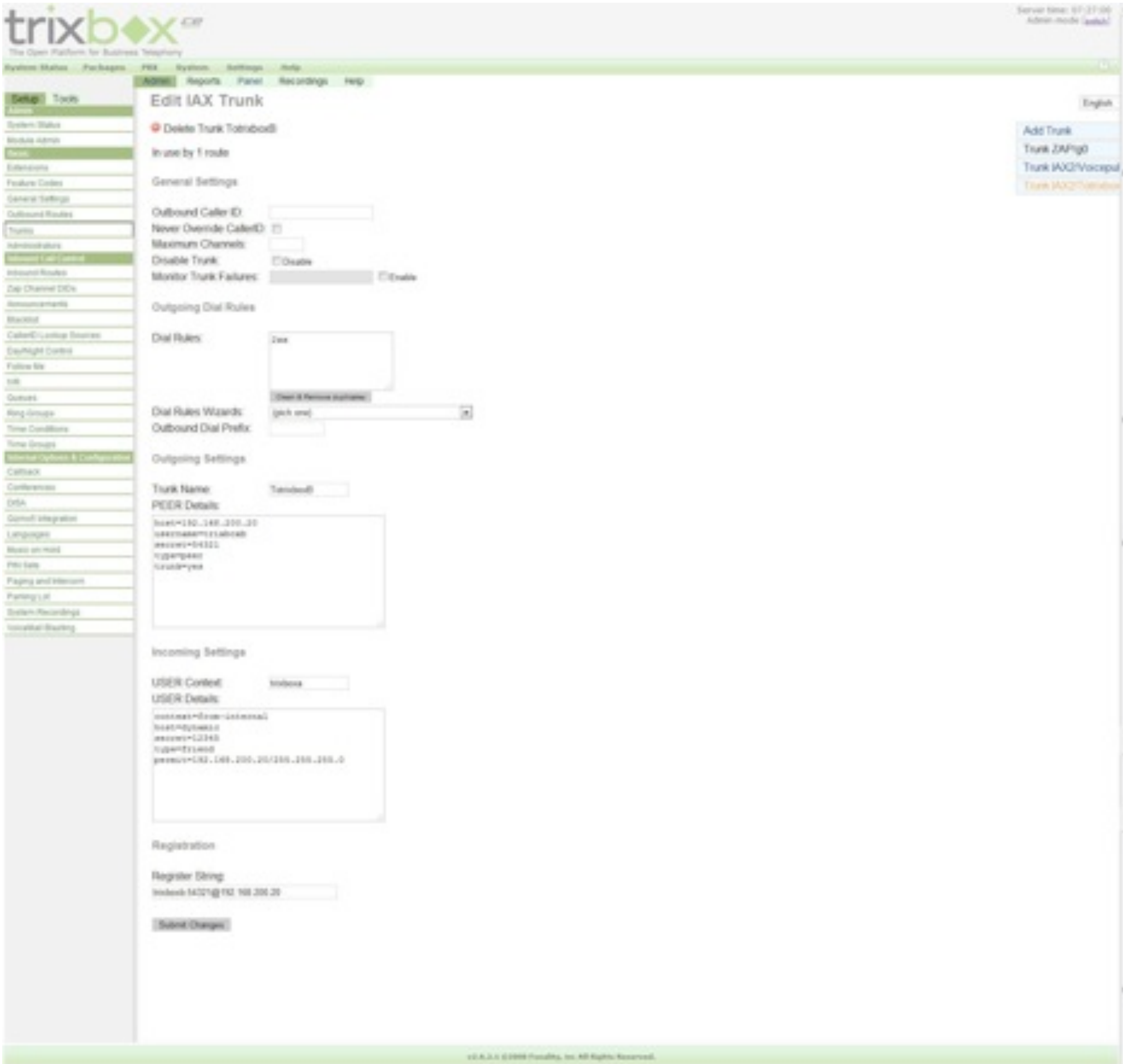
Incoming Settings: (these are the settings needed to create a user for trixboxb to connect to us)
User Context: trixboxa

User Details:

```
context=from-internal
host=dynamic
secret=12345
type=friend
permit=192.168.200.20/255.255.255.0
```

Register String: trixboxb:54321@192.168.200.20

Click Submit followed by the orange bar to apply changes.



