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NOTE: Because of the fast pace of software development it is possible that there will be minor differences between the manual and the actual release of the program.

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Overview

The Ringdale IP-PBX is ideally suited to the home, small business and home-office environments. It consists of a small yet powerful solid-state PBX Appliance that supports up to 12 SIP phone handsets and up to 4 external analog (POTS) lines. This document is intended as an installation guide to get the Ringdale IP-PBX Appliance system up and running. There are many configurations that this PBX system supports. Not all possibilities are considered here. There are many other resources to assist in configuring your system. Some of these are listed in the appendix.

The Ringdale IP-PBX is wholly based on PiaF (PBX in a Flash) which runs on the CentOS version of Linux. PiaF uses Asterisk to run the PBX, FreePBX is the interface to configure Asterisk. Like with all Linux applications, you can edit the conf files that run Asterisk, but FreePBX will overwrite all files except the custom files, and some exceptions.

Hardware Platform

IP-PBX Appliance 00-32-4000-0001

This thin client contains the following Operating System:

Operating System & Software

CentOS release 5.2 (Final) - 32 Bit Kernel: 2.6.18-92.1.6.el5. This is a version of Linux.

PBX in a Flash Version 1.3 Daemon Status

Asterisk 1.4 This is a Linux based IPBX application developed by Mark Spencer of Digium™, the company behind Asterisk.

Telephones

Telephone Deskset, SIP-based and Ethernet-Connected 00-11-0998-0000

NOTE: PC with a browser required (not included)

While it is not absolutely necessary, most of the configurations are done using a browser. Only a Linux programmer (and a pretty insane one at that) might be able to complete an installation without one. Configuration without a browser is not supported.
Hardware Installation

The hardware installation requires the following:

1) Connect the PBX to a monitor and keyboard. (The monitor and keyboard are not supplied.) You do not need a mouse.

2) Connect the PBX to an Ethernet network that is not exposed to the Internet. (Cables not supplied.)

3) Connect a minimum of 1 phone line. While it will function with one line, with two lines or more available, you can transfer a call to an outside line.

4) Install the phone(s). There is a switch on the back of the phone so you can use one long cable to your network and use a short cable to connect a PC.

5) Connect the handset and power the phone up.

Initial Software Configuration

When you boot the PBX Server you may see some errors. This is normal. Once the system rebooted, you will be greeted by the PBX in a Flash splash screen. After loading all the modules, it will continue loading PiaF. Once it’s finished doing its thing you will be greeted with the Linux login prompt:

root@pbx:~ $

Changes to Default Settings

Once PiaF has been installed, some changes need to be made to secure your installation. Log in to your new PBX (user: root, password: 123456).

To get Help

At the command line, type

help-pbx

A list of help will be displayed.

The help screen will also tell you the necessary command required to change the password of the various default users e.g. passwd-maint for user maint and passwd for user root.

Press the Enter key to see more of the commands.

For more information on how to use Linux commands, see the Linux Commands.
Change IP Address (set IP address to Static)

If you are using PiaF, it is highly recommended that you use Webmin to change the IP address
(See Webmin for PiaF to run webmin).

However if you use Netconfig, you may have to run Netconfig again after the first time and after rebooting your PC. To change Asterisk IP address from DHCP to Static. At the command prompt enter:

```
Netconfig etconfig etconfig etconfig
```

You will see the following:

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

You will then see the following screen.

Use the Tab key to cycle through the fields. Enter the IP address that is to be allocated to the Asterisk box, the Netmask (subnet mask), Default Gateway, and Primary nameserver as per the example above.

- In the IP address field, enter an IP address PiaF making sure it is within your network range e.g.: 192.168.1.100
- Netmask is normally 255.255.255.0 unless your network has different Network mask.
- Default gateway IP is the address of your router. In my case, my router address is 192.168.1.1
- Primary nameserver is the address of your Name Server, usually your domain server if you have a network domain server but if you are running workgroup, use the address of your default gateway e.g.: 192.168.1.1

Once done, select OK.
1. CentOS

CentOS is a distribution of Linux that runs a Security Enhanced Linux kernel called SELinux. Linux as used in PBX in a Flash is the command line operating system. It also uses logical volume management LVM instead of hard-drive partitions. The interface to the CentOS/Linux operating system is referred to as the Linux command line.

2. Asterisk PBX

The core application is Asterisk PBX which provides the telecommunication functionality on the PC. The interface to Asterisk PBX is either through the Asterisk command line interface referred to as Asterisk CLI or through the web-based interface: FreePBX.

3. FreePBX

FreePBX is the web based interface that is used to configure the Asterisk PBX server from another PC's web-browser. It is a graphical user interface (GUI). It is an amazing combination of applications that does pretty much everything you would want in configuring an Asterisk PBX Server.

4. Webmin

Webmin is the web based interface that is used to configure Linux and the servers that are running on it.

5. MySQL Database

The configuration data for Asterisk is saved in a database running on a mySQL server. FreePBX and Webmin use the data to form the text configuration files for Linux, the support servers and the Asterisk PBX server. mySQL is running in the background and is transparent to the user (the user does not know that it exists).

6. Daemons

There are many server applications (daemons) running in the background that we don't really see. For example: a web server, FTP server, email server, DHCP server, etc. These daemons are required to make the PBX in a Flash system work seamlessly.
**PiaF Menus**

Once the PBX has an IP address accessible from your network, you can connect to it using your browser at: http://ipaddress/ (e.g. http://192.168.1.100) to configure PiaF.

You will be presented with the PiaF initial User Mode splash screen as illustrated below.

![Initial Welcome Screen (User Mode)](image)

Initial Welcome Screen (User Mode)

This screen enables users to check VoiceMail and create Recordings, and gives access to the FOP (flash operator panel). PiaF needs to be configured before any of those facilities are operable by users. To start configuring PiaF, it is necessary to switch to Administration Mode. To do this, click on the “Admin” label situated on the bottom left hand corner of the screen. A password dialog screen will appear where the password needs to be entered as per the illustration below:

![Password Dialog](image)

On entering the correct password, you will be presented with the other options where the admin can make other changes if necessary (The default password is admin. Naturally you can change it).
Here you will have 3 extra options:

- **FreePBX Administration** – To manage the PBX through FreePBX
- **Linux Webmin** – This is a system utility that allows you to maintain the virtually the entire system. It is best to restrict this to the System Administrator and Root users only.
- **Menu Configuration** – This determines what the User menu shows.

Click on the FreePBX Administration icon to login to Asterisk Mgmt (FreePBX).
The default username is `admin` and the default password is `admin`. It can be changed using the `passwd` command. This should be done after the system has been configured. For more information on how to use Linux commands, see the [Linux Commands](#).

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

Admin Mode Initial Configuration Screen
Installing Modules and Updates

FreePBX is the GUI that PiaF uses to manage the IP PBX. In most cases, all the modules that you will require have already been installed as default. However, should you need other modules that were not included or update existing modules; you will need to install them from the FreePBX online repository via Module Admin. Updates are available on a regular basis.

From the FreePBX GUI, select Module Admin on the left.

The next screen will then be presented with the list of available FreePBX modules. Some of them may not be installed or enabled.

It is up to you to select whichever modules you require to be installed in FreePBX but all those that are initially required by PiaF would have already been installed by default.

Start selecting all the modules that you want to install or enabled. It is safe to upgrade and install all of the upgradable modules at the same time. You can also install all the modules and remove modules or choose to disable some of the modules that you do not need at a later stage.

If you click on the Check for updates online it will check for updates and change appropriately.
To update a module, click on it to see the options. Choose **Download and Install**.

Click **Process** at the bottom of the list of modules once you have selected all of the modules.

On the confirmation page, click **Confirm**. The module will download, expand, and upgrade automatically.

**Updated phpbiconf**
**Updated music**
**Updated announcement**
**tw_langpacks installed successfully**

You should get a page telling us that all modules were installed successfully. Then click **Return** to go back to the modules list page.
At this stage, click on the orange bar at the top of the page that says 'Apply Configuration Changes'.

At the next prompt, select "Continue with reload" on the window that pops up. This reloads the Asterisk configuration and the orange bar will then go away.

**NOTE:** This process must be performed for any system changes. It is okay to do this while users are using the system.
Quick Startup FreePBX Basics

The Linux and FreePBX software installation has been performed on your Ringdale PBX. While there are other ways to change the configuration of the system, the FreePBX web GUI should be used as the first option to configure your system because FreePBX will overwrite many manual changes when it restarts. Once you have the base system running, you can use other tools to fine tune your system as necessary, but only after reading the forums and gaining a comfortable understanding of what you are doing.

General Settings

Select General Setting and set it up similar to below. Generally the default values are sufficient. For the time being it is recommended to use the settings below. You can find help for most settings by hovering your mouse pointer over the parameter.
IMPORTANT NOTE:

The following parameters are extra information that define the way Asterisk behaves and are also required in FreePBX. Set the fields to the following vital settings:

**Asterisk Outbound Dial command option:**
"r" plays the ring when you dial out. "m" plays music.

**Voicemail:**
Change the Direct Dial to Voicemail message type to something other than default – typically Unavailable.

**Country Indications:** United States

**Allow Anonymous Inbound SIP Calls?** Select Yes. If set to ‘No’, all inbound unidentified SIP calls will not be accepted. Most inbound callers are not identified.

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

**Extensions**

The configuration in this document is directed towards the Ringdale Telephone Deskset, SIP-based and Ethernet-Connected 00-11-0998-0000. You can have soft phones installed in computers or mixture of ATAs (Analog Terminal Adapters) and SIP SoftPhones.

It’s best to avoid the following extension numbers:
- 70-79 - Reserved for calls on hold (Definable)
- 700-799 - Reserved for calls on hold (Definable)
- 7777 - Reserved extension for incoming calls simulation

**Create Extensions**

To create extensions, do the following:
1. Select the FreePBX Setup tab.
2. Click on Extensions.
3. Click on Add Extension.
4. From the drop down selection box, select Generic Sip Device since we are going to create a SIP extension.
5. Click **Submit** (See illustration below).
This shows the basic settings for a SIP Extension.

For more information, see the BASIC Setup.

There are a few fields that you will need to populate. For example:

- **User Extension** is a number, typically 3-5 digits.
- **Display Name**: Reception for example. This is the name that will display on the phone that you call.

**NOTE:** The “secret” must match the Authenticate Password set in the Grandstream phone. Some recommend using the extension number as the secret password, BUT do not do this if your PBX is not behind a firewall and exposed to the Internet.

**NOTE:** `dtmfmode` must be set to `rfc2833` in order for voice mail to work. This must be configured in the IP phone as well.

You can leave the rest of the fields at their default values.

After configuring, click on the **Submit Changes** button and the red bar on top of the screen for the change to take effect.
Enable Voicemail

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

Go back to the Extension Option of FreePBX and click on extension 2001. Scroll down to the Voicemail and directory section and do the following:

- Status: Enabled (use the drop down selection to select it)
- Voicemail password: (to keep it simple you can use the extension number)
- Email address: Enter an email address of the person
- Email attachment: yes
- Play CID: yes
- Play envelope: yes
- Vm context: default

Click Submit when done, and then

Apply Configuration Changes
Ring Groups

A ring group is a group of extensions that will ring when there is an external incoming call. You can even put your Mobile Phone number in the ring group if you want to.

You do not need a ring group. If you don’t require a ring group, you may ignore this section, but it is easiest in the initial setup to configure a ring-group so that all incoming calls ring on all extensions.

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

Ringall: Ring all the available extensions in this group until the call is answered.
Hunt: Take turns ringing each available extension
MemoryHunt: Ring the first extension in the list, then the first and second, then the first second and third extensions in the list, and so on.
Ringall-Prim: Ring all the available extensions in this group until the call is answered. Unless the first extension in the list is busy or on Do Not Disturb, then do not ring the other listed extensions.
Hunt-Prim: Take turns ringing each available extension, unless the first extension listed is busy or on Do Not Disturb, then do not ring the other listed extensions.
MemoryHunt-Prim: Ring the first extension in the list, then the first and second, then the first second and third extensions in the list, and so on. Unless the first extension in the list is busy or on Do Not Disturb, then do not ring the other listed extensions.
FirstAvailable: Ring only the first available extension.
FirstNotOnPhone: Ring only the first extension which is not off hook, ignoring call waiting.

Select the [Ring Group] button on the left side of the screen.
This example used Ring All as the description. You could have different groups. One incoming line might call Ring Group 200, and the other Ring Group 201.

A single digit ring group is not recommended.

You can include a Mobile Phone number in as the last one in the group.

If no one answers the call, Asterisk will go to Voice Mail.

Click on the **Submit Changes** button when done.
Trunks

At this point your PBX should be able to send and receive calls between internal extensions. In order to make an outside call, you will need to add a trunk. A trunk is the telephone service line that you will be using to make an external call on. A trunk can support as many outbound calls as is configured in the Maximum Channels parameter for that trunk.

A trunk can be any of the following:

- a connection to another PBX
- a VOIP service provider (VSP) that you have signed up with
- a POTS line to the PSTN

You can have several trunks if you want to. For example, you might have the following trunks:

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

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

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

You need at least one trunk to make external, PSTN or VOIP calls.

The Ringdale PBX package uses a Zap Trunk, so click on the Add Zap Trunk.
When you first add the ZAP Trunk it will ask you for the Outbound Caller ID and Maximum Channels.

1. Enter the phone number for your POTS line in the Outbound Caller ID field
2. Enter 1 for Maximum Channels
3. Set a dial rule you want for this trunk
4. Select an outbound dial prefix to select this trunk when dialing.
5. Set the Zap Identifier to 1 (the default is g0)

Once the card is configured, you must add a route for Incoming Calls (the catch all incoming route will do) or Asterisk will not answer this line.

**Caller ID (CID) using ZAP Device**

Caller IDs require that you apply to your Telco to have it activated on your line. In some countries this is not activated by default by your Telco. If caller ID is not activated on your line, you will not get CID.

If you have caller ID activated and still don’t get caller ID, then look at the ZAP configuration files:

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

```plaintext
zapata.conf
usecallingpres=yes, callwaitingCaller ID=yes, threewaycalling=yes, useCaller ID=yes, hideCaller ID=no, relaxdtmf=yes

zapata-auto.conf
The following switches may need to be added to the existing ones.
useincommingCaller IDonzaptransfer=yes, adsi=yes, sendCaller IDafter=2
```

After the above are done, restart PiaF: `Amportal restart`
Trunk Outgoing Dial Rules

The most important part of the Trunk setup is the Dial Rules. This is what the trunk is required to send to the VSP to make a successful call to the number you dialed. Trunk Dial Rules allow you to add or strip digits. Trunk Dial Rules are NEVER used to allow or restrict numbers that may be dialed. Call filtering is done in Outbound Routes.

Trunk Dial Rules:

The various patterns you can enter:
- 0 to 9
- X — Refers to any digit between 0 and 9
- N — Refers to any digit between 2 and 9
- Z — any digit that is not zero. (E.g. 1 to 9)
- . — Wildcard. Match any number of anything. Must match *something*.
- + — allows you to add a prefix

In the example above, the Dial Rules permit the following:

XXXXXXX allows any 7 digits for local calls.

NXXXXXXXXX you can dial any digit between 2 & 9 followed by any nine digits, for long distance calls.

Sometimes you might have the Ringdale PBX installed internally to a main PBX in which case you need to add the 9 to get an outside line from the main PBX for example 901144XXXXXXXXX allows calls to the UK.

If you needed a prefix to dial a long distance number, in your Trunk Outgoing Dial Rule, you will need the following pattern: 61+NXXXXXXXXX That pattern tells Asterisk to add 61 in front of the 291234567 before dialling the number via the trunk.

For more info:
Inbound Routes ( Incoming Calls)

When an incoming call from PSTN or VoIP trunk is received, asterisk needs to know where to direct it. It can be directed to a ring group, an extension, Digital Receptionist (IVR) or Queue. For this purpose, Inbound Route needs to be set up. For each trunk, a corresponding inbound route must be created in order to use that route for calls. At least one Inbound Route must be created for PiaF to answer incoming calls, whether from PSTN trunk or SIP calls.

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

Inbound calls can be routed to specific extensions, to a Ring Group, or to an IVR. The easiest configuration is ring all to a Ring Group:

For an Incoming Route that is destined for a Ring Group, all that is required is to give it a name and a destination.