Now switch over to trixbox B and click on Trunks --> Add IAX2 Trunk and enter in the following information:

Outbound Caller ID: (can be whatever)
Never Override CallerID: unchecked
Maximum Channels: <blank>
Disable Trunk: unchecked
Monitor Trunk Failures: unchecked

Dial Rules: 1xx (the dial pattern for the trixbox A extensions)
Outbound Dial Prefix: <blank>

Outgoing Settings
Trunk Name: TotrixboxA

Peer Details:

```
host=192.168.200.30
username=trixboxa
secret=12345
type=peer
trunk=yes
```

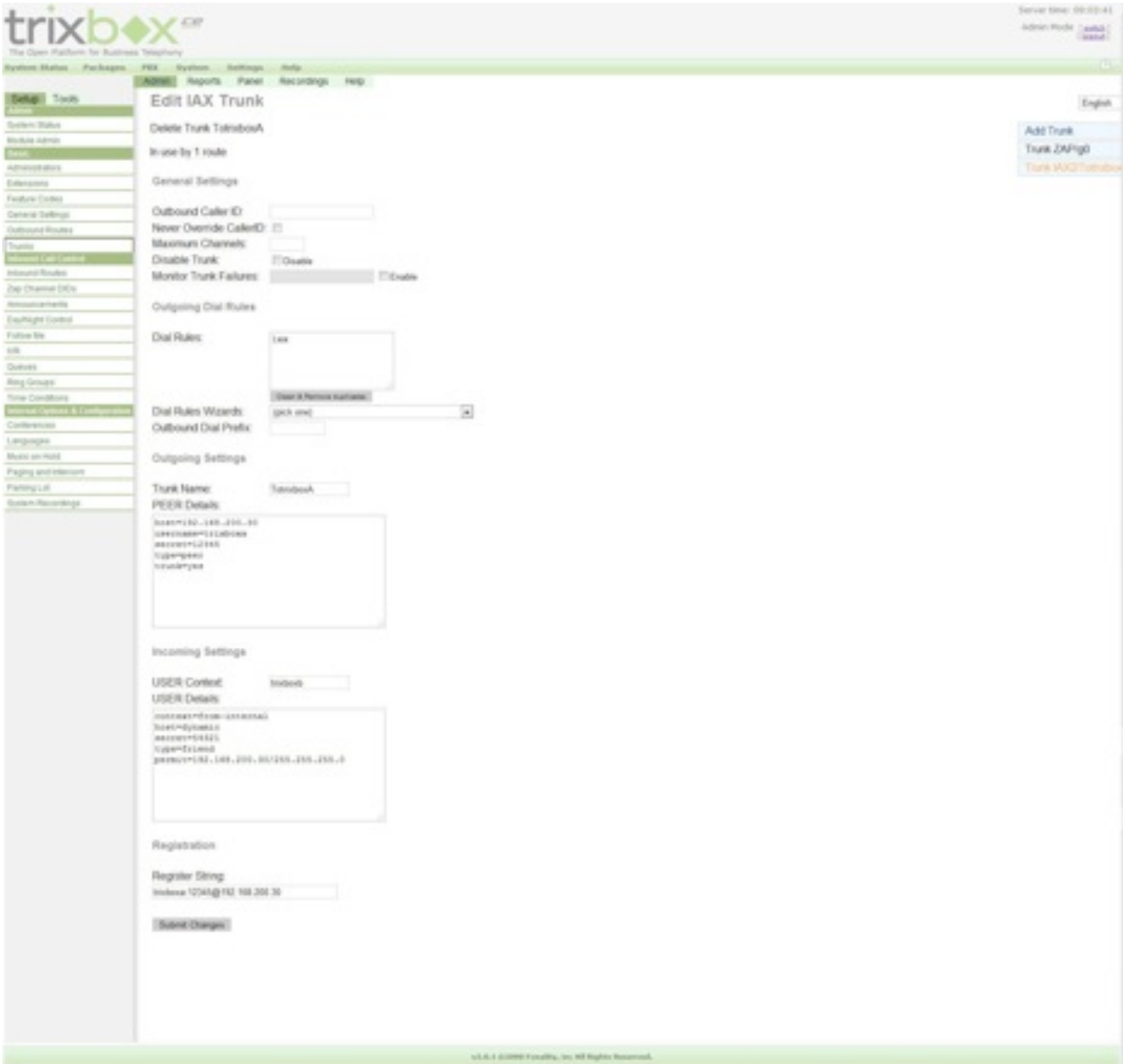
Incoming Settings:
User Context: trixboxb

User Details:

```
context=from-internal
host=dynamic
secret=54321
type=friend
permit=192.168.200.30/255.255.255.0
```

Register String: trixboxa:12345@192.168.200.30

Click Submit followed by the orange bar to apply changes.