Leave DID Number blank.
Leave CID Number blank.
See the example below for defining an Incoming Route to a specific extension.

If Privacy Manager is set to Yes and Caller ID is not passed, the Privacy Manager will ask the caller for their 10-digit phone number. The caller is given 3 tries. If the caller has call screen enabled, then they will be asked the above and then asked to say their name.

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

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

Select your Ring Group.
Incoming Route to a Specific Extension

Use Tools / Config Edit to change `zapata.conf` and `zapata-channels` from context=from-pstn to context=from-zaptel

Set up the Zap Channel DIDs to the extension used by outside the PBX.

Then set up your Inbound Routes to that DID Number with the check in CID Priority Route.

You can set the CID name prefix.

Set the Destination - Extension.

The system must be restarted for this to work.
ZAP Channel DIDs

To handle inbound calls from a ZAP trunk, you simply enter the Zaptel Channel number in the zaptel channel field. This will determine which zap call be directed to where. If there are 4 FXO modules in a TDM400 card, the zap channels will be 1, 2, 3 and 4. Each FXO is a channel (this can be confirmed in zapata-channels.conf file) and each channel can have a DID assigned to it. If you want to direct channels to specific extensions, you need to define your DID (incoming phone number) for each ZAP channel.

This is done through the FreePBX ZAP Channel DIDs option.

![Edit Zap Channel: 1](image)

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

Time Conditions

You can create various time conditions and use these time conditions in conjunction with your Inbound Route to individualize each of the incoming trunk’s behavior.

You may create several time conditions and give each of the time condition a Short Name to identify it.

These time conditions can then be assigned to each individual Inbound Route if you choose to do so or they can be nested.

NOTE: Day/Night Controls override Time Conditions.
Outbound Routes

Outbound Routes direct calls through predefined routes to the trunks. This is where call filtering is done.

When you dial out, Asterisk will do the following:

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

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

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

Dial Patterns act like a filter for matching numbers dialed with trunks.

In this example the FreePBX trunk feeds to another PBX.

The various patterns you can enter are similar to Asterisk's definition of them:

- 0 to 9
- X — Refers to any digit between 0 and 9
- N — Refers to any digit between 2 and 9
- Z — any digit that is not zero. (E.g. 1 to 9)
- . . . . — Wildcard. Match any number of anything. Must match *something*.
- [Various] — Match only one character that matches any of the one in the square brackets. (E.g. [02-68*#] would match 0, any number between 2 and 6 inclusive, 8, * and #. Or, another way of saying this would be 'Match * or #, or a number that isn't 1, 7 or 9') –Do not use this notation in the Trunk Dial Rule. It is only for Outgoing Route Dial Patterns.
- | — This lets you use a '9 to dial out' by matching anything before the line, but not sending it to the trunk.
IP Phone Configuration

These are the basic required steps to configure the GrandStream Telephone Deskset, SIP-based and Ethernet-Connected 00-11-0998-0000.

First you must set an Administrator password for the IP phone. Using your browser, connect to the IP phone by entering its IP address in the browser.

![Grandstream Device Configuration]

The default password is admin. You should set all of the IP phones to a strong password. This is done under the Advanced Settings tab.

![Grandstream Device Configuration]

You can also set a password for the basic settings that the end user can change. This is done under the basic settings tab.

![Grandstream Device Configuration]

If you log in with the End User Password, you will only have access to the Status, Basic Settings, and the Ext1 and Ext 2 tabs.

After making changes, at the bottom of the page, click on the Update button.

![Update]

When you update it will offer to reboot.

![Reboot]
Basic Settings

Next you will select either dynamic or static IP addressing and fill in the blanks as appropriate for your network.

You will typically also set the End User’s time settings.
Advanced Settings & Account Setup

Under the Advanced Settings tab, aside from the Admin password, there are not any settings that you need to change. Click on the Account 1 tab to select that menu.

<table>
<thead>
<tr>
<th>Account Active:</th>
<th>☒ No ☐ Yes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Account Name:</td>
<td>(e.g., MyCompany)</td>
</tr>
<tr>
<td>SIP Server:</td>
<td>(e.g., sip:mycompany.com, or IP address)</td>
</tr>
<tr>
<td>Outbound Proxy:</td>
<td>(e.g., proxy.myprovider.com, or IP address)</td>
</tr>
<tr>
<td>SIP User ID:</td>
<td>(the user part of an SIP address)</td>
</tr>
<tr>
<td>Authenticate ID:</td>
<td>(can be same or different from SIP User ID)</td>
</tr>
<tr>
<td>Authenticate Password:</td>
<td>(not displayed for security protection)</td>
</tr>
<tr>
<td>Name:</td>
<td>(optional, e.g., John Doe)</td>
</tr>
</tbody>
</table>

Under Account Setup you must set the following parameters:

Set the **Account Active**.

**Account Name** is not required.

**SIP Server** is the IP Address of the Ringdale PBX Server.

The **SIP User ID** is the extension number of the phone.

**Authenticate ID** is set to the SIP User ID. It must match the User Extension defined in FreePBX.

The **Authenticate Password** should be set to the strong password that matches the **Secret** set in the FreePBX Extension setup. **NOTE:** Failure to select a strong password could leave your system prone to hackers and thieves wishing to use your system for long distance calls.

The **Name** if used is shown on the front of the phone.

Make certain that there is a check in **Send DTMF: “in-audio” and “via RTP (RFC2833)”**.

Set the **Voice Mail UserID** to the same as you configured in the FreePBX Administration – Extension setup.

You can configure multiple accounts in the Grandstream phones to match FreePBX Extensions.

Under the Ext tabs you can configure speed-dial functions.

To save the changes, at the bottom of the page click on **Update** and then **Reboot** for them to take effect.
Grandstream Parameters that must match the PBX

You must configure the password for voicemail, the authenticate IDs. The following parameters must match for a GrandStream phone to communicate with the Free PBX Server, as follows:

1. **PBX Configuration Browser**
   
   Using your browse access the FreePBX Administration, then click on the Setup tab, select Extensions and then select or add the Extension number.

   There is a sub-section headed “This device uses SIP technology”. In the secret field option enter a strong password that will match the Authenticate Password set in the Grandstream phone.

   **NOTE:** Failure to select a strong password could leave your system prone to hackers and thieves wishing to use your system for long distance calls.

   Under Voicemail & Directory, the Voicemail Password can be set to the same as the extension unless you have a need for higher security.

   At the bottom of this page click on Submit, Apply Configuration Changes and then Continue with reload to apply any changes / updates.

2. **GrandStream Device Configuration Browser** under the Account 1 tab:

   The SIP Server must match the Ringdale PBX.

   The SIP User ID and Authenticate ID field are set to the extension number of the phone. User ID is the user part of the SIP address of the phone and this is usually the information displayed as Caller ID on the LCD. e.g., typically it is a phone number or extension number or a user’s name. Authentication ID is an ID used strictly for authentication purpose when the phone attempts to contact the SIP server. This may or may not be the same as User ID.

   The Authenticate Password should be set to the strong password that matches the Secret set in the FreePBX Extension setup. **NOTE:** Failure to select a strong password could leave your system prone to hackers and thieves wishing to use your system for long distance calls.

   Make certain that there is a check in Send DTMF: “in-audio” and ‘via RTP (RFC2833)”.

   Set the Voice Mail UserID to the same as you configured in the FreePBX Administration – Extension setup.

   To save the changes, at the bottom of the page click on Update and then Reboot for them to take effect.

3. **GrandStream Phone Configuration**

   If a browser is not handy, you can configure the Grandstream SIP Password at the phone itself. From the phone hardware, select the round ‘setup’ button (in the middle of the four arrows on the right side of the phone), scroll down and select Config, SIP, and SIP Password. Enter the same password for the Secret in the FreePBX Extension setup as used in step (1) above and then OK. **Note** that whenever you go back to this SIP Password selection there is no password displayed (not even a * to indicate an entry). If the SIP Password is not correct then the phone will not connect to the PBX Server.
How to Change Personal Voice Mail Greetings

The default Comedian Voicemail greeting functions quite well, but if you wish to provide more options or a more personal message, you can. Ensure that you have changed your Direct Dial to Voicemail message type, in General Setting, from Default to something else e.g. Unavailable. Otherwise when you use follow-me and the like, your custom recorded message will not be played back. The Asterisk default message will be played back instead. Here are the steps to create a personal message:

1. Use your existing extension and dial *97
2. You will be asked for your password (if you have entered one)
3. When the Voicemail IVR starts, press 0
4. You will then be given the choice what type of message you want to record. 1 is for your Unavailable message and 2 is for your Busy message. Choose the appropriate message you want to record
5. Record your message “Thank you for calling, I am either busy elsewhere or on the phone. Please leave me your message and I will call you back as soon as I can”.
6. Press 2 to review your message. Press 1 to save. Press 3 to re-record.
Stopping and Starting the System

If you press the power button on the front of the Ringdale PBX for about 10 seconds, a clean power-down will occur. If you wish to do this manually, the Asterisk PBX server must be stopped first and the MySQL database must be properly shutdown or the configuration will become corrupted. The first step is stopping Asterisk; the second is issuing the shutdown command for Linux. During the Linux shutdown, it will stop the MySQL database properly.

For more information on how to use Linux commands, see the Linux Commands.

Stopping Asterisk PBX

Click on the Tools tab in the FreePBX Administration.

Click on the Asterisk CLI

Issue the following command at the Asterisk CLI.

```
stop gracefully
```

Another option is to stop the Asterisk PBX at the Linux command line:

```
root@pbx:~ $ amportal stop (amportal is the original name for Asterisk)
```

Shutting Down the Linux server

After Asterisk is gracefully stopped, you can shutdown the PC using the following command at the root prompt:

```
root@pbx:~ $ shutdown -h now (the -h means halt)
```

Rebooting the Linux server

If you want to reboot the system instead of shutting down, first stop Asterisk PBX using one of the methods explained previously then use the following command:

```
root@pbx:~ $ shutdown -r now (the -r means restart)
```

The server will shutdown and then reboot.
Restarting Asterisk without shutting down

Sometimes you want to stop Asterisk and restart it without shutting down Linux, here's how:

At the Asterisk command line prompt:

**Asterisk CLI**

```plaintext
CLI> restart now
```

Or at the Linux command prompt:

```plaintext
root@pbx:~ $ amportal restart
```

Or if you want to shut it down, and then start it up later:

```plaintext
root@pbx:~ $ amportal stop
root@pbx:~ $ amportal start
```

**NOTE:** This does not affect currently connected calls.

For more information on how to use Linux commands, see the [Linux Commands](#).
FreePBX Menu Tab Bar

Admin

Under the **Admin** tab you will have the **Setup** and **Tools** tab. Both are covered extensively in the following pages.

Reports

Click on the **Reports** tab brings up the following report:

There is also an exportable graph.
Panel

Clicking on the Panel tab brings up the Flash Operator Panel. This is also accessible from the PBX in a Flash – Main Menu.

When you are in the PIAF Flash Operator Panel there is a lock icon: Open Security Code Input Box. When you click on it, it says “Please enter the Security Code”. The default password is passw0rd.

<table>
<thead>
<tr>
<th>The following information are displayed on FOP</th>
<th>Functions you can perform on FOP</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Which extensions are busy, ringing or available</td>
<td>• Hang-up a channel</td>
</tr>
<tr>
<td>• Who is talking and to whom (CLID, context, priority)</td>
<td>• Using drag-&amp;-drop to transfer a call</td>
</tr>
<tr>
<td>• SIP and IAX registration status and reachability</td>
<td>• Initiate calls by drag-&amp;-drop</td>
</tr>
<tr>
<td>• MeetMe room status (number of participants)</td>
<td>• Barge in on a call using drag-&amp;-drop</td>
</tr>
<tr>
<td>• Queue status (number of users waiting)</td>
<td>• Set the caller id when transferring or originating a call</td>
</tr>
<tr>
<td>• Message Waiting Indicator and count</td>
<td>• Automatically pop up web page with customer details</td>
</tr>
<tr>
<td>• Parked channels</td>
<td>• Click-to-Dial from a web page</td>
</tr>
<tr>
<td>• Logged in Agents</td>
<td>• Mute/Unmute meet-me participants</td>
</tr>
</tbody>
</table>
Recordings

Clicking on the Recordings tab will take you to the Voicemail & Recordings interface.

Login

Use your Voicemail Mailbox and Password
This is the same password used for the phone
For password maintenance or assistance, contact your Phone System Administrator.

This is the same password used for your phone and voicemail.

These features are self-explanatory.
Help

Clicking on the Help tab will take you to the FreePBX website.
http://www.freepbx.org/freepbx-help-system?freepbx_version=2.6.0.0
FreePBX - Setup Menus

Notices, FreePBX & System Statistics, Uptime, and Server Status
Check for updates online & if available update modules.
Add, remove and define the extension settings like name, password, and voice mail.
Enable or disable Feature Codes, such as Do Not Disturb
General is for basic setup information
Configure dial patterns to filter calls and point to trunks.
Add the trunk and configure dial rules for adding or stripping digits.
Add Administrators and passwords.

Either catch-all or specific DIDs to specific extensions
Assign Zap channel ports to DIDs.
Custom announcements can be added.
Add phone numbers that you do not wish to hear from
Specify a source for resolving numeric caller IDs of incoming calls
Send night calls to voice mail using a feature code.
Forward unanswered calls to other extensions.
Configure Interactive Voice Response
Queue Priority allows you to set a caller’s priority in a queue.
You can dial into the outgoing queue when the line is busy
A group of extensions that an incoming route can ring
Send night calls to voice mail at specified times.
Used with Time Conditions

Disconnects caller and calls back, connecting to selected destination
Add a conference with name, ext. number, PINs, etc.
DISA allows outside callers access to the PBX to dial out.
Support for other languages.
For adding feature codes
For adding destinations to be used by other FreePBX modules
Rock ‘n’ Roll.
PIN Sets are used to restrict Feature Code changing.
For specific phones that are capable of this.
Where to put calls on hold and what to do when orphaned.
Use your phone or upload waves for custom messages
For sending announcements to groups of voice mail boxes.
Admin

FreePBX System Status


FreePBX System Status

FreePBX Notices

System Statistics

- Processor:
  - Load Average: 0.00
  - CPU: 0%

- Memory:
  - Total Memory: 19%
  - Swap: 0%

- Disks:
  - /dev/sda1: 72%
  - /dev/sda3: 10%
  - /dev/sda4: 0%

- Networks:
  - eth0 receive: 0.32 Kbps
  - eth0 transmit: 0.83 Kbps

Server Status

- Asterisk: OK
- Op Panel: OK
- MySQL: OK
- Web Server: OK
- 33H Server: OK
Module Admin

In the Module Administration panel you can check for update, enable, disable, and uninstall modules. It is intuitive and easy to use. Upgrade modules rarely cause problems. Some new function do not always work initially, but do not usually cause problems.

If you have requested update notification, you will get an email from Asterisk VoIP PBX with the subject: FreePBX: New Online Updates Available.

### Module Administration

<table>
<thead>
<tr>
<th>Module</th>
<th>Version</th>
<th>Publisher</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Core</td>
<td>2.6.0RC1.1</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
<tr>
<td>Feature Code Admin</td>
<td>2.6.0.0</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
<tr>
<td>FreePBX ARI Framework</td>
<td>2.6.0.2</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
<tr>
<td>FreePBX FOP Framework</td>
<td>2.6.0.2</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
<tr>
<td>FreePBX Framework</td>
<td>2.6.0RC1.1</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
<tr>
<td>FreePBX Localization Updates</td>
<td>2.6.0.2</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
<tr>
<td>System Dashboard</td>
<td>2.6.0.1</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
<tr>
<td>Voicemail</td>
<td>2.6.0.2</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

### CID & Number Management

<table>
<thead>
<tr>
<th>Module</th>
<th>Version</th>
<th>Publisher</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phonebook Directory</td>
<td>2.6.0.1</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
<tr>
<td>Speed Dial Functions</td>
<td>2.6.0.0</td>
<td>FreePBX</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

### Inbound Call Control

The first thing to do in Module Administration is click on the **Check for updates online**.

If there are, they will be highlighted in red.

<table>
<thead>
<tr>
<th>Module</th>
<th>Version</th>
<th>Publisher</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Built-in</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Core</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Feature Code Admin</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>FreePBX ARI Framework</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>FreePBX FOP Framework</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>FreePBX Framework</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>FreePBX Localization Updates</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>System Dashboard</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Voicemail</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Click the **Show only upgradable** to condense the list.

**Manage local modules**

<table>
<thead>
<tr>
<th>Module</th>
<th>Version</th>
<th>Publisher</th>
<th>Action</th>
<th>Description</th>
<th>Changelog</th>
</tr>
</thead>
<tbody>
<tr>
<td>Core</td>
<td>2.6.0RC1.1</td>
<td>FreePBX</td>
<td>![ ]*</td>
<td>![ ]* Download and Upgrade to 2.6.0RC2.1</td>
<td>![ ]*</td>
</tr>
<tr>
<td>FreePBX EOP Framework</td>
<td>2.6.0.2</td>
<td>FreePBX</td>
<td>![ ]*</td>
<td>![ ]* Download and Upgrade to 2.6.0.3</td>
<td>![ ]*</td>
</tr>
<tr>
<td>FreePBX Framework</td>
<td>2.6.0.RC1.1</td>
<td>FreePBX</td>
<td>![ ]*</td>
<td>![ ]* Download and Upgrade to 2.6.0.RC2.1</td>
<td>![ ]*</td>
</tr>
</tbody>
</table>

You can click on the line for an available upgrade, and then individually choose the **Download and Upgrade** option and then click on **Download all** to download the upgrades individually or select **Upgrade all** to open them all up. You can still change any of them to **No Action**. Occasionally separate upgrades are required, and a message will inform you of this.

After selecting the modules to upgrade, click on the **Process** button.

**Please confirm the following actions:**

- Core 2.6.0RC1.1 will be upgraded to online version 2.6.0RC2.1
- FreePBX Framework 2.6.0 RC1.1 will be upgraded to online version 2.6.0 RC2.1
- FreePBX EOP Framework 2.6.0.2 will be upgraded to online version 2.6.0.3

Click on the **Confirm** button.
An update window will show the progress and tell whether the module installed successfully.

Use the slide bar to go to the bottom and click on **Return**.

Click on the **Apply Configuration Changes** button.

You may want to use the FreePBX Flash Operator Panel to make certain no phones are active, then click on **Continue with reload**.

In some cases, you will have to do an Amportal Restart from the command line or restart the whole PBX.

**Extended Repository** – The extended repository contains some Third Party modules. These modules are believed to work with FreePBX, but they are developed by third parties in conjunction with optional PBX components, and they are not directly supported by the core FreePBX team. They may not receive the same level of responsiveness to issues as the main code base does.
Basic
This section contains more information on the Basic Setup menus.

Extensions
Click the Add Extension on the right of the Extension menu.

NOTE: It’s best to avoid the following extension numbers:
- 70-79 - Reserved for calls on hold (Definable)
- 700-799 - Reserved for calls on hold (Definable)
- 7777 - Reserved extension for incoming calls simulation

The Add an Extension menu comes up.

Add an Extension
Select your Device then click the Submit button.

NOTE: If you are installing the Ringdale Telephone Deskset, 00-11-0998-0000, it is SIP-based, so you will select Generic SIP Device.

This will bring up the Add SIP Extension dialogue to set it up as follows:

Add Extension
- User Extension - The extension number to dial to reach this user.
- Display Name - The caller id name for calls from this user will be set to this name. Only enter the name, NOT the number. This is the name that will display on the phone that you call.
CID Num Alias - The CID Number to use for internal calls, if different from the extension number. This is used to masquerade as a different user. A common example is a team of support people who would like their internal caller ID to display the general support number (a ring group or queue). There will be no effect on external calls.

SIP Alias - If you want to support direct sip dialing of users internally or through anonymous sip calls, you can supply a friendly name that can be used in addition to the user’s extension to call them.

Extension Options

Outbound CID - Overrides the caller id when dialing out a trunk. Any setting here will override the common outbound Caller id set in the Trunks admin.

Format: "caller name" <#######>

Leave this field blank to disable the outbound Caller ID feature for this user.

Ring Time - Number of seconds to ring prior to going to voicemail. Default will use the value set in the General Tab. If no voicemail is configured this will be ignored. Options are Default 1 - 120.

Call Waiting - Set the initial/current Call Waiting state for this user’s extension to Enable or Disable.

Call Screening - Call Screening requires external callers to say their name, which will be played back to the user and allow the user to accept or reject the call. Options are Disable, Screen Caller: No Memory, Screen Caller: Memory.

- Screen Caller: Memory only verifies a caller for their Caller ID once.
- Screen Caller: No Memory always requires a caller to say their name. Either mode will always announce the caller based on the last introduction saved with that Caller ID. If any user on the system uses the memory option, when that user is called, the caller will be required to reintroduce themselves and all users on the system will have that new introduction associated with the caller’s Caller ID.

Emergency CID - This caller id will always be set when dialing out an Outbound Route flagged as Emergency. The Emergency CID overrides all other caller id settings.

Assigned DID/CID

DID Description - A description for this DID, such as “Fax”

Add Inbound DID - A direct DID that is associated with this extension. The DID should be in the same format as provided by the provider (e.g. full number, 4 digits for 10x4, etc). Format should be: X0000000XX

Add Inbound CID - (Optional) Add a CID for more specific DID + CID routing. A DID must be specified in the above Add DID box. In addition to standard dial sequences, you can also put Private, Blocked, Unknown, Restricted, Anonymous andUnavailable in order to catch these special cases if the Telco transmits them.
Device Options
This device uses sip technology.

<table>
<thead>
<tr>
<th>Device Options</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>secret</strong></td>
</tr>
<tr>
<td><strong>dtmfmode</strong></td>
</tr>
<tr>
<td><strong>canreinvite</strong></td>
</tr>
<tr>
<td><strong>context</strong></td>
</tr>
<tr>
<td><strong>host</strong></td>
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<td><strong>type</strong></td>
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<td><strong>nat</strong></td>
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<td><strong>port</strong></td>
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<tr>
<td><strong>qualify</strong></td>
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<tr>
<td><strong>callgroup</strong></td>
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<tr>
<td><strong>pickupgroup</strong></td>
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<tr>
<td><strong>disallow</strong></td>
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<tr>
<td><strong>allow</strong></td>
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<tr>
<td><strong>dia</strong></td>
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<tr>
<td><strong>accountcode</strong></td>
</tr>
<tr>
<td><strong>mailbox</strong></td>
</tr>
<tr>
<td><strong>deny</strong></td>
</tr>
<tr>
<td><strong>permit</strong></td>
</tr>
</tbody>
</table>

**NOTE:**
In the initial setup, you only configure the Secret and dtmfmode.

**Secret** – This must match the Authenticate Password set in the Grandstream phone.

**NOTE:** Failure to select a strong password could leave your system prone to hackers and thieves wishing to use your system for long distance calls.

**dtmfmode** - rfc2833 – DTMF stands for Dual-tone multi-frequency. It is used for telecommunication signaling over analog telephone lines in the voice-frequency band between telephone handsets and other communications devices and the switching center. You should leave this at rfc2833.

Grandstream Settings (under the ACCOUNT setup) that also affect it:
- Disable in-call DTMF display: set to No
- DTMF Payload Type: 101
- Send DTMF: x in-audio  x via RTP (RFC2833) via SIP INFO
- context – A context is just a collection of extensions. In Asterisk, outgoing numbers are divided in groups called contexts in order to separate/define different needs for different user types. For example, a context for local calls, another for within the city, and another for international calls and so on. See http://www.voip-info.org/wiki/view/Asterisk+Dialplan+Introduction
- host – dynamic. This sets dynamic IP for the host. You may also define this as a static IP
- type – friend means the user can place or receive calls. For INBOUND calls only, use ‘peer’ as type. For outbound calls only use ‘user’ as type.
- nat – should be ‘no’ for security. Set this to ‘yes’ only if you want to be able to use your phone behind a NAT firewall separate from the LAN that your PBX Server is on, AND you have strong passwords set.
- port – 5060 is the default port used. The default installation of FreePBX is configured to use UDP port 5060 as the SIP signaling port and UDP ports 10001-20000 as the RTP Media ports. If your SIP phone are behind a NAT firewall, all these ports must be forwarded to your FreePBX System. How to do this varies widely depending on the firewall or equipment that you are using. It is commonly referred to as Port Forwarding or maybe Destination NAT (DNAT). However it is referred, if we assume in this example that your FreePBX system has an internal IP address of 192.168.1.100 then you will want:

```
UDP/5060 -> Forward to 192.168.1.100
UDP/10001-20000 -> Forward to 192.168.1.100
```

For info: http://www.freepbx.org/support/documentation/howtos/howto-setup-a-remote-sip-extension
**qualify**=xxx|no|yes where XXX is the number of milliseconds used. If yes the default timeout is used, 2 seconds. If you turn on qualify in the configuration of a SIP device in sip.conf, Asterisk will send a SIP OPTIONS command regularly to check that the device is still online. If the device does not answer within the configured (or default) period (in ms) Asterisk considers the device off-line for future calls. This status can be checked by the SIPPEER function, and inversely this function will only provide status information for peers which have qualify=yes.

This feature may also be used to keep a UDP session open to a device that is located behind a network address translator (NAT). By sending the OPTIONS request, the UDP port binding in the NAT (on the outside address of the NAT/firewall device) is maintained by sending traffic through it. If the binding were to expire, there would be no way for Asterisk to initiate a call to the SIP device. This can be used in conjunction with the nat=yes setting.

**callgroup** – xx – where xx (under 63) is the group that this extension is in. This is not the same as a Ring Group. Another extension must have this callgroup in its pickupgroup, in order to pick this call up, using *8.

**pickupgroup** – xx – where xx (under 63) is the group that this extension can pick up calls for. This is not the same as a Ring Group.

**Call groups and pickup groups**

These group are used to allow picking up remotely a ringing phone through *8 (by default, it is denied). The call group is what the extension belongs in; the pickup group is which callgroups the extension can remotely pick up.

The basic functionality is this:

1) **Asterisk General Call Pickup** under the Setup tabs General menu must be enabled. It is usually set to *8.

2) Extension As callgroup is to one or several callgroups.

3) Extension B (within hearing distance) has one of those callgroups in its pickupgroup.

4) A call comes in to Extension A and the intended user does not pick it up.

5) Extension B may pickup the incoming call by calling *8 on the phone.

**disallow** – Disallow any codec you do not want to use. Common setting for this if you want to make sure a device only uses the codec set in the allow section is “all” (without the quotes).

**allow** – Allow a codec of your choice. For example use “g729” or “GSM” (without the quotes); only one codec can be set here. This is useful if used with the disallow option set to “all” and you set the definitive codec you want to use on allow, guaranteeing that you will use that codec.

**dial** – SIP/xxx where xxx is the extension

**accountcode** – This field, if defined, is used to populate the 'accountcode' field of the CDR.

**mailbox** – xxx@default – where xxx is the extension. Normally you would use 'your extension@device, but if you use two extensions for example, if you want to have the other extension's phone's light or dial tone indicate when a different box has voicemail you can set it to 'extension vm@device. For example if extension 1002 wants to know when 1001 has voice mail then set this to 1001@device on 1002's mailbox setting.
deny 0.0.0.0/0.0.0.0
permit 0.0.0.0/0.0.0.0

Syntax
  deny= <ipaddress>/<network mask>
  permit= <ipaddress>/<network mask>

Order Matters! - The last matching rule is the one used. If no rule matches, then the connection is permitted.

Examples:
deny=192.168.40.38/255.255.255.255 - Denies traffic from this IP address
permit=192.168.40.0/255.255.255.0 - Allows traffic from this network
deny=0.0.0.0/0.0.0.0 Don’t deny from anywhere.
permit=216.207.245.47/255.255.255.255 Deny every address except this one.

You may have multiple rules for masking traffic. Rules are processed from the first to the last.

Dictation Services

Dictation Service - Can be Enabled or Disabled.
Dictation Format – Can be Ogg Vorbis, GSM, or WAV.
Email Address - The email address that completed dictations are sent to.

Language
Language Code - This will cause all messages and voice prompts to use the selected language if installed.

Recording Options
Record Incoming - Record all inbound calls received at this extension. Options are On Demand, Always, or Never.
Record Outgoing - Record all outbound calls received at this extension. Options are On Demand, Always, or Never.

Voicemail & Directory
Status – Can be set to Enabled or Disabled.
Voicemail Password - This is the password used to access the voicemail system. This password can only contain numbers. A user can change the password you enter here after logging into the voicemail system (*98) with a phone.
Email Address - The email address that voicemails are sent to.
Pager Email Address - Pager/mobile email address that short voicemail notifications are sent to.
Email Attachment - Option to attach voicemails to email. This option can be set to yes or no.
Play CID - Read back caller’s telephone number prior to playing the incoming message, and just after announcing the date and time the message was left. This option can be set to yes or no.
Play Envelope - Envelope controls whether or not the voicemail system will play the message envelope (date/time) before playing the voicemail message. This setting does not affect the operation of the envelope option in the advanced voicemail menu. This option can be set to yes or no.
Delete Voicemail - If set to "yes" the message will be deleted from the voice mailbox (after having been emailed). This allows a user to receive their voicemail via email alone, rather than having the voicemail able to be retrieved from the Web interface or the Extension handset. This option can be set to yes or no.

CAUTION: If this is set to “Yes” then you must have Email Attachment set to Yes, otherwise your messages will be lost.

VM Options - Separate options with pipe ( | ) i.e.: review=yes|maxmessage=60

VM Context - This is the Voicemail Context which is normally set to default. Do not change unless you understand the implications.

VmX Locater

VmX Locater™ - Enable/Disable the VmX Locater feature for this user. When enabled all settings are controlled by the user in the User Portal (ARI). Disabling will not delete any existing user settings but will disable access to the feature.

Use When: unavailable and/or busy - Check both to use at all times.

Voicemail Instructions: Standard voicemail prompts. Uncheck to just play a beep after your personal voicemail greeting.

NOTE: If you uncheck this, then you must create your own personal voicemail greeting. See the How to Change Voice Mail Greetings instructions below for this.

Menu options below are available during your personal voicemail greeting playback.

Press 0: If Go To Operator is checked, pressing 0 during your personal voicemail greeting goes to the Operator. Uncheck to enter another destination here. This feature can be used while still disabling VmX to allow an alternative Operator extension without requiring the VmX feature for the user.

NOTE: Operator Extension is defined in the Company Directory section of the General settings under the Setup tab.

Press 1: The remaining options can have internal extensions, ring groups, queues and external numbers that may be rung. It is often used to include your cell phone. You should run a test to make sure that the number is functional any time a change is made so you don't leave a caller stranded or receiving invalid number messages.

Press 2: Use any extensions, ringgroups, queues or external numbers. If FollowMe is configured there will be a checkbox for Send to Follow-Me.

Remember to re-record your personal voicemail greeting and include instructions.

Run a test to make sure that the number is functional.

At the bottom of this page click on Submit, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Feature Codes

Click on the Feature Codes button to bring up the Feature Codes Admin. For each of the features, you must check Use Default or not, and select the Feature Status of enabled or disabled. You can also change the code used to implement that feature.

Feature Code Admin

<table>
<thead>
<tr>
<th>Feature</th>
<th>Use Default?</th>
<th>Feature Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Blacklist</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Blacklist a number</td>
<td>*30</td>
<td>Enabled</td>
</tr>
<tr>
<td>Blacklist the last caller</td>
<td>*32</td>
<td>Enabled</td>
</tr>
<tr>
<td>Remove a number from the blacklist</td>
<td>*31</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

For compatibility with other FreePBX systems and for module integrity it is generally best to use the default code for features, but if you must change the code dialed to implement a feature you must remove the check from Use Default, then make the change. If you decide to go back to the default, put the check back in.

If you want to disable a feature, just click the down arrow to select Disabled. After doing so, if you use the Print Extensions under Tools, this feature will not print.