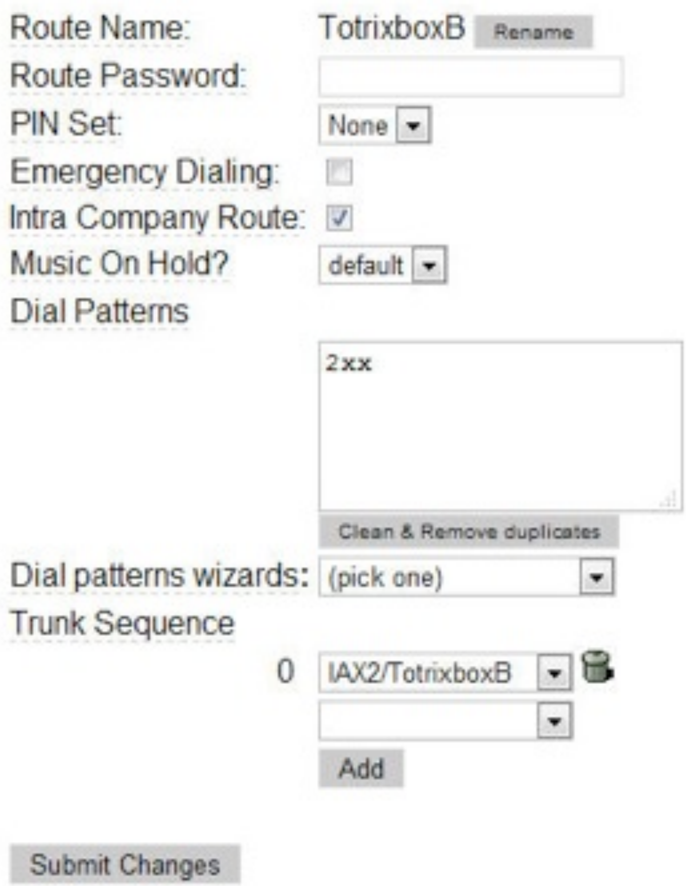
Now that we have our trunks set up on both boxes, we need to add some outbound routes that use these trunks. On trixbox A, click on Outbound Routes and enter in the following information:

Route Name: TotrixboxB
Route Password: <blank>
PIN Set: None
Emergency Dialing: unchecked
Intra Company Route: checked
Music On Hold?: default

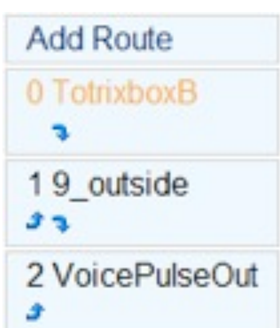
Dial Patterns: 2xx

Trunk Sequence: IAX2/TotrixboxB

Click Submit followed by the orange bar to apply changes.



You will also want to click the 'up' arrows on this trunk underneath 'Add Route' to bring this route to the top of the list of routes. This way, it will check for this intra-company route before any of your other outbound routes.



Now we need to do the same thing on TrixboxB. Click on Outbound Routes and enter in the following information:

Route Name: TotrixboxA
Route Password: <blank>
PIN Set: None
Emergency Dialing: unchecked
Intra Company Route: checked
Music On Hold?: default

Dial Patterns: 1xx

Trunk Sequence: IAX2/TotrixboxA

Click Submit followed by the orange bar to apply changes.

You should now be able to call 2xx trunks on trixbox B from trixbox A and 1xx trunks on trixbox A from trixbox B.

16.4 - Parking Lot

A nice feature of trixbox CE is the parking lot. This allows a receptionist or user to 'park' a call by transferring to the call park extension. The call is essentially placed on hold for anyone to pick up by dialing the parking lot extension they were placed on.

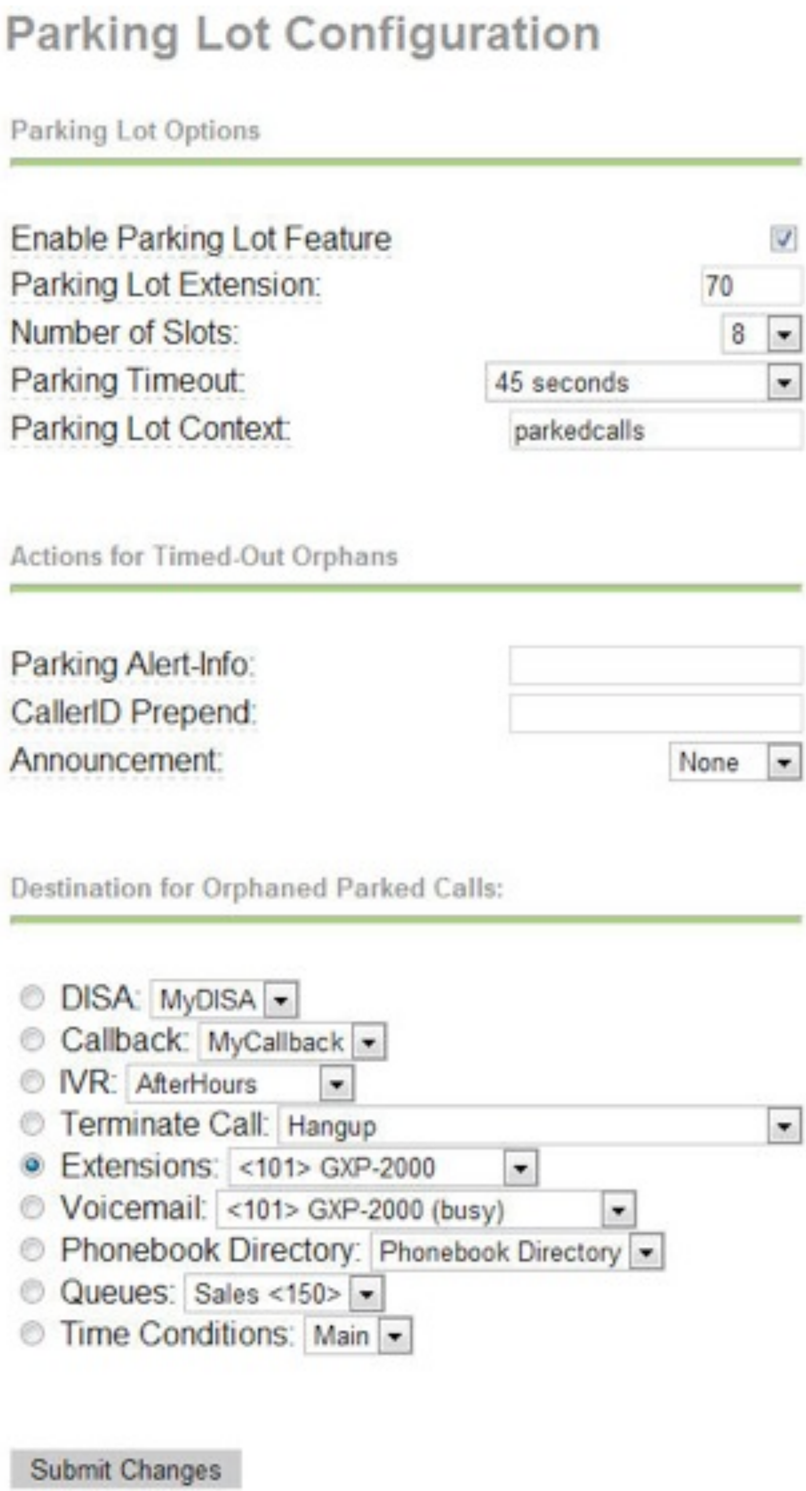
Click on Setup --> Parking Lot in the 'Internal Options & Configuration' and use these settings:

Enable Parking Lot Feature: checked
Parking Lot Extension: 70 (***Note: I have seen a few installations where the Parking Lot does not work on extension 70, but if you change it to 700, it works fine)
Number of Slots: 8 (This is the number of parking lot 'spaces' available for putting calls on hold...unless this is a BIG install, you should never need more than 8)
Parking Timeout: 45 seconds (the amount of time a call can sit in the parking lot before going back to the original person who put the call on hold)
Parking Lot Context: parkedcalls (take the default)

Parking Alert-info: <blank>
CallerID Prepend: <blank>
Announcement: None

Destination for Orphaned Parked Calls: (I set this to my main phone...you would most likely want to put the receptionist's extension in here).

To use the Parking Lot, transfer a call to the Parking Lot Extension above. You will then hear a voice tell you which space it put the call into (ie. 'seven one'). If you used 70 as your parking lot extension, your spaces are 71 through 78 (assuming 8 slots selected above). If you heard 'seven one' you then dial 71 to pick up the call.