Blacklist

*30  Blacklist a number
*31  Remove a number from the blacklist
*32  Blacklist the last caller

Call Forward

*52  Call Forward No Answer/Unavailable Activate
*53  Call Forward No Answer/Unavailable Deactivate
*72  Call Forward All Activate
*73  Call Forward All Deactivate
*74  Call Forward All Prompting Deactivate
*90  Call Forward Busy Activate
*91  Call Forward Busy Deactivate
*92  Call Forward Busy Prompting Deactivate

Call Waiting

*70  Call Waiting - Activate
*71  Call Waiting - Deactivate
Core

*8 Asterisk General Call Pickup
555 ChanSpy
666 Dial System FAX
** Directed Call Pickup
*2 In-Call Asterisk Attended Transfer
### In-Call Asterisk Blind Transfer
** In-Call Asterisk Disconnect Code
*1 In-Call Asterisk Toggle Call Recording
7777 Simulate Incoming Call
*12 User Logoff
*11 User Logon
888 ZapBarge

Day/Night Mode

*280 Day/Night Control Toggle
*281 Set to Night
*282 Set to Day

Dictation

*34 Perform dictation
*35 Email completed dictation

Do-Not-Disturb (DND)

*78 DND Activate
*79 DND Deactivate
*76 DND Toggle

Gabcast

*422 Connect to Gabcast

Info Services

*69 Call Trace
# Directory
*43 Echo Test
*65 Speak Your Exten Number
*60 Speaking Clock

Paging and Intercom

*54 User Intercom Allow
*55 User Intercom Disallow
*80 Intercom prefix
Phonebook Directory

411  Phonebook dial-by-name directory

Queues

*45  Queue Toggle

Recordings

*99  Check Recording
*77  Save Recording

Speed Dial Functions

*0  Speeddial prefix
*75  Set user speed dial

Voicemail

*97  My Voicemail
*98  Dial Voicemail

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
General Settings

Dialing Options

**Asterisk Dial command options:**
- t: Allow the called user to transfer the call by hitting #
- T: Allow the calling user to transfer the call by hitting #
- r: Generate a ringing tone for the calling party
- w: Allow the called user to start recording after pressing *1 (Asterisk v1.2)
- W: Allow the calling user to start recording after pressing *1 (Asterisk v1.2)

**Asterisk Outbound Dial command options:**
- t: Allow the called user to transfer the call by hitting #
- T: Allow the calling user to transfer the call by hitting #
- w: Allow the called user to start recording after pressing *1 (Asterisk v1.2)
- W: Allow the calling user to start recording after pressing *1 (Asterisk v1.2)
- r: You SHOULD NOT use this option on outbound trunks

Call Recording

**Extension Recording Override:** This will override the recording settings of all extensions/users. If enabled, the system will ignore all Record Always settings of a user and will not turn on recording. This does not affect On Demand recording controlled by the dial options 'w' and 'W' above. It does not affect other recording settings in modules such as Queues and Conferences. If you don't use recordings, setting this is beneficial to system performance as it removes the check that is otherwise done on every single call. Enabled or Disabled

**Call recording format:** Pick the format in which to save recorded calls: WAV, wav, ulaw, alaw, sln, gsm, or g729.

**Recording Location:** Override the default location where asterisk will store call recordings. Include the trailing /. Be sure to set proper permissions on the directory for the asterisk user.

**Run after record:** An optional script to be run after the call is hung up. You can include channel and MixMon variables like ${CALLFILENAME}, ${MIXMON_FORMAT} and ${MIXMON_DIR}. To ensure that you variables are properly escaped, use the following notation: ^{MY_VAR}

Voicemail

**Ringtime Default:** Default number of seconds to ring phones before sending callers to voicemail. This can be set per extension/user and will have no effect on phones that do not have voicemail.

**Direct Dial Voicemail Prefix:** Prefix used to dial directly to someone's voicemail. Caution should be taken in choosing this prefix to avoid conflicts with feature codes.

**Direct Dial to Voicemail message type:** Default message type to use when dialing direct to an extensions voicemail. Options are Default, Unavailable, Busy, or No Message.

**Optional Voicemail Recording Gain:** Use the specified amount of gain when recording the voicemail message. The units are whole-number decibels (dB).
Do Not Play "please leave message after tone" to caller - Check this to remove the default message "Please leave your message after the tone. When done, hang-up, or press the pound key." That is played after the voicemail greeting (the $ option). This applies globally to all voice mail boxes.

Voicemail VmX Locator

Default Context & Pri: Default to use if only a number/extension are provided.

Timeout/#-press default: This is the default location that a caller will be sent if they don't press any key (timeout) or press # which is interpreted as a timeout. Set context to 'dovm' to go to voicemail (default).

Loop Exceed default: This is the default location that a caller will be sent if they press an invalid options too many times, as defined by the Maximum Loops count. Set context to 'dovm' to go to voicemail (default).

Timeout VM Msg: If this destination is voicemail, select whether or not to play the standard voicemail instructions or just beep. Std Instructions or Beep Only

Max Loop VM Msg: If this destination is voicemail, select whether or not to play the standard voicemail instructions or just beep. Std Instructions or Beep Only

Direct VM Option: If a user defined option is to go to voicemail (using the 'dovm' extension) this is the default option if not specified by the user's settings. Std Instructions or Beep Only

Msg Timeout: Time to wait after message has played to timeout and/or repeat the message if no entry pressed. 0-15 seconds

Msg Play: Number of times to play the recorded message if the caller does not press any options and it times out. 1-4 times

Error Re-tries: Number of times to play invalid options and repeat the message upon receiving an undefined option. 1-4 times

Company Directory

Find users in the Company Directory by: The Company Directory allows a caller to spell the user's first name, last name, or both when searching for a user. This will select which of these modes are used: first name, last name, or first or last name.

Announce Extension: Plays a message "Please hold while I transfer you to extension xxx" that lets the caller know what extension to use in the future when connecting from the company directory.

Operator Extension: When users hit '0' in the directory, they are put through to this number. Note that it does NOT need to be an extension; it can be a Ring Group, or even an external number.

Fax Machine

Extension of fax machine for receiving faxes: Select 'system' to have the system receive and email faxes. Selecting 'disabled' will result in incoming calls being answered more quickly.

Email address to have faxes emailed to: Email address used if 'system' has been chosen for the fax extension above.

Email address that faxes appear to come from: Email address that faxes appear to come from if 'system' has been chosen for the fax extension above.

International Settings

Country Indications - Select which country you are in.

24-hour format - Select Yes if you use 24-hour format or No if you are using 12-hour am/pm format.
Security Settings

Allow Anonymous Inbound SIP Calls?

** WARNING ** Setting this to 'yes' will potentially allow ANYBODY to call into your Asterisk server using the SIP protocol. It should only be used if you fully understand the impact of allowing anonymous calls into your server.

Online Updates

Check for Updates - Choosing Yes will result in the system automatically checking for updates nightly. The resulting information will be displayed in the dashboard and will be optionally emailed to the address below if provided. This will transmit your FreePBX and Asterisk version numbers along with a unique but random identifier. This is used to provide proper update information and to track version usage to focus development and maintenance efforts. No private information is transmitted.

Update Email: Email address where online updates will be sent. Leaving blank will result in no updates being sent.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Outbound Routes

This is where you configure call filtering. Click on Add Route to bring up the following settings:

- **Route Name**: Name of this route. Should be used to describe what type of calls this route matches (for example, 'local' or 'longdistance').

- **Route Password**: A route can prompt users for a password before allowing calls to progress. This is useful for restricting calls to international destinations or 1-900 numbers. A numerical password or the path to an Authenticate password file can be used. This field is optional. Leave this field blank to not prompt for password.

- **PIN Set**: (Optional) Select a PIN set to use. If using this option, leave the Route Password field blank.

- **Emergency Dialing**: (Optional) Selecting this option will enforce the use of a device's Emergency CID setting (if set). Select this option if this set of routes is used for emergency dialing (i.e.: 911).

- **Intra Company Route**: (Optional) Selecting this option will treat this route as an intra-company connection, preserving the internal Caller ID information and not use the outbound CID of either the extension or trunk.

- **Music On Hold?** - You can choose which music category to use. For example, choose a type appropriate for a destination country which may have announcements in the appropriate language.

- **Dial Patterns** - A Dial Pattern is a unique set of digits that will select this trunk. Enter one dial pattern per line.
  
  Rules:
  - X matches any digit from 0-9
  - Z matches any digit from 1-9
  - N matches any digit from 2-9
  - [1237-9] matches any digit or letter in the brackets (in this example: 1,2,3,7,8,9)
  - . wildcard, matches one or more characters
  - | separates a dialing prefix from the number (for example, 9|NXXXXXX would match when some dialed "95551234" but would only pass "5551234" to the trunks)

- **Dial patterns wizards** - These options provide a quick way to add outbound dialing rules. Options are: Local 7 digit, Local 7/10 digit, Toll-free, Long-distance, International, Information, Emergency, and Lookup local prefixes. Follow the prompts for each.

- **Lookup local prefixes** - This looks up your local number on www.localcallingguide.com (NA-only), and sets up so you can dial either 7, 10 or 11 digits (5551234, 6135551234, 16135551234) to access this route.

- **Trunk Sequence** - The Trunk Sequence controls the order of trunks that will be used when the above Dial Patterns are matched. For Dial Patterns that match long distance numbers, for example, you’d want to pick the cheapest routes for long distance (i.e., VoIP trunks first) followed by more expensive routes (POTS lines). Usually the only here will be ZAP/g0.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Trunks

A trunk is the telephony service line that you will be using to make an external call on, e.g. the local telephone line that you might have at home can be used as a trunk. A VOIP service provider (VSP) that you have signed up with is also a trunk.

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

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

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

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

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

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

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

This is an example of how to add a ZAP Trunk.

Click on the Add a Trunk

**Add ZAP Trunk**

**General Settings**

**Outbound Caller ID** - Caller ID for calls placed out on this trunk. Setting this option will override all clients' caller IDs for calls placed out this trunk. Leave this field blank to simply pass client caller IDs. Quotes are optional around the caller name, but highly recommended. Format: "caller name" <#######>. You can also use the magic string 'hidden' to hide the CallerID sent out over Digital lines ONLY (E1/T1/J1/BRI/SIP/IAX).

**Never Override CallerID** - Some VoIP providers will drop the call if you try to send an invalid CallerID. An invalid CallerID is defined as one that you don't 'own'. Use this to never send a CallerID that you haven't explicitly specified in this trunk or in the outbound callerid field of an extension/user. You might notice this problem if you discover that Follow-Me or RingGroups with external numbers don't work properly. Checking this box has the effect of disabling 'foreign' callerids from going out this trunk. You must define an Outbound Caller ID on the trunk when checking this.

**Maximum Channels** - Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Leave blank to specify no maximum.

**Disable Trunk** - Check this to disable this trunk in all routes where it is used.

**Monitor Trunk Failures** - If checked, supply the name of a custom AGI Script that will be called to report, log, email or otherwise take some action on trunk failures that are not caused by either NOANSWER or CANCEL.
Outgoing Dial Rules

Dial Rules - A Dial Rule controls how calls will be dialed on this trunk. It can be used to add or remove prefixes. Numbers that don't match any patterns defined here will be dialed as-is. Call filtering is done in Outbound Routes, not Trunks. Note that a pattern without a + or | (to add or remove a prefix) will not make any changes but will create a match. Only the first matched rule will be executed and the remaining rules will not be acted on.

Rules:
- X matches any digit from 0-9
- Z matches any digit from 1-9
- N matches any digit from 2-9
- [1237-9] matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)
- . wildcard, matches one or more characters (not allowed before a | or +)
- | removes a dialing prefix from the number (for example, 613|NXXXXXX would match when some dialed "6135551234" but would only pass "5551234" to the trunk)
- + adds a dialing prefix from the number (for example, 1613+NXXXXXX would match when some dialed "5551234" and would pass "16135551234" to the trunk)

You can also use both + and |, for example: 01+0|1ZXXXXXXX would match "016065551234" and dial it as "0116065551234" Note that the order does not matter, e.g. 0|01+1ZXXXXXXXXX does the same thing.

Dial Rules Wizards - Always dial with prefix is useful for VoIP trunks, where if a number is dialed as "5551234", it can be converted to "16135551234".
- Remove prefix from local numbers is useful for ZAP trunks, where if a local number is dialed as "6135551234"; it can be converted to "555-1234".

Look up numbers for local trunk looks up your local number on www.localcallingguide.com (NA-only), and sets up so you can dial either 7 or 10 digits (regardless of what your PSTN is) on a local trunk (where you have to dial 1+area code for long distance, but only 5551234 (7-digit dialing) or 6135551234 (10-digit dialing) for local calls.

Outbound Dial Prefix - The outbound dialing prefix is used to prefix a dialing string to all outbound calls placed on this trunk. For example, if this trunk is behind another PBX or is a Centrex line, then you would put 9 here to access an outbound line. Another common use is to prefix calls with 'w' on a POTS line that need time to obtain dial tone to avoid eating digits. Most users should leave this option blank.
Outgoing Settings

Zap Identifier (trunk name)

ZAP channels are referenced either by a group number or channel number (which is defined in zapata.conf). The default setting is g0 (group zero).

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Administrators

Add Administrator allows you to grant other users password access to extension ranges and specific FreePBX modules at the expense of exposing FreePBX to the network.

**IMPORTANT WARNING! Read before using this module**

*Do not* use this if the PBX is exposed to the Internet. The default FreePBX uses Apache security. Enabling database mode changes the security mode and could result in having your server hacked and thousands of dollars in phone charges.

**IMPORTANT WARNING! Read before using this module**

If you enable AUTHTYPE after you have added users it will lock you out of FreePBX. You must have NO USERS CREATED before you turn it on.

If you have already tried to add users before changing AUTHTYPE to 'database', delete them, before enabling 'database'.

If you have already tried to add users and changed AUTHTYPE to 'database', turn it back off, and then delete any existing users.

This module is not active by default, and will say

NOTE: AUTHTYPE is not set to 'database' in /etc/amportal.conf - note that this module is not currently providing access control, and changing passwords here or adding users will have no effect unless AUTHTYPE is set to 'database'.

Changing AUTHTYPE to database

To change AUTHTYPE, bring up the FreePBX Administration.

Click on the Tools tab, and then choose Config Edit.

When the “phpconfig for Asterisk PBX” comes up, select /etc. Click on amportal.conf.

```
Edit: amportal.conf
```

# This file contains settings for components of the Asterisk Management Portal
# Spaces are not allowed!
# Run /usr/etc/AOX/apply_conf.sh after making changes to this file
**AUTHTYPE** - Type of authentication to use

**none** - (default, uses the Apache Security Server) No authentication is used, the username is assumed to be AMPDBUSER and always has admin access (rights to everything).

**database** – Use the FreePBX Administrator Module. Authentication is done through the ampuser table, managed by the 'Administrators' page in the GUI. Each user can have different access rights.

There is a fallback that if the AMPDBUSER logs in with AMPDBPASS, they are granted admin rights.

**webserver** - FreePBX provides access control only, no authentication. A username passed by the webserver is assumed to be logged in successfully, and is given the rights that user has in the ampuser table (managed from the 'Administrators' page in the GUI). If the username is the same as AMPDBUSER, then they are given admin rights.

This method should only be used when the webserver is providing some kind of authentication, for example by .htaccess files, Apache's mod_auth_ldap, etc. Warning: without providing protection, ANY user name passed, regardless of the password, will be accepted! A misconfigured webserver can leave your system unsecure.

Run `/usr/src/AMP/apply_conf.sh` after making changes to this file

After enabling database mode, if you go to /admin/modules, you will get a directory listing. With that you could access many areas of FreePBX by typing the URL directly. This is not good. Re-enabling .htaccess in the module directory will prevent this.

The default login for using AUTHTYPE=database is admin/admin.

### Adding a User

**Username:**

**Password:**

**Access Restrictions**

**Department Name:**

**Extension Range:**

**Admin Access:**

Submit Changes

**General Settings**

Enter a username and password in the General Settings section. If this is the first user, make sure that you select 'ALL SECTIONS' in the 'Admin Access' list so you can get back in there. As soon as you add the first user, you will then be prompted for a username and password. Log in as the user you've just created.

**Access Restrictions**

**Department Name** - Restrict this user's view of Digital Receptionist menus and System Recordings to only those for this department.

**Extension Range** - Restrict this user's view to only Extensions, Ring Groups, and Queues within this range. When this user is logged in, they will only see the range specified here. This is useful if you're setting up multiple tenants on one system.
Admin Access - This is a multiple-selection box. You can select a range of areas they're allowed to access by either holding down Control (or 'apple' on Mac's) and selecting individual ones, or dragging the mouse over the list of ones you want to give them access to.

Select the Admin Sections this user should have access to:

- FreePBX System Status
- Module Admin
- Administrators
- Extensions
- Feature Codes
- General Settings
- Outbound Routes
- Trunks
- Announcements
- Blacklist
- CallerID Lookup Sources
- Day/Night Control
- Follow Me
- IVR
- Inbound Routes
- Queue Priorities
- Queues
- Ring Groups
- Time Conditions
- Time Groups
- Zap Channel DIDs
- Callback Conferences
- DISA
- Languages
- Misc Applications
- Misc Destinations
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- VoiceMail Blasting
- A2Billing Admin
- Config Edit
- Sys Info
- phpMyAdmin
- Asterisk
- Logfiles
- Asterisk Info
- Asterisk CLI
- Asterisk Phonebook
- Backup & Restore
- Custom Destinations
- Custom Extensions
- DUNDi Lookup
- Java SSH
- PHP Info
- PHPAGI Config
- Weak Password Detection
- Customer DB
- Gabcast
- Inventory
- Print Extensions
- Apply Changes
- Bar
- Add Extension
- ALL SECTIONS

At the bottom of this page click on Submit Changes, then click on Apply Configuration Changes and then Continue with reload to apply any changes / updates.
**Inbound Call Control**

When an incoming call from PSTN or VoIP trunk is received, asterisk needs to know where to direct it. It can be directed to a ring group, an extension, Digital Receptionist (IVR) or Queue. For this purpose, Inbound Route needs to be set up. For each trunk, a corresponding inbound route must be created in order to use that route for calls. At least one Inbound Route must be created for the PBX to answer incoming calls, whether from PSTN trunk or SIP calls.

The Inbound Call Control menus setup everything that happens to an inbound call. These functions include: Inbound Routes, Zap Channel DIDs, Announcements, Blacklist, CallerID Lookup Sources, Day/Night Control, Follow Me, IVR, Queue Priorities, Queues, Ring Groups, Time Conditions, and Time Groups.

**Inbound Routes**

**Add Incoming Route**

- **Description** - Provide a meaningful description of what this incoming route is.
- **DID Number** - Define the expected DID Number if your trunk passes DID on incoming calls. Leave this blank to match calls with any or no DID info. You can also use a pattern match (e.g. _2[345]X) to match a range of numbers. A DID number can only be used once.

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

- **Caller ID Number** - Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no CID info. In addition to standard dial sequences, you can also put Private, Blocked, Unknown, Restricted, Anonymous and Unavailable in order to catch these special cases if the Telco transmits them.

- **CID Priority Route** - This effects CID ONLY routes where no DID is specified. If checked, calls with this CID will be routed to this route, even if there is a route to the DID that was called. Normal behavior is for the DID route to take the calls. If there is a specific DID/CID route for this CID, that route will still take the call when that DID is called.

**Options**

- **Alert Info** - ALERT_INFO can be used for distinctive ring with SIP devices.
- **CID name prefix** - You can optionally prefix the Caller ID name. i.e.: If you prefix with "Sales:" a call from John Doe would display as "Sales:John Doe" on the extensions that ring.

- **Music On Hold** - Set the MoH class that will be used for calls that come in on this route. For example, choose a type appropriate for routes coming in from a country which may have announcements in their language.

- **Signal RINGING** - Some devices or providers require RINGING to be sent before ANSWER. You'll notice this happening if you can send calls directly to a phone, but if you send it to an IVR, it won't connect the call.

- **Pause Before Answer** - An optional delay to wait before processing this route. Setting this value will delay the channel from answering the call. This may be handy if external fax equipment or security systems are installed in parallel and you would like them to be able to seize the line.
Privacy

**Privacy Manager** - If no Caller ID is sent, Privacy Manager will ask the caller to enter their 10 digit phone number. The caller is given 3 attempts. The number of digits and attempts can be defined in privacy.conf. If a user has Call Screening enabled, the incoming caller will be asked to enter their Caller ID here if enabled, and then to say their name once determined that the called user requires it.

Fax Handling

**Fax Extension** - Select 'system' to have the system receive and email faxes. Options are “FreePBX default”, disabled, system, and any defined extensions. The “FreePBX default” is defined in General Settings.

**Fax Email** - Email address is used if 'system' has been chosen for the fax extension above. Leave this blank to use the FreePBX default in General Settings.

**Fax Detection Type** - Selecting Zaptel or NVFax will immediately answer the call and play ringing tones to the caller for the number of seconds in Pause below. Use NVFax on SIP or IAX trunks.

**Pause After Answer** - The number of seconds we should wait after performing an Immediate Answer. The primary purpose of this is to pause and listen for a fax tone before allowing the call to proceed.

CID Lookup Source

**Source** - Sources can be added in Caller Name Lookup Sources section.

Set Destination

The caller can be sent to any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

At the bottom of this page click on **Submit, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.

If a destination is chosen, but you do not want a specific destination for this route, you can select **Clear Destination & Submit**.
Zap Channel DIDs

A DID (Direct Inward Dialing) is an incoming phone number.

There are 4 FXO modules in a TDM410 card, so the zap channels will be 1, 2, 3 and 4. Each FXO is a channel (this can be viewed in zapata_auto.conf file). You need to define your DID (incoming phone number) for each ZAP channel.

Zap Channel DIDs allow you to assign a DID to specific Zap Channels. You can supply the same DID to multiple channels. This would be a common scenario if you have multiple POTS lines that are on a hunt group from your provider. You MUST assign the channel's context to from-zaptel for these settings to have effect. It will be a line that looks like: “context = from-zaptel” in your zapata.conf configuration affecting the specified channel(s). Once you have assigned DIDs you can use standard Inbound Routes with the specified DIDs to route your calls.

To handle inbound calls from a ZAP trunk, you simply enter the Zaptel Channel number in the zaptel channel field. This will determine which zap call be directed to where.

To add the channel, click on Add Zap Channel.

Add Channel

**Channel:** The Zap Channel number to map to a DID

**Description:** A useful description describing this channel

**DID:** The DID that this channel represents. The incoming call on this channel will be treated as if it came in with this DID and can be managed with Inbound Routing on DIDs.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Announcements

The Announcement module allows you to play an announcement to a caller. You can then send the caller to another destination or back to the IVR that sent him to the announcement.

Click on the **Add Announcement** button.

**Description** - The name of this announcement.

**Recording** - Message to be played. To add additional recordings use the "System Recordings" MENU to the left.

**Repeat** - Key to press that will allow for the message to be replayed. If you choose this option there will be a short delay inserted after the message. If a longer delay is needed it should be incorporated into the recording. Options are Disable, 0 – 9, *, or #.

**Allow Skip** - If the caller is allowed to press a key to skip the message.

**Return to IVR** - If this announcement came from an IVR and this box is checked, the destination below will be ignored and instead it will return to the calling IVR. Otherwise, the destination below will be taken. Don't check if not using in this mode. The IVR return location will be to the last IVR in the call chain that was called so be careful to only check when needed. For example, if an IVR directs a call to another destination which eventually calls this announcement and this box is checked, it will return to that IVR which may not be the expected behavior.

**Don't Answer Channel** - Check this to keep the channel from explicitly being answered. When checked, the message will be played and if the channel is not already answered it will be delivered as early media if the channel supports that. When not checked, the channel is answered followed by a 1 second delay. When using an announcement from an IVR or other sources that have already answered the channel, that 1 second delay may not be desired.

**Destination after playback**

After the announcement is played, the caller can be sent to any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

At the bottom of this page click on **Submit Changes, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
Blacklist

By using CID Screening, this module allows you to manage a blacklist. Adding a number to the blacklist will not allow the number into your system. The blacklisted caller will hear "The number you have reached is not in service."

<table>
<thead>
<tr>
<th>Number entries</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number</td>
</tr>
<tr>
<td>Delete</td>
</tr>
<tr>
<td>Edit</td>
</tr>
<tr>
<td>Submit Changes</td>
</tr>
</tbody>
</table>

Hit *32 to blacklist the last number called into your system.
On some phones you can setup a speeddial for *32 to blacklist callers.

Click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Caller ID Lookup Services

This module provides the ability to specify Sources where inbound calls can have their Caller ID looked up so Caller ID Names can be used or changed.

The Caller ID Lookup Sources module enables your FreePBX system to lookup Caller Names that are related to your number whether they are in your phonebook, in a database, or via an HTTP lookup. The module can also be used with scripts in the agi-bin directory of your asterisk configuration.

Add Source

A Lookup Source let you specify a source for resolving numeric caller IDs of incoming calls; you can then link an inbound route to a specific CID source. This way you will have more detailed CDR reports with information taken directly from your CRM. You can also install the phonebook module to have a small number <-> name association. Pay attention, name lookup may slow down your PBX.

Source Description - Enter a description for this source.

Source type - Select the source type, you can choose between:

- Internal: Use the astDB (Asterisk database) as a lookup source, use phonebook module to populate it.
- ENUM: Use DNS to lookup caller names, it uses ENUM lookup zones as configured in enum.conf.
- HTTP: It executes an HTTP GET passing the caller number as argument to retrieve the correct name
- MySQL: It queries a MySQL database to retrieve caller name

NOTE: If you select HTTP or MySQL, additional parameters will pop up.

Cache results - You can check this and the system will keep successful lookups for future use, thus eliminating excessive remote lookups for numbers that have already called you in the past. If checked, the results are cached to astDB; it will overwrite present values. It does not affect internal source behavior.

MySQL parameters

- Host: MySQL Host
- Database: Database name
- Query: Query, special token '[NUMBER]' will be replaced with caller number
  e.g.: SELECT name FROM phonebook WHERE number LIKE '%[NUMBER]%'
- Username: MySQL Username
- Password: MySQL Password
HTTP Lookup Configuration

Configuring the module to lookup a Caller Name (cn) via HTTP lookup is easy. Most HTTP lookup providers will provide you with a string you'll need to query with (query string). It might look similar to this: `http://cnam1.edicentral.net/getcnam?q=C&f=S&dn=[NUMBER]` where `[NUMBER]` is a 10 digit telephone number. It is necessary to break up the string into its various components to populate the CID Lookup Source Fields.

**Host:** Host name or IP address

**Port:** Port HTTP server is listening at (default 80)

**Username:** Username to use in HTTP authentication issued by the provider if it's needed, otherwise, it's blank. In this case, it's blank.

**Password:** Password to use in HTTP authentication issued by the provider if it's needed, otherwise, it's blank. In this case, it's blank.

**Path:** Path of the file to GET. This is the first thing after the hostname (including the slash. Up to, but NOT including the '?' symbol. That's reserved for the query line otherwise it just won't work.

In the above example it would be: `/getcnam`

**Query:** Query string, special token '[NUMBER]' will be replaced with caller number

In the above example it would be: `?q=C&f=S&dn=[NUMBER]`

At the bottom of this page click on **Submit Changes, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
Day / Night Mode Control

This module allows for the inbound routing changes to be placed via a phone call - without requiring access to the web configuration page. The module is useful for any situation where inbound routing needs to be changed by end users on-the-fly. An example of this would be a situation where an office would close early because of bad weather. In this case, just dial the appropriate feature code, and voila! All inbound calls go straight to the "night" IVR (or voice mail or calls get forwarded to an answering service, etc.)

Day/Night Feature Code Index - There are a total of 10 Feature code objects, 0-9, each can control a call flow and be toggled using the day/night feature code plus the index.

Description - Description for this Day/Night Control

Current Mode - This will change the current state for this Day/Night Mode Control, or set the initial state when creating a new one. Select Day or Night

Optional Password - You can optionally include a password to authenticate before toggling the day/night mode. If left blank anyone can use the feature code and it will be un-protected.

DAY - Destination to use when set to DAY mode. The caller can be sent to any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

NIGHT - Destination to use when set to NIGHT mode. The caller can be sent to any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

See also Day/Night Control with a Time Condition.

Click on Save when your configurations are done.
Follow Me

Follow-Me settings are configured exactly the same way as a ring group, but it's tied directly to your extension. Like Ring Groups, you have the ability to use an announcement to alert people that they're being transferred elsewhere.

You can simply put in the extension of the Follow Me number with a choice to go to its voicemail if the first in the Follow-Me List is not answered and you will be accomplishing exactly the same thing as if the extension was being dialed. However, you can now diverge with such simple things as changing your ring time to override the default, adding an announcement, going to an alternative voicemail or other destination if not reached, and of course adding multiple numbers and ring strategies when someone tries to call that number.

**Choose a user/extension:** Choose a user/extension from the right side of the screen.

### Follow Me: 212

<table>
<thead>
<tr>
<th>Disable</th>
<th>Initial Ring Time</th>
<th>Ring Strategy</th>
<th>Ring Time (max 60 sec)</th>
<th>Follow-Me List</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1</td>
<td>hunt</td>
<td>20</td>
<td>212 9918125212110#</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Extension Quick Pick</th>
<th>Announcement</th>
<th>Play Music On Hold?</th>
<th>CID Name Prefix</th>
<th>Alert Info</th>
<th>Confirm Calls</th>
<th>Remote Announce</th>
<th>Too-Late Announce</th>
</tr>
</thead>
<tbody>
<tr>
<td>pick extension</td>
<td>None</td>
<td>Ring</td>
<td></td>
<td></td>
<td></td>
<td>Default</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Disable** - By default (not checked) any call to this extension will go to this Follow-Me instead, including directory calls by name from IVRs. If checked, calls will go only to the extension. However, destinations that specify FollowMe will come here. Checking this box is often used in conjunction with VmX Locator, where you want a call to ring the extension, and then only if the caller chooses to find you do you want it to come here.

**Initial Ring Time** - This is the number of seconds to ring the primary extension prior to proceeding to the follow-me list. A 0 setting will bypass this.

**Ring Strategy:**

- **hunt:** take turns ringing each available extension
- **ringallv2:** ring primary extension for initial ring time followed by all additional extensions until one answers
- **ringall:** ring all available channels until one answers (default)
- **memoryhunt:** ring first extension in the list, then ring the 1st and 2nd extension, then ring 1st 2nd and 3rd extension in the list.... etc.

**-prim:** these modes act as described above. However, if the primary extension (first in list) is occupied, the other extensions will not be rung. If the primary is FreePBX DND, it won't be rung. If the primary is FreePBX CF unconditional, then all will be rung

**firstavailable:** ring only the first available channel
**firstnotonphone**: ring only the first channel which is not off hook - ignore CW

**Ring Time** (max 60 sec) Time in seconds that the phones will ring. For all hunt style ring strategies, this is the time for each iteration of phone(s) that are rung.

**Follow-Me List:**
List extensions to ring, one per line, or use the Extension Quick Pick. The extension can also be included in the follow-me list.

You can include an extension on a remote system, or an external number by suffixing a number with a pound (#). ex: 2448089# would dial 2448089 on the appropriate trunk (see Outbound Routing).

**Extension Quick Pick** - Choose an extension to append to the end of the extension list above.

**Announcement** - Message to be played to the caller before dialing this group. To add additional recordings please use the "System Recordings" MENU to the left.

**Play Music On Hold?** If you select a Music on Hold class to play, instead of 'Ring', they will hear that instead of Ringing while they are waiting for someone to pick up.

**CID Name Prefix** - You can optionally prefix the Caller ID name when ringing extensions in this group. For example, if you prefix with "Sales:"; a call from John Doe would display as "Sales:John Doe" on the extensions that ring.

**Alert Info** - You can optionally include an Alert Info which can create distinctive rings on SIP phones.

**Confirm Calls** - Enable this if you’re calling external numbers that need confirmation – for example a mobile phone may go to voicemail which will pick up the call. Enabling this requires the remote side push 1 on their phone before the call is put through. This feature only works with the ringall/ringall-prim ring strategy.

**Remote Announce** - Message to be played to the person RECEIVING the call, if 'Confirm Calls' is enabled. To add additional recordings use the "System Recordings" option on the left.

**Too-Late Announce** - Message to be played to the person RECEIVING the call, if the call has already been accepted before they push 1. To add additional recordings use the "System Recordings" MENU to the left.

**Destination if no answer:**
The caller can be sent to any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

At the bottom of this page click on **Submit Changes, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
Follow Me Example

After setting up the extensions, you may decide that you want Asterisk to call another pre-arranged extension, if an extension called does not answer.