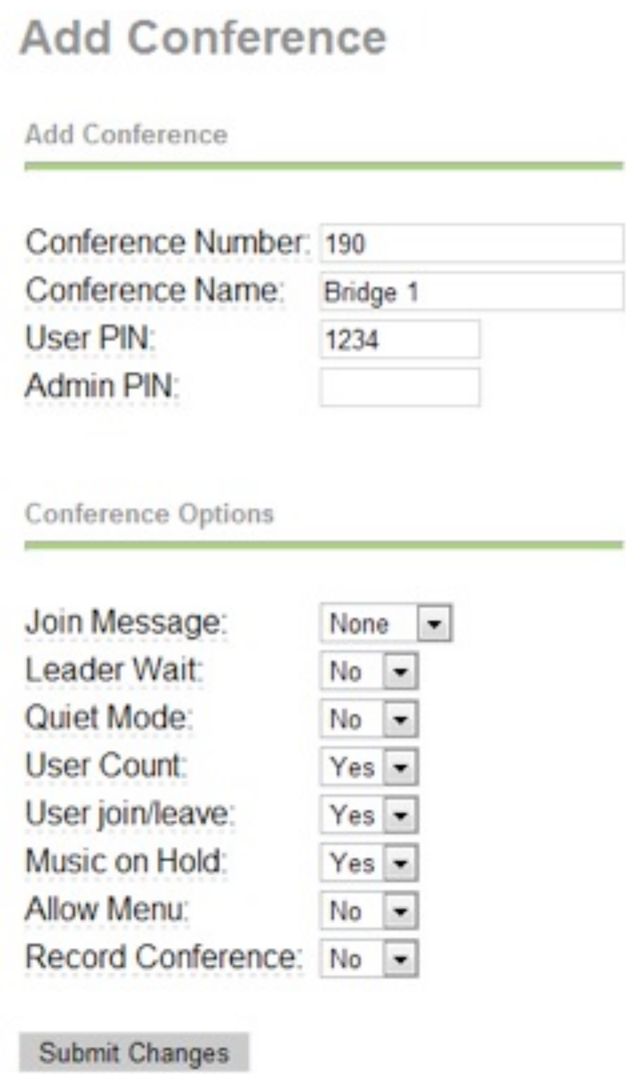
16.5 - Conferences

To add a conference bridge (or multiple bridges) to your system, open up PBX Settings and click on 'Conferences' under 'Internal Options & Configuration.' Use these settings:

Conference Number: 190 (can be whatever - this is your conference bridge extension)
Conference Name: (can be whatever)
User PIN: 1234 (if left blank, there will be no conference password for users)
Admin PIN: <blank> (if left blank, there will be no conference Admin functionality)

Conference Options:
Join Message: None (recording played to conference attendees upon entering the conference bridge)
Leader Wait: No (plays hold music for everyone until a conference leader shows up)
Quiet Mode: No (turns off join/leave notifications when set to yes)
User Count: Yes (tells the attendee how many people are on the bridge when they join)
User join/leave: Yes (plays a sound when users join or leave)
Music on Hold: Yes (plays music instead of silence when there is only 1 person on the bridge)
Allow Menu: No (this enables the Admin menu if you set an Admin password above)
Record Conference: No

Click 'Submit' followed by the orange bar.



16.6 - Web MeetMe

Web MeetMe is a User mode program which allows your users to add/delete/schedule conferences. If you are in Admin mode, you will need to click the 'switch' button in the upper left-hand corner of the screen to switch yourself back to User mode.

First things first though...there is a typo in one of the .php files for Web Meetme (this may be corrected in a future version, so if you are reading this, just check to see if this typo exists). Go to the Linux CLI and type this in:

```
nano -w /var/www/html/web-meetme/user_add_sqldb.php
```

Press CTRL+W to search and type in 'userPassr' and press Enter. If you get taken to a result, change 'userPassr' to 'userPass' (removing the 'r' on the end). If you get no result, you are done...the typo has been fixed already. Press CTRL+X followed by 'Y' to save and exit. (Or just CTRL+X if no changes were made).

Back in the User mode GUI, click on 'MeetMe' in the menu bar. When you click on anything, you will be asked for the admin email and password. By default, the admin email and password is:

U: wmm@localhost
P: wmmpw

Lets start by changing our wmm@localhost password to something else. Click on 'Update User' followed by wmm@localhost. Set the 'User email:' and 'Password:' fields to something else and click 'Update User.'

*****NOTE:** If you try ONLY changing the password, it will probably give you an 'invalid email' error message. So be sure to change both the admin email and password.

Now that we have updated our Web Meetme Admin user, let's add a conference. Click on 'Add Conference' to get started. Enter in the following information:

Conference Name: (whatever)
Conference Owner: (change this to the email address you used for the Admin user above)
Conference Number: default (or change this to an easy to remember conference number that matches your internal dial plan)
Moderator PIN: <blank> (or put in a PIN code for the Moderator/admin of the conference)
Moderator Options: unchecked
User PIN: <blank> (or put in a PIN code for the conference users)
User Options: unchecked
Start Time: Enter in your start time
Duration: Enter in your conference duration
Recurs: unchecked (check if you want this to be a Daily/Weekly/Bi-weekly reoccurring conference)
Max Participants: 10

Click on 'Add Conference' to schedule your conference bridge. You should now be able to click on 'Future Conferences' to see your newly created conference bridge. When the time comes, the bridge will be created automatically, and will be removed once the Duration has been achieved.