To do this, select the Follow me option; Setup -> Follow Me. Select the extensions that you want to define (the extension selection is on the right of the screen).

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

- **Initial Ring Time**: (in seconds)
- **Ring strategy**: hunt (to call the numbers in sequence)
- **Ring time**: For each iteration on the list
- **Follow-Me List**: 212 followed by 212’s cell phone.
- **Destination if no answer**: Core–Hangup

Every time 212’s extension is called, Asterisk will try to connect to extension 212 and if no answer, it will call the cell phone. If still no answer, it will simply hang up.

Click **Submit** followed by clicking the bar on top of the screen to finalize this selection.
IVR - Digital Receptionist

The IVR selection brings up the Digital Receptionist GUI. You use the Digital Receptionist GUI to add and edit an Interactive Voice Response system. When you configure your IVR, it is best to plan it out properly. Please see the separate section: Interactive Voice Response System, for complete details on doing this.

To create a default IVR, just click on the Add IVR button on the right.

**Digital Receptionist**

**Edit Menu Main IVR**

<table>
<thead>
<tr>
<th>Change Name</th>
<th>Announcement</th>
<th>Timeout</th>
<th>Enable Directory</th>
<th>VM Return to IVR</th>
<th>Directory Context</th>
<th>Enable Direct Dial</th>
<th>Loop Before t-dest</th>
<th>Timeout Message</th>
<th>Loop Before i-dest</th>
<th>Invalid Message</th>
<th>Repeat Loops</th>
</tr>
</thead>
<tbody>
<tr>
<td>Main IVR</td>
<td>T-is-not-available</td>
<td>10</td>
<td>Yes</td>
<td>Yes</td>
<td>default</td>
<td>Yes</td>
<td>Yes</td>
<td>T-is-not-available</td>
<td>Yes</td>
<td>wrong-try-again-smarty</td>
<td>2</td>
</tr>
</tbody>
</table>

**Change Name** - This changes the short name, visible on the right, of this IVR.

**Announcement** - Message to be played to the caller. To add additional recordings use the “System Recordings”.

**Timeout** - The amount of time (in seconds) before the 't' option, if specified, is used.

**Enable Directory** - Let callers into the IVR dial '#' to access the directory.

**VM Return to IVR** - If checked, upon exiting voicemail a caller will be returned to this IVR if they got a user's voicemail.

**Directory Context** - When '#' is selected, this is the voicemail directory context that is used.

**Enable Direct Dial** - Let callers into the IVR dial an extension directly.

**Loop Before t-dest** - If checked, and there is a 't' (timeout) destination defined below, the IVR will loop back to the beginning if no input is provided for the designated loop counts prior to going to the timeout (t) destination.

**Timeout Message** - If a timeout occurs and a message is selected, it will be played in place of the announcement message when looping back to the top of the IVR. It will not be played if the t destination is the next target.

**Loop Before i-dest** - If checked, and there is an 'i' (invalid extension) destination defined below, the IVR will play invalid option and then loop back to the beginning for the designated loop counts prior to going to the invalid (i) destination.

**Invalid Message** - If an invalid extension is pressed and a message is selected, it will be played in place of the announcement message when looping back to the top of the IVR. It will not be played if the t destination is the next target. If nothing is selected, the system will play a default invalid extension message before going back to the main announcement.

**Repeat Loops** - The number of times we should loop when invalid input or no input has been entered before going to the defined or default generated 'i' or 't' options. If the 'i' or 't' options are defined, the above check boxes must be checked in order to loop. 0-9.
Destinations:
The caller can be sent to any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

If the first destination does not answer, it will go to the second destination, and so forth. Destinations are only displayed if there is at least one entry in there. So if, for example, you have the DISA module enabled, but no DISA entries, it will not appear in the list.

Use 'Increase Options' or 'Decrease Options' to alter the number of options available. This won't let you decrease it to less than the number of options that are currently set. To delete an option, simply leave the selection blank.

**Return to IVR** - Check this box to have this option return to a parent IVR if it was called from a parent IVR. If not, it will go to the chosen destination. The return path will be to any IVR that was in the call path prior to this IVR which could lead to strange results if there was an IVR called in the call path but not immediately before this.

In the box on the left, enter the option for the user. This may be one, or a series of numbers, or, 'i', or 't'. 'i' and 't' have special meanings:

- **i**: This is the destination used when a caller enters an invalid option - if you only have 1 2 and 3 defined, and they push 4, it will jump to this destination. The default option, if you don't supply an 'i' destination, is to replay the current menu. If they hit 'i' more than three times, the call is hung up.

- **t**: This is the destination used when nothing happens. You might wish to have this one go directly to an operator, in case the caller doesn't have a DTMF phone. As with 'i', the default is to replay, and if it's been replayed three times, hang up.

Click on **Save, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.

For complete details on setting up an IVR, please see the section **Interactive Voice Response System**.
Queue Priorities

Queue Priority allows you to set a caller's priority in a queue. By default, a caller's priority is set to 0. Setting a higher priority will put the caller ahead of other callers already in a queue. The priority will apply to any queue that this caller is eventually directed to. You would typically set the destination to a queue, however that is not necessary. You might set the destination of a priority customer DID to an IVR that is used by other DIDs, for example, and any subsequent queue that is entered would be entered with this priority.

To add a Queue Priority click on the Add Queue Priority button to the right.

Add Queue Priority Instance

**Description:** The descriptive name of this Queue Priority instance.

**Priority:** The Queue Priority to set, from 0-20

**Destination:**

- Main IVR
- Ring All <200>
- Day/Night Switch
- Olmous <500>
- Hangup
- <204> Marc Hall
- <204> Marc Hall (busy)
- Phonebook Directory

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Queues

Queues allow you to manage a large number of incoming calls, as you would expect to have in a Call Center. This is very intelligent application, and as such, it has a lot of configuration options.

Add Queue

Queue Number: Use this number to dial into the queue, or transfer callers to this number to put them into the queue. Agents will dial this queue number plus * to log onto the queue, and this queue number plus ** to log out of the queue. For example, if the queue number is 123:

123* = log in
123** = log out

Queue Name: Give this queue a brief name to help you identify it.

Queue Password: You can require agents to enter a password before they can log in to this queue. This setting is optional.

CID Name Prefix: You can optionally prefix the Caller ID name of callers to the queue. i.e.: If you prefix with "Sales: ", a call from John Doe would display as "Sales: John Doe" on the extensions that ring.

Wait Time Prefix: Yes no option. When set to Yes, the CID Name will be prefixed with the total wait time in the queue so the answering agent is aware how long they have waited. It will be rounded to the nearest minute, in the form of Mnn: where nn is the number of minutes. If the call is subsequently transferred, the wait time will reflect the time since it first entered the queue or reset if the call is transferred to another queue with this feature set.

Alert Info - ALERT_INFO can be used for distinctive ring with SIP devices.

Static Agents: Static agents are extensions that are assumed to always be on the queue. Static agents do not need to 'log in' to the queue, and cannot 'log out' of the queue. List the extensions to ring, one per line. You can include an extension on a remote system, or an external number (Outbound Routing must contain a valid route for external numbers).

Extension Quick Pick - Choose an extension to append to the end of the static agents list above.

Queue Options

Agent Announcement: Announcement played to the Agent prior to bridging in the caller. Example: "the Following call is from the Sales Queue" or "This call is from the Technical Support Queue". To add additional recordings please use the "System Recordings" MENU to the left.

Compound recordings composed of 2 or more sound files are not displayed as options since this feature can not accept such recordings.

Join Announcement: Announcement played to callers once prior to joining the queue. To add additional recordings please use the "System Recordings" MENU to the left.

Music on Hold Class: Music (or Commercial) played to the caller while they wait in line for an available agent. Choose "inherit" if you want the MoH class to be what is currently selected, such as by the inbound route. This music is defined in the "Music on Hold" Menu to the left. Options are inherit, default, and none.

Ringing Instead of MoH: Enabling this option will make the callers hear a ringing tone instead of Music on Hold. If this option is enabled, settings of the previous drop down are ignored.

Max Wait Time: The maximum number of seconds a caller can wait in a queue before being pulled out. 0 for unlimited or select 10 seconds up to 1 hour.
**Max Callers:** Maximum number of people waiting in the queue (0 for unlimited) up to 50.

**Join Empty:** If you wish to allow callers to join queues that currently have no agents, set this to Yes. This is not recommended. Set to **Strict** if callers cannot join a queue with no members or only unavailable members.

**Leave When Empty:** If you wish to remove callers from the queue if there are no agents present, set this to Yes. Set to **Strict** if callers cannot join a queue with no members or only unavailable members.

**Ring Strategy:**
- **ringall:** ring all available agents until one answers (default)
- **roundrobin:** take turns ringing each available agent
- **leastrecent:** ring agent which was least recently called by this queue
- **fewestcalls:** ring the agent with fewest completed calls from this queue
- **random:** ring random agent
- **rrmemory:** round robin with memory, remember where we left off last ring pass

**Agent Timeout:** The number of seconds an agent's phone can ring before we consider it a timeout. Unlimited or other timeout values may still be limited by system ringtime or individual extension defaults. Unlimited or from 1 to 60 seconds

**Retry:** The number of seconds we wait before trying all the phones again. Choosing 'No Retry' will exit the Queue and go to the fail-over destination as soon as the first attempted agent times-out, additional agents will not be attempted. No Retry, 0 to 20 seconds

**Wrap-Up-Time:** After a successful call, how many seconds to wait before sending a potentially free agent another call (default is 0, or no delay)

**Call Recording:** Incoming calls to agents can be recorded. (saved to /var/spool/asterisk/monitor) wav49wavgsm or No

**Event When Called:** When this option is set to YES, the following manager events will be generated:
- AgentCalled, AgentDump, AgentConnect and AgentComplete.

**Member Status:** When this option is set to YES, the following manager event will be generated:
- Queue Member Status

**Skip Busy Agents:** When set to Yes, agents who are on an occupied phone will be skipped as if the line were returning busy. This means that Call Waiting or multi-line phones will not be presented with the call and in the various hunt style ring strategies, the next agent will be attempted.

**Queue Weight:** Gives queues a 'weight' option, to ensure calls waiting in a higher priority queue will deliver its calls first if there are agents common to both queues. 0-10.

**Autofill:** If this is checked, and multiple agents are available, Asterisk will send one call to each waiting agent (depending on the ring strategy). Otherwise, it will hold all calls while it tries to find an agent for the top call in the queue making other calls wait.

**Agent Regex Filter:** Provides an optional regex expression that will be applied against the agent callback number. If the callback number does not pass the regex filter then it will be treated as invalid. This can be used to restrict agents to extensions within a range, not allow callbacks to include keys like *, or any other use that may be appropriate. An example input might be:  

```
^([2-4][0-9]{3})$
```

This would restrict agents to extensions 2000-4999. Or  

```
^([0-9]+)$
```

would allow any number of any length, but restrict the * key.

**WARNING:** Make sure you understand what you are doing or otherwise leave this blank!
Caller Position Announcements

**Frequency**: How often to announce queue position and estimated holdtime: (0 to Disable Announcements) up to 20 minutes.

**Announce Position**: Announce position of caller in the queue?

**Announce Hold Time**: Should we include estimated hold time in position announcements? Either yes, no, or only once; hold time will not be announced if < 1 minute. Yes, No, or Once

Periodic Announcements

**IVR Break Out Menu**: You can optionally present an existing IVR as a ‘break out’ menu. This IVR must only contain single-digit ‘dialed options’. The Recording set for the IVR will be played at intervals specified in ‘Repeat Frequency’, below. Options are None, IVR.

**Repeat Frequency**: How often to announce a voice menu to the caller (0 to Disable Announcements). 15 seconds to 20 minutes

Fail Over Destination

The Fail Over destination could be any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Ring Groups

This defines a 'virtual' extension that rings a group of phones simultaneously, stopping when any one of them is picked up. This is basically just a dumber version of Queues for those that don't need the extra functionality of it.

Click on the **Add Ring Group** button

- **Ring-Group Number**: The number users will dial to ring extensions in this ring group.
- **Group Description**: Provide a descriptive title for this Ring Group.
- **Ring Strategy**: Defines how the ring group will ring.
  - **ringall**: Ring all available channels until one answers (default)
  - **hunt**: Take turns ringing each available extension
  - **memoryhunt**: Ring first extension in the list, then ring the 1st and 2nd extension, then ring 1st 2nd and 3rd extension in the list, etc.
  - **-prim**: These modes act as described above. However, if the primary extension (first in list) is occupied, the other extensions will not be rung. If the primary is FreePBX DND, it won't be rung. If the primary is FreePBX CF unconditional, then all will be rung.
  - **firstavailable**: Ring only the first available channel
  - **firstnotonphone**: Ring only the first channel which is not offhook - ignore CW
- **Ring Time**: Time in seconds that the phones will ring. For all hunt style ring strategies, this is the time for each iteration of phone(s) that are rung.
- **Extension List**: List extensions to ring, one per line, or use the Extension Quick Pick below to insert them here. You can include an extension on a remote system, or an external number by suffixing a number with a '#'. Example: 2448089# would dial 2448089 on the appropriate trunk (see Outbound Routing).
  - Extensions (without a '#') will not ring a user's Follow-Me. To dial Follow-Me, Queues and other numbers that are not extensions, put a '#' at the end.

**Extension Quick Pick** - Choose an extension to append to the end of the extension list above.

- **Announcement**: Message to be played to the caller before dialing this group. To add additional recordings please use the "System Recordings" menu.
- **Play Music On Hold?**: If you select a Music on Hold class to play, instead of 'Ring', they will hear that instead of Ringing while they are waiting for someone to pick up. Ring default none.
CID Name Prefix: You can optionally prefix the Caller ID name when ringing extensions in this group. i.e.: If you prefix with "Sales:", a call from John Doe would display as "Sales:John Doe" on the extensions that ring.

Alert Info - ALERT_INFO can be used for distinctive ring with SIP devices.

Ignore CF Settings - When checked, agents who attempt to Call Forward will be ignored; this applies to CF, CFU and CFB. Extensions entered with '#' at the end, for example to access the extension's Follow-Me, might not honor this setting.

Skip Busy Agent - When checked, agents who are on an occupied phone will be skipped as if the line were returning busy. This means that Call Waiting or multi-line phones will not be presented with the call; and in the various hunt style ring strategies, the next agent will be attempted.

Confirm Calls - Enable this if you’re calling external numbers that need confirmation - e.g., a mobile phone may go to voicemail which will pick up the call. Enabling this requires the remote side push 1 on their phone before the call is put through. This feature only works with the ringall ring strategy:

Remote Announce: Message to be played to the person RECEIVING the call, if 'Confirm Calls' is enabled. To add additional recordings use the "System Recordings" MENU to the left.

Too-Late Announce: Message to be played to the person RECEIVING the call, if the call has already been accepted before they push 1. To add additional recordings use the "System Recordings" MENU to the left.

Destination If No Answer
The destination if there is no answer could be any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Time Conditions

Time Conditions are a module that appears as a destination when installed. It allows you to do an 'if' based on the current Time, Weekday, Day of the Month, or Month. At the moment it's reasonably basic with no support for 'AND' or 'OR', but you can chain together time conditions to do the same thing.

Click on the Add Time Condition button on the right.

**Time Condition name**: Give this Time Condition a brief name to help you identify it.

**Time Group**: Select a Time Group created under Time Groups. Matching times will be sent to matching destination. If no group is selected, call will always go to no-match destination. --Select a Group--

**Day/Night Mode Association**

Associate with: If a selection is made, this timecondition will be associated with that featurecode and will allow this timecondition to be direct overridden by that day/night mode feature code. No Association - Force Day - Force Night

**Destination if time matches:**

- IVR: Main IVR
- Ring Groups: Ring All <200>
- Day Night Mode: [0] Day/Night Switch
- VoiceMail Blasting: [0] Extension <500>
- Terminate Call: Hangup
- Extensions: <204> Marc Hall
- Voicemail: <204> Marc Hall (busy)
- Phonebook Directory: Phonebook Directory

**Destination if time does not match:**

- IVR: Main IVR
- Ring Groups: Ring All <200>
- Day Night Mode: [0] Day/Night Switch
- VoiceMail Blasting: [0] Extension <500>
- Terminate Call: Hangup
- Extensions: <204> Marc Hall
- Voicemail: <204> Marc Hall (busy)
- Phonebook Directory: Phonebook Directory

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.

See also Day/Night Control with a Time Condition.
The Day/Night Control module is similar to many other modules in the system. You can create multiple day/night feature codes and each can be used as a destination from within FreePBX as well as provide two destinations of their own. The feature code itself is nothing more than a two way toggle switch. You can change it to either DAY or NIGHT and the state is toggled each time you dial the feature code.

If you start with a piece of paper and a flow chart, drawing what you need from left to right, then create the destinations, time conditions, and day/night controls from right to left, you find it much easier.

In effect, the call comes in via inbound routes, and then you pass the call from FreePBX module to FreePBX module, and at each step, a decision can be made about the call, and where to send it next until it reaches the destination.

Example of Day/Night Control Feature Code used with a Time Condition

You have an Inbound Route for your main Company DID, that points to a Time Condition set to ring your receptionist at queue 1200 during the normal business hours of 8:00am to 5:00pm Monday-Friday. After hours, it rings the After Hours IVR. You would like to provide an ability to override the Time Condition with a day/night feature code.
Example:

Create our new day/night feature code and give it index 0 (or whatever you prefer).

Give this a useful name: Receptionist Override

Set the Current Mode to the time of day that it currently is. Note that all of your Day/Night Codes will be set the same.

Provide an optional Password if you want to require password access to this feature code.

Under DAY destination, choose the Receptionist Extension.

Under NIGHT destination, choose the IVR.

Press Save

Go to the Inbound Route that was pointing to the Time Condition and change it to point to your newly created Receptionist Override Day Night mode destination.

Now you can dial *280, optionally entering a password if configured, and change between the Day mode, where the call continues to be controlled by the Time Condition, or the Night mode, where the call is routed around the Time Condition to the normal After Hours IVR.

Accessing the Day/Night Control from Outside

To enable/disable the feature code from an outside line, use the Misc Destination:

Go to the Misc Destinations Module

Type in a meaningful description: Receptionist Override

In the Dial Box type: *280

Now use this Misc Destination as a destination for an Inbound Route, IVR selection, or other means.

NOTE: Day/Night Controls override Time Conditions.

NOTE: There are plenty of other ways you can use the day/night feature code and there is no need to do it in conjunction with a Time Condition. It is simply a module that can be chosen as a destination by other modules and provides a two way toggle switch to continue the call routing to either its Day or Night destinations.

See also:

**Time Groups**

Time, days, and months are handled by the time group’s module. The user will simply select which time group to use for this time condition. The Time Conditions module will attempt to upgrade current conditions. The time group’s module should be installed first and is marked as a prerequisite.

Time Groups can have multiple times defined for the same group, greatly simplifying complex holiday time conditions which used to require multiple time conditions strung together.

**Description:** This will display as the name of this Time Group.

<table>
<thead>
<tr>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Time Group</strong></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>New Time</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Time to start:</strong> 00:00 to 23:59</td>
</tr>
<tr>
<td><strong>Time to finish:</strong> 00:00 to 23:59</td>
</tr>
<tr>
<td><strong>Week Day Start:</strong> Monday-Sunday</td>
</tr>
<tr>
<td><strong>Week Day finish:</strong> Monday-Sunday</td>
</tr>
<tr>
<td><strong>Month Day start:</strong> 00:00 to 23:59</td>
</tr>
<tr>
<td><strong>Month Day finish:</strong> 00:00 to 23:59</td>
</tr>
<tr>
<td><strong>Month start:</strong> January-December</td>
</tr>
<tr>
<td><strong>Month finish:</strong> January-December</td>
</tr>
</tbody>
</table>

At the bottom of this page click on **Submit, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
Internal Options & Configurations

Callback

The Callback module allows you to setup a destination that calls a user back and provides them access to an application. An example of this would be a caller that dials your system, disconnects, and is called back and then provided a DISA dial-tone to make a phone call. This is useful for reducing mobile phone charges as well as other applications. Outbound calls will proceed according to the dial patterns in Outbound Routes.

Click on the [Add Callback] button.

Callback Description: Enter a description for this callback.

Callback Number: (Optional) Enter the number to dial for the callback. Leave this blank to just dial the incoming Caller ID Number.

Delay Before Callback: (Optional) Enter the number of seconds the system should wait before calling back.

Destination after Callback:

Any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.

IMPORTANT NOTES:

If 1) the Telco only provides a 10-digit CLI, 2) your provider requires you to add a ‘1’, or 3) your own dialplan requires an access code like ‘9’ you may have problems.

There is no way to provide a number of retries, retry time or wait times for an answer. This app will try once, and if it fails, will not try again.
**Conferences (MeetMe)**

“Conferences”, also known as MeetMe, is a standard multi-party conferencing facility that is available as a destination.

Click on the Add Conference button.

- **Conference Number**: Use this number to dial into the conference.
- **Conference Name**: Give this conference a brief name to help you identify it.
- **User PIN**: You can require callers to enter a password before they can enter this conference. This setting is optional. If either PIN is entered, the user will be prompted to enter a PIN.
- **Admin PIN**: Enter a PIN number for the admin user. This setting is optional unless the 'leader wait' option is in use, then this PIN will identify the leader.

**Conference Options**

- **Join Message**: Message to be played to the caller before joining the conference. To add additional recordings please use the "System Recordings" MENU to the left.
- **Leader Wait**: Wait until the conference leader (admin user) arrives before starting the conference.
- **Quiet Mode**: Quiet mode (do not play enter/leave sounds)
- **User Count**: Announce user(s) count on joining conference
- **User join/leave**: Announce user join/leave
- **Music on Hold**: Enable Music On Hold when the conference has a single caller
- **Allow Menu**: Present Menu (user or admin) when '*' is received (‘send’ to menu)
- **Record Conference**: Record the conference call

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.

You can see your MeetMe Conference Activity in the Flash Operator Panel. Click on the Flash Operator Panel button.

When the conference is over, be certain to delete it.

**Delete Conference 8000**

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
DISA

DISA (which stands for Direct Inward System Access) allows you to provide an internal dial tone to external callers. When you configure a DISA destination, you can use it as a menu destination within a Digital Receptionist, so that you can get an internal Asterisk dialtone. This means you could call into your Asterisk system and dial out as if you were using an extension connected to the Asterisk box itself.

**IMPORTANT NOTE:** The security implications of DISA are obvious! Make sure you have a proper authentication scheme in place so that unauthorized callers cannot abuse your system!

Click on the **Add DISA** button.

**DISA name:** Give this DISA a brief name to help you identify it.

**PIN:** The user will be prompted for this number. If you wish to have multiple PINs, separate them with commas.

**Response Timeout:** The maximum amount of time it will wait before hanging up if the user has dialled an incomplete or invalid number. Default of 10 seconds

**Digit Timeout:** The maximum amount of time permitted between digits when the user is typing in an extension. Default of 5

**Require Confirmation:** Require Confirmation before prompting for password. Used when your PSTN connection appears to answer the call immediately.

**Caller ID:** (Optional) When using this DISA, the users CallerID will be set to this. Format is "User Name" <5551234>

**Context:** (Experts Only) Sets the context that calls will originate from. Leave this as from-internal unless you know what you're doing.

**Allow Hangup:** Allow the current call to be disconnected and dial tone presented for a new call by pressing the Hangup feature code: ** while in a call.

At the bottom of this page click on **Submit Changes, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
Languages

Languages allow you to change the language of the call flow and then continue on to the desired destination. For example, you may have an IVR option that says "For French Press 5 now". You would then create a French language instance and point it's destination at a French IVR. The language of the call's channel will now be in French. This will result in French sounds being chosen if installed. If the language directory does not exist, the system will play the general sound files, which are placed in the /var/lib/asterisk/sounds.

If your preferable language is set to de (German), then the system will try to play the sound files from the /var/lib/asterisk/sounds/de directory. The messages in these files are recorded in German. If your language is set to fr (French), the system will try to play the sound files from the /var/lib/asterisk/sounds/fr directory. The messages in these files are recorded in French. And so on.

Also, if there is no directory for your language, you can create one. Then you can record sound messages in your own language and to put them into the directory. You could also create male and female options.

Click on the Add Language button.

Add Language Instance

Description: The descriptive name of this language instance. For example "French Main IVR"

Language Code: The Asterisk language code you want to change to. For example: "fr" for French, or "de" for German.

See http://www.voip-info.org/wiki/view/Asterisk+sound+files+international
And http://downloads.asterisk.org/pub/telephony/sounds/

Destination:

Any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Misc Applications

Misc Applications are for adding feature codes that you can dial from internal phones that go to various destinations available in FreePBX. This is in contrast to the Misc Destinations module, which is for creating destinations that can be used by other FreePBX modules to dial internal numbers or feature codes.

See http://www.voip-info.org/wiki/view/Asterisk+addons

Click on the Add Misc Application button.

**Description:** The name of this application

**Feature Code:** The feature code/extension users can dial to access this application. This can also be modified on the Feature Codes page.

**Feature Status:** If this code is enabled or not.

**Destination:**

Any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
**Misc Destination**

Misc Destinations allow you to use anything you could dial from a standard extension as a destination. Misc Destinations are for adding destinations that can be used by other FreePBX modules, generally used to route incoming calls. If you want to create feature codes that can be dialed by internal users and go to various destinations, please see the Misc Applications module.

**Example:**

You might want to have an IVR option that is 'If you want to speak to Rob, you can connect to his mobile by pushing 2', and having a Misc Destination of Rob's Mobile 00402077155 (Note the leading 0, as that's what I use for an 'external' call)

Then in the IVR menu, you simply select 'Rob's Mobile' as a destination, and it will connect the caller through.

Click on the **Add Misc Destination** button.

---

**Description:** Give this Misc Destination a brief name to help you identify it.

**Dial:** Enter the number this destination will simulate dialing, exactly as you would dial it from an internal phone. When you route a call to this destination, it will be as if the caller dialed this number from an internal phone.

---

**Example:**

If we create this:

In the destinations, you will now see this:

---

At the bottom of this page click on **Submit Changes, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
Music on Hold

Here you can configure the Music on Hold files that will be played. You can configure various 'Classes' of Music on Hold, which are used in Queues. The idea behind that is your 'default' MoH is standard music, and your various queues can have different 'hold' music while they're waiting.

The default category has three wav's: fpm-world-mix.wav, fpm-calm-river.wav, and fpm-sunshine.wav.

**Upload a .wav or .mp3 file:** Select 'browse' and pick a MP3 file on your system. Then click 'Upload'. It will appear in the list of MOH files below.

**Volume Adjustment** - The volume adjustment is a linear value. Since loudness is logarithmic, the linear level will be less of an adjustment. You should test out the installed music to assure it is at the correct volume. This feature will convert MP3 files to WAV files. If you do not have mpg123 installed, you can set the parameter: AMPMPG123=false in your amportal.conf file

**Disable Random Play**

On the right you can choose

- **Add Music Category**
  - **Category Name:** Allows you to set up different categories for music on hold. This is useful if you would like to specify different hold music or commercials for various ACD Queues.

- **Add Streaming Category**
  - **Category Name:** Allows you to set up different categories for music on hold. This is useful if you would like to specify different hold music or commercials for various ACD Queues.
  - **Application:** This is the "application=" line used to provide the streaming details to Asterisk. See information on musiconhold.conf configuration for different audio and internet streaming source options.
  - **Optional Format:** Optional value for "format=" line used to provide the format to Asterisk. This should be a format understood by Asterisk such as ulaw, and is specific to the streaming application you are using. See information on musiconhold.conf configuration for different audio and internet streaming source options.

At the bottom of this page click on **Submit Changes, Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
NOTE: Music on Hold is also defined in the following menus:

Queues

Music on Hold Class: Music (or Commercial) played to the caller while they wait in line for an available agent. Choose "inherit" if you want the MoH class to be what is currently selected, such as by the inbound route. This music is defined in the "Music on Hold" Menu to the left. Options are inherit, default, and none.

Ringing Instead of MoH: Enabling this option will make the callers hear a ringing tone instead of Music on Hold. If this option is enabled, settings of the previous drop down are ignored.

Ring Groups

Play Music On Hold? If you select a Music on Hold class to play, instead of ‘Ring’, they will hear that instead of Ringing while they are waiting for someone to pick up. Ring default none

Conferences (MeetMe)

Music on Hold: Enable Music On Hold when the conference has a single caller.

Customizing MOH

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

PIAF supports native MP3. To change or add to your music on hold collections, simply upload all your MP3 to PIAF through the Music On Hold screen of FreePBX.

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

```
    musiconhold=default
```

Pretty much any mp3 will work with PiaF. Prior to that, it's better to convert it to a standard format. When converting MP3 music use the following settings for the best result:

- Bit Rate: 128
- Mono
- Constant Bit Rate (CBR)
- no ID3 Tagging

You may want to create a new music category directory to hold your favorite music. Do this through the FreePBX “On Hold Music” GUI.

Upload your favorite music to this directory through the FreePBX GUI.

Once finished uploading, click the “Enable Random Play” button and the red bar.

Make a little change to your musiconhold.conf file like the example below (this way you don’t have to delete the music in your default directory).
The musiconhold.conf file is now set out differently as per the example of my file below:

```conf
musiconhold.conf

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

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

```conf
musiconhold_additional.conf

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

**Free Classical Music on Hold**

Classic Cat: [http://www.classiccat.net/](http://www.classiccat.net/)

This site is basically royalty free (the owner requests that you email him though):

[http://ghostnotes.blogspot.com/](http://ghostnotes.blogspot.com/)
PIN Sets

The PIN Set module is used to manage lists of PINs that can be used to restrict access to features such as Outbound Routes. The PIN can also be added to the CDR record's 'accountcode' field.

It is typically used by Trunks, but it potentially could be used in DISA or anything else that uses PIN's for authentication.

Click on the Add Password Set button, to bring up the New PIN Set menu

**PIN Set Description:** Give the PIN Set a description.

**Record In CDR?** Select this box if you would like to record the PIN in the call detail records when used.

**PIN List:** Enter a list of one or more PINs. Enter only one PIN per line.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Paging and Intercom

This module is for specific phones that are capable of Paging or Intercom. This section is for configuring group paging; intercom is configured through Feature Codes. Intercom must be enabled on a handset before it will allow incoming calls. It is possible to restrict incoming intercom calls to specific extensions only, or to allow intercom calls from all extensions but explicitly deny from specific extensions.

This module works with Grandstream phones. Any phone that is always set to auto-answer should also work (such as the console extension if configured).

Example:
*80nnn: Intercom extension nnn
*54: Enable all extensions to intercom you (except those explicitly denied)
*54nnn: Explicitly allow extension nnn to intercom you (even if others are disabled)
*55: Disable all extensions from intercom you (except those explicitly allowed)
*55nnn: Explicitly deny extension nnn to intercom you (even if generally enabled)

Click the Add Paging Group button to define.

Paging and Intercom

Paging Extension: The number users will dial to page this group
Group Description: Provide a descriptive title for this Page Group.
Device List: Select Device(s) to page. This is the phone that should be paged. In most installations, this is the same as the Extension. If you are configured to use "Users & Devices" this is the actual Device and not the User. Use Ctrl key to select multiple.
Force if busy: If selected, will not check if the device is in use before paging it. This means conversations can be interrupted by a page (depending on how the device handles it). This is useful for "emergency" paging groups.

Duplex Paging: Typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all. This makes it like an "instant conference".
Default Page Group: Each PBX system can have a single Default Page Group. If specified, extensions can be automatically added (or removed) from this group in the Extensions (or Devices) tab. Making this group the default will uncheck the option from the current default group if specified.

At the bottom of this page click on Submit Changes, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Parking Lot

This module allows you to configure all the normal features.conf settings for the parking lot functionality of Asterisk.

**Parking Lot Configuration**

- **Enable Parking Lot Feature**
- **Parking Lot Extension:**
- **Number of Slots:**
- **Parking Timeout:**
- **Parking Lot Context:**

**Actions for Timed-Out Orphans**

- **Parking Alert-Info:**
- **CallerID Prepend:**
- **Announcement:**

**Destination for Orphaned Parked Calls:**

**Parking Lot Options**

**Enable Parking Lot Feature** - Check this box to enable the parking feature.

**Parking Lot Extension:** This is the extension where you will transfer a call to park it.

**Number of Slots:** The total number of parking lot spaces to configure. Example, if 70 is the extension and 8 slots are configured, the parking slots will be 71-79.