*****NOTE:** Generally, I try to stay away from Web Meetme. Unless you have a large number of employees who need to be able to schedule their own bridges constantly, I would recommend setting up 4-5 conference bridges in PBX Settings (as shown in chapter 16.5) and then leaving them static. Or, perhaps create each employee that needs conferencing their own conference bridge. Why you ask? Because Web Meetme is flakey, and doesn't really work that well. Perhaps it will be more stable in a future revision...it's a good concept, but not ready for an enterprise environment (in my opinion).

16.7 - Bulk Extensions

trixbox CE includes a Bulk Extension module for creating multiple extensions. You need to provide a comma delimited (.csv) file that you can create with either Excel or a text file editor. Once you have created the file, you can then upload the file to trixbox CE and have many extensions created at once.

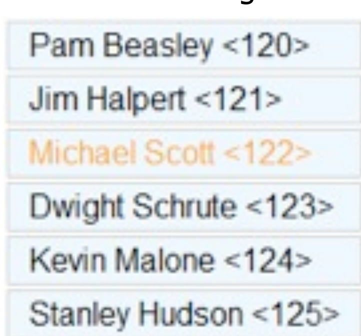
The .csv file needs to be in the format:

Extension, Name, Secret, Voicemail, VM Password, VM Email

So, if I wanted to create x120 through x125 for my trixbox CE box, my .csv file would look like this:

```
120, Pam Beasley, 12345, 120, 120, pam@dundermifflin.com
121, Jim Halpert, 12345, 121, 121, jim@dundermifflin.com
122, Michael Scott, 12345, 122, 122, michael@dundermifflin.com
123, Dwight Schrute, 12345, 123, 123, dwight@dundermifflin.com
124, Kevin Malone, 12345, 124, 124, kevin@dundermifflin.com
125, Stanley Hudson, 12345, 125,125, stanley@dundermifflin.com
```

I have saved that file to my desktop and called it 'extensions.csv.' I now click on the 'Choose File' button and browse to that file. Once it is listed, I click 'submit.' trixbox CE thinks for a few moments, and then tells me that my upload was accepted. If I now go to 'PBX Settings' and click on 'Extensions,' I can see all of my newly created extensions, and further configure them from there.



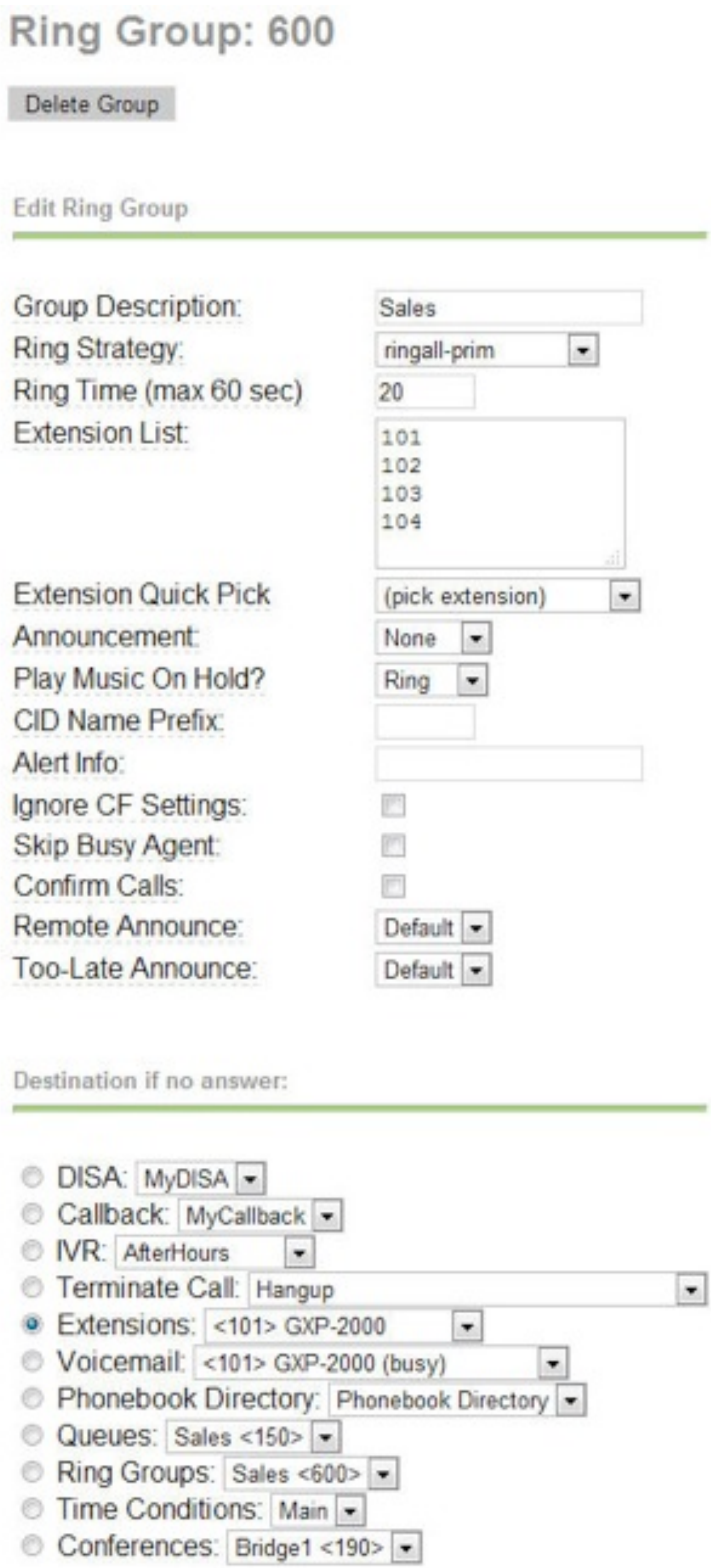
16.8 - Ring Groups

Ring groups are pretty straight forward and easy to configure. Basically, a ring group is an extension in trixbox CE that rings multiple phones when dialed. To set up a ring group, click on 'Ring Groups' under 'Inbound Call Control' in the Admin GUI. Enter in the following information:

Ring-Group Number: 180 (can be whatever...this is the extension number for this ring group)
Group Description: Sales (can be whatever)
Ring Strategy: ringall-prim (this will ring any phones in the ring group that aren't currently in use)
Ring Time: 20 (about 4 rings)
Extension list: (put in the extensions you want to ring when someone dials this ring group)
Announcement: None (if you want a recorded message to play when someone dials this ring group)
Play Music on Hold?: Ring
CID Name Prefix: <blank>
Alert Info: <blank>
Ignore CF Settings: unchecked
Skip Busy Agent: unchecked (pretty much the same as selecting ringall-prim above)
Confirm Calls: unchecked (when checked, agents have to press 1 to answer the call)
Remote Announce: Default
Too-Late Announce: Default

Destination if no answer: Extensions - x101 (my receptionist)

Click 'Submit Changes' followed by the orange bar. Your ring group has been created and can now be used directly or through the IVR.


Ring Group: 600
Delete Group
Edit Ring Group
Group Description: Sales
Ring Strategy: ringall-prim
Ring Time (max 60 sec) 20
Extension List: 101
102
103
104
Extension Quick Pick (pick extension)
Announcement: None
Play Music On Hold? Ring
CID Name Prefix:
Alert Info:
Ignore CF Settings:
Skip Busy Agent:
Confirm Calls:
Remote Announce: Default
Too-Late Announce: Default
Destination if no answer:
DISA: MyDISA
Callback: MyCallback
IVR: AfterHours
Terminate Call: Hangup
Extensions: <101> GXP-2000
Voicemail: <101> GXP-2000 (busy)
Phonebook Directory: Phonebook Directory
Queues: Sales <150>
Ring Groups: Sales <600>
Time Conditions: Main
Conferences: Bridge1 <190>

17.0 - Backups

One of the most important parts of running a successful trixbox server (or any enterprise product) is ensuring that you have reliable backups. This section will explain how to create a scheduled backup and how to automate the export of that backup to an FTP server.