**Parking Timeout:** This is how long a call will remain parked. After the time is up, the PBX reconnects it to the original parker.

- 15 seconds-10 minutes

**Parking Lot Context:** This is the parking lot context. You should not change it from the default unless you know what you are doing.

**Actions for Timed-Out Orphans**

The more useful part of this module is to specify a destination for parked calls that get orphaned. This can occur if the call is not picked up and for some reason the original parker cannot be reached. (E.g. the original parker is on the phone and does not have call waiting or ignores it). In this case, call is diverted to the chosen destination which is any of the standard destinations provided in all modules that include such an option. Prior to sending the call to that destination, you can configure the following options to further identify the orphaned call:

- **Parking Alert-Info:** This is alert-Info to put in channel before going to defined destination below. This can create distinct rings on some SIP phones and can serve to alert the recipients that the call is from an Orphaned parked call.

- **CallerID Prepend:** String to attach to the front of the current Caller-ID associated with this call (if any), before going to defined destination below. This can serve to alert the recipients that the call is from an Orphaned parked call.

- **Announcement:** Optional message to be played to the orphaned caller prior to going on to the supplied destination below to reassure them that you are trying to get them back to someone.

To add additional recordings please use the “System Recordings” menu.

**Destination for Orphaned Parked Calls:**

Any of these destinations: IVR, Ring Groups, Day Night Mode, VoiceMail Blasting, Terminate Call (Various ways to terminate the call), any of the Extensions, Voicemail, or the Phonebook Directory.
System Recordings

System Recordings are used in Ring Groups and Conferences for various announcements.

Click on the Add Recording button.

System Recordings

Add Recording

Step 1: Record or upload

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

Alternatively, upload a recording in any supported asterisk format. Note that if you’re using .wav (e.g., recorded with Microsoft Recorder) the file must be PCM Encoded, 16 Bits, at 8000Hz.

Step 2: Name

Name this Recording: 

Click “SAVE” when you are satisfied with your recording

Step 1: Record or upload

You can make a recording from your phone or upload one from your PC in any supported asterisk format: wav, ulaw, alaw, sln, gsm, or g729.

NOTE: If you use .wav format (e.g., recorded with Microsoft Recorder) the file must be PCM Encoded, 16 Bits, at 8000Hz.

Step 2: Name

Enter a good name for the recording in the Name this Recording: field.

Step 3:

Click the Save button when you are satisfied with your recording.

At the bottom of this page click on Save, Apply Configuration Changes and then Continue with reload to apply any changes / updates.
Upload Pre-Recorded Material

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

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

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

Converting WAV Files

For those who wish to record their own sound prompts, the following may be of some assistance.
PiaF will be able to play virtually any sound that it has the codec for e.g. wav(pcm), wav49, gsm, g711, g729 etc. However, the gsm format seems to be the common format used for the default voice prompts.

Converting WAV to GSM

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

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

After recording the WAV sound files, transfer the sound files to the TEMP directory of the PiaF PBX. In this example, call one of the sound files “hello.wav”.

1) Login as root and change directory to the directory where you have transferred the sound files to e.g. cd /tmp

2) At the prompt issue the following commands.

• If your sound files were recorded in mono, 16 bit, 8000 Hz

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

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

```
sox hello.wav -r 8000 hello.gsm resample -q1
```

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

```
sox hello.wav -r 8000 -cl hello.gsm resample –q1
```

• If your sound files were recorded in ADPCM wav files, to convert to standard

```
sox file; sox hello.wav -r 8000 -c1 -s -w hello-out.wav resample -q1
```

3) Move the sound file to the sound directory where all your prompts are stored and you are done. Click on the Others menu to open it up. Click on Upload and Download. It is fairly intuitive from there.

You may also convert a number of WAV files at once using the following command.

In this example, let’s assume that the files were all recorded in stereo;

```
for a in *.wav; do sox "$a" -r 8000 -c1 "" echo $a|sed -e s/wav/\"gsm\" resample –q1
```
Converting WAV to SLN

SLN is the Asterisk native SLINEAR format. Recordings that are in SLN format will have the same quality and file size as WAV recordings. To convert wav file to sln, use the following command:

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

Further reference for converting wav sound files can be found at voip-info.org.

Customized Voice

The default pre-recorded voice prompts and announcements in PIAF are suitable for almost every situation, however there will be times when customised voice prompts or announcements will be required, e.g. to mix English and other language on a single announcement.

Although there are separate voice sets that can be used, sometimes we only require a few prompts and not the entire voice set. To satisfy this requirement, custom voice prompts can be recorded individually by using the System Recording facility of FreePBX as covered in this document. However, if you want to manually cut the script, you can do the following:

Under the [from-internal-custom] context of `extensions_custom.conf`, add the following codes.

```
[from-internal-custom]
; For custom recording
exten => 5678,1,Wait(2)
exten => 5678,2,Record(/tmp/my-recording:gsm)
exten => 5678,3,Hangup
; For playback of custom recording
exten => 5679,1,Playback(/tmp/my-recording)
exten => 5679,2,Hangup
```

To start recording, use one of the phone extension and dial 5678. At the beep, start recording the voice prompt.

The voice prompt will be saved as my-recording.gsm (gsm format) in the /tmp directory.

When completed, hang up and dial 5679. The voice prompt will be played back.

If you are not satisfied, repeat the above process.

Once you are satisfied, rename the file to something related and recognisable e.g. my_office_business_hours.gsm. The file should then be moved to the /var/lib/asterisk/sound directory.

The file can be played through your custom applications or prompts using the Playback or Background function of Asterisk.

A good information source on Asterisk sound files and how to create them can be found at http://voip-info.org/wiki/view/Asterisk+sound+files

To play these messages, use Webmin's Others / File Manager.

Go to the: /var/lib/asterisk/sounds/custom directory.

Double-click on the files to play them.

If they do not play, download the Apple QuickTime Player to play gsm files. You can get it at http://www.apple.com/quicktime/download/
Voice Mail Blasting

You can set up groups to send Voice Mail Blasts to.

Click on the Add VMBlast Group button.

Add VMBlast Group

VMBlast Number: The number users will dial to voicemail boxes in this VMBlast group.

Group Description: Provide a descriptive title for this VMBlast Group.

Audio Label: Play this message to the caller so they can confirm they have dialed the proper voice mail group number, or have the system simply read the group number.

Optional Password: You can optionally include a password to authenticate before providing access to this group voicemail list.

Voicemail Box List: Select voice mail boxes to add to this group. Use Ctrl key to select multiple.

Default VMBlast Group: Each PBX system can have a single Default Voicemail Blast Group. If specified, extensions can be automatically added (or removed) from this default group in the Extensions (or Users) tab. Making this group the default will uncheck the option from the current default group if specified.

At the bottom of the page click on Submit Changes. At the top of the page click Apply Configuration Changes and then Continue with reload to apply any changes / updates.
## FreePBX Tools

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- **Notices, FreePBX & System Statistics, Uptime, and Server Status**
  - Check for updates online & if available update modules.
- **Easy access to edit or view FreePBX modules**
  - Hardware and operating system info
  - Handle the administration of MySQL over the Web
- **Asterisk debug logfiles (if enabled)**
  - Link to the FreePBX Channel on IRC
  - Link to http://freepbx.org/
- **NAT settings, select audio and video codecs, media & RTP, etc.**
- **Allow external systems to communicate directly with Asterisk**
- **Asterisk Command Line Interface**
  - Summary, Registries, Channels, Peers, Sip Info, Full Report, etc.
- **This is a phonebook that you can add users to.**
- **System Backup Setup, Scheduling, and Restore**
  - Register your custom destinations that point to custom dialplans
  - Facility to register any custom extensions
  - Distributed Universal Number lookup.
  - SSH session to the root of the IPBX Server.
  - Returns info, in HTML form, about the server’s PHP environment
  - This module helps administrators to configure phpagi.conf file.
  - Print PBX extensions and enabled feature codes
  - Messages or tones to send if the route is congested
  - This module points out weak passwords: the type, the name, the secret, and why it is weak.
  - Add Customer and contact info
  - This is a hosted service to record and create podcasts
  - Inventory of equip. serial numbers, models, locations, etc.
Maintenance

A2Billing Admin

A2Billing is a free component that offers the following services:

• Traditional Calling Card services – A2Billing can be configured to provide standard calling card services via traditional “dial through”, with PIN or CID authentication.

• Callback services, A2Billing supports a number of call-back methods including ANI, DID and web based call-backs.

• VoIP residential services – Customers can be issued with a softphone or hard phone and be billed for calls made via Voice over IP.

• VoIP wholesale termination - A2Billing and Asterisk can be used as a softswitch to terminate and bill large numbers of VoIP minutes from a number of sources such as resellers and distributors of your services.

• VoIP termination for Asterisk and FreePBX systems. – With the growth of IP based PBX systems, as well as the asterisk based systems, A2Billing can be used to provide services and billing to IP PBX resellers and Asterisk system integrators.

• DID termination and redirection. DID can be redirected to any SIP, IAX or PSTN destination with monthly charges and duration based charges.

It can be disabled or uninstalled by using the Module Admin application.
Config Edit

At this point your basic system should be working.

A number of .conf files may require editing to get asterisk to improve performance or for special configurations for individual requirements. The following website tells who owns what files in /etc/asterisk when FreePBX is installed: http://www.freepbx.org/configuration_files.

**NOTE:** Config files manual tweaks in this guide are meant for advanced users. They should only be used when the FreePBX does not offer the setting needed. If you have read this far, you are probably an advanced user. Unless you have a compelling reason to do so, users are advised against any manual config file tweaks.

**NOTE:** Config Edit only gives access to the FreePBX files. If you need to edit other files, you should use a proper editor and terminal access to the server. You can use your own SSH client or the Java SSH provided by FreePBX.

The use of tools like Config Edit lead to problems with novice users especially that they present configuration files that should never be edited and novice users are not aware of these restrictions. Experienced users typically have no issues using an editor to access configuration files and can further benefit from smart editors that color code Asterisk and other configuration files (such as vim) as well as aid in other syntax related requirements in config files.

**NOTE:** NEVER manually edit a configuration file that does not have the word "custom" in the file name. There are a few minor exceptions - you can modify sip.conf and iax.conf to allow additional codecs. You can modify sip_nat.conf to resolve one-way audio problems. You can modify zaptel.conf to allow pointing channels to extensions. Any file with the word "additional" in it will be rewritten by the system every time you click the Apply Configuration Changes in FreePBX, and most other configuration files (except for the ones with "custom" in their name) will be overwritten every time FreePBX itself is upgraded.

**Typically you must restart Asterisk every time you make changes to the config file.**

The configuration files (.conf) reside both in the /etc/asterisk directory and /etc directories. Configuration files in the /etc/asterisk are generally editable through Config Edit. To start editing the .conf files we need to select Tools tab, select the Config Edit selection from the dropdown menu as illustrated below.

Config Edit does not show all of the files that you can edit or view. If it does show the file and you need to look at it or edit it, it is better to use Config Edit than other tools, like the editor nano.

**NOTE:** If you type a long line, nano will wrap it. You can join the wrapped line back to the original by backspacing on the wrapped line. When you edit hosts file use: nano -w /etc/hosts

(-w disables wrapping.)

The FreePBX files are installed in /etc/asterisk when FreePBX is installed. The basic rule is that all files are owned and modified by FreePBX unless they end “_custom.conf”. There are a few exceptions to this rule but not many. If the file is owned by FreePBX you should find this statement at the top of the file making it clear that it is owned by FreePBX:

```
;--------------------------------------------------
; Do NOT edit this file as it is auto-generated by FreePBX. All modifications to
; this file must be done via the web gui. There are alternative files to make
; custom modifications, details at: http://freepbx.org/configuration_files
;```
This is the list of files as of version 2.4. Those owned by FreePBX will be in bold underline. If they become owned in a later version that version will be stated to the right of the file name.

agents.conf
alarmreceiver.conf
applications.conf
asterisk.conf

**backup.conf** - This file contains the crontab line(s) that will get executed for backup job scheduling. Cron is a time-based job scheduler in Unix-like computer operating systems. Cron is driven by a crontab, a configuration file that specifies shell commands to run periodically on a given schedule.

cdr_mysql.conf - If you want to use the userfield in the CDR reporting you will need to add this line to the file: userfield=1 then restart Freepbx by typing amportal restart
Default file should look like this:

```
; Note - if the database server is hosted on the same machine as the.
; asterisk server, you can achieve a local Unix socket connection by
; setting hostname=localhost
;
; port and sock are both optional parameters. If hostname is specified
; and is not "localhost", then cdr_mysql will attempt to connect to the
; port specified or use the default port. If hostname is not specified
; or if hostname is "localhost", then cdr_mysql will attempt to connect
; to the socket file specified by sock or otherwise use the default socket
; file.
;
[global]
hostname=localhost
dbname=asteriskcdrdb
password=amp109
user=asteriskuser
;port=3306
;sock=/tmp/mysql.sock
```

codecs.conf
dnsmgr.conf
dundi.conf
enum.conf
extconfig.conf

**extensions.conf** - If you need to modify existing code/context in extensions.conf please place your modifications in extensions_override_freepbx.conf as asterisk uses the code for the first context reference and ignores additional occurrences.

**extensions_additional.conf** - **DO NOT EDIT THIS FILE**, it is regenerated each and every time you apply changes.

If you need to expand on functionality of a section of code check to see if there is a include context line in the code (will end in _custom.conf) if so create that context in extensions_custom.conf and it will get called.

If you need to replace the functionality in extensions_additional.conf please place it in extensions_override_freepbx.conf but read the notes about this file first.
extensions_custom.conf - This is the file that you place all your custom contexts, and additional code enhancements to the FreePBX dial plan. This file will not be overwritten.

extensions_override_freesbx.conf - If extensions.conf (or extensions_additional.conf) has a context or macro that you NEED to modify, you place that code here as asterisk will only execute the first occurrences of that code and ignores other occurrences. This file will not be overwritten. Be very careful as replacing an existing piece of code this way is the fastest possible way to break your system. If you are doing this you should probably think about filing for a feature request or bug fix to get it addressed properly.

features.conf
features_applicationmap_additional.conf
features_applicationmap_custom.conf
features_featuremap_additional.conf
features_featuremap_custom.conf
features_general_additional.conf
features_general_custom.conf
globals_custom.conf
iax.conf
iax_additional.conf
iax_custom.conf
iax_custom_post.conf
iax_general_additional.conf
iax_general_custom.conf
iaxprov.conf
iax_registrations.conf
iax_registrations_custom.conf
indications.conf
localprefixes.conf
logger.conf
manager_additional.conf
manager.conf
manager_custom.conf
meetme.conf
meetme_additional.conf
mgcp.conf
modem.conf
modules.conf
musiconhold_additional.conf
musiconhold.conf
musiconhold_custom.conf
oss.conf
parking_additional.inc - Should no longer be used as parking was moved to features.
phone.conf
phpagi.conf
privacy.conf
queues.conf - Do not edit this file in any way. Anything you can think of putting in this file can be placed into one of the _custom.conf files where it will not get removed or replaced.

queues_additional.conf- Do not edit this file in any way. Anything you can think of putting in this file can be placed into one of the _custom.conf files where it will not get removed or replaced.

queues_custom.conf - This is the proper location for placing any of the context specific options and lines that you might need to add before the processing of the queues_additional.conf file for your queues setup.
queues_custom_general.conf - This is the proper location for placing any of the [general] context option lines that you might need to add to your queues setup.

queues_general_additional.conf - Do not edit this file in any way. Anything you can think of putting in this file can be placed into one of the _custom.conf files where it will not get removed or replaced.

queues_post_custom.conf - This is the proper location for placing any of the context specific options that you might need to add to the end queues setup. This is the file that allows you to add or remove values to those entries found in the auto-generated queue_additional.conf file. So for example you have a queue 79 that needs an additional parameter added. Create a context line: [79](+) then on the next line add the item(s) you need to add. To remove use (-) instead followed by the line(s) you want removed.

res_mysql.conf
rtp.conf
sip.conf - Do not edit this file in any way. Anything you can think of putting in this file can be placed into one of the _custom.conf files where it will not get removed or replaced. If you are looking to do nat'ing, see sip_general_custom.conf or if it is a legacy system sip_nat.conf. If you want to add additional setup parameters