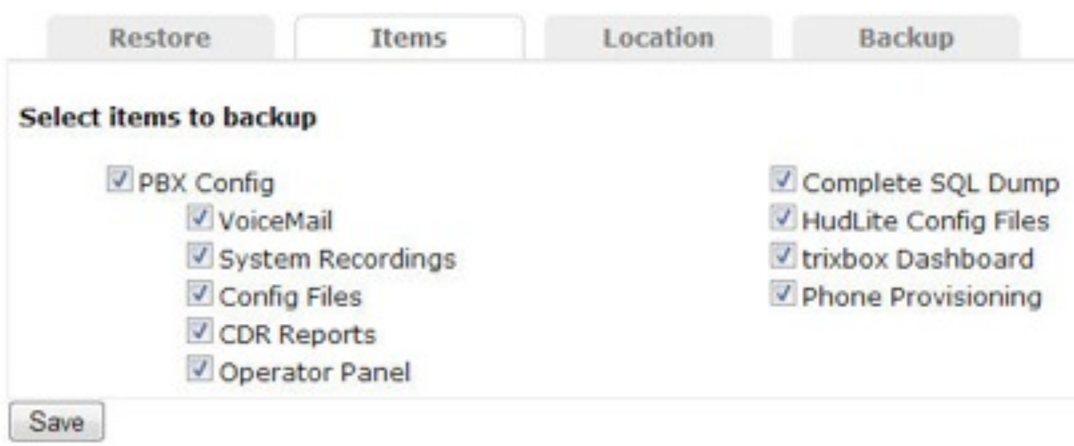
17.1 - Install the backup package

To install the backup package, you need to be in the Admin mode GUI. Click on 'Packages' on the menu bar and wait a few moments for the package list to load. Find the package named 'tbm-backup' and click the checkbox next to 'Install.' Once the installation is complete, you will now be able to click 'System --> Backup' from the menu bar in the Admin GUI.

17.2 - Configure a backup

To configure a backup of your system, click on 'System --> Backup' in the Admin GUI. In the box at the bottom of the backup module, enter in a name for your backup (such as 'Weekly') and click the 'New' button. Your new backup name appears in the box above...click on it, and the details of the backup appear to the right.

There shouldn't be anything listed under the 'Restore' tab yet, so just click on the 'Items' tab. Here you are able to select all of the items you want backed up. For our purposes, I will click on all of them and then click 'Save.'

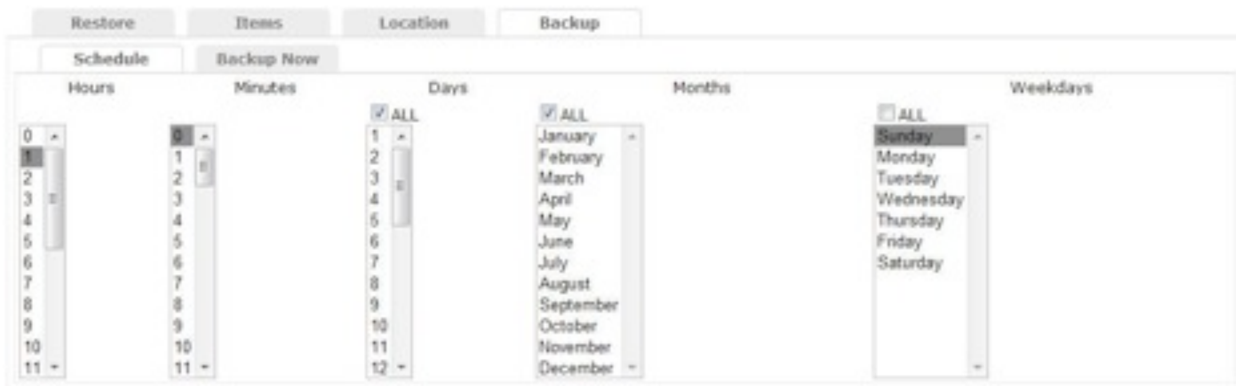


The next tab is the 'Location' tab. This is where you can specify alternate locations (such as another place on the local HDD, or an FTP site) for your backup. If you leave everything blank (default), backups are stored in /backups/(backup name) on the local HDD.

The last tab is the 'Backup' tab. This is where you can set the schedule for when to back up your system. Since I am doing a weekly backup, I want to back up my system every Sunday at 1am since there is little or no activity at that time. I choose the following settings:

Hours: 1
Minutes: 0
Days: check the ALL box
Months: check the ALL box
Weekdays: Sunday

So this means that I would like to back up at 1:00am, any calendar day (that matches Sunday), any calendar month, and on Sundays.



Since there is no 'Save' button on this screen, click back to either 'Items' or 'Location' and click 'Save.' Your backup is now scheduled, but it would be prudent to go back to the 'Backup' tab, click on 'Backup now' and then click 'Start Backup.' That way you have a working backup of the current configuration, and then backups will be done automatically from here on out.

***TIP: I HIGHLY recommend using both a local and FTP backup. That way, if your trixbox CE's HDD fails, you have a remote copy to work with. If you are a consultant setting up a system for a customer, you can schedule a daily or weekly FTP backup to your own servers and then add 'remote backups' to your list of provided services...perhaps with the purchase of a service contract.

A note from the author:
Hey everyone!
If you have found this document useful and plan on purchasing some VoIP equipment, I would greatly appreciate if you use the banner link for Voipsupply.com below. I have purchased equipment from Voipsupply.com many times, and I have always found their pricing to be great and their shipping fast, so I definitely recommend them. By using the link below, you won't be raising YOUR purchase price at all...you would simply be donating a small percentage of your purchase to me, and I would definitely appreciate it!
Thanks,
-Chris Sherwood

My Voipsupply.com banner link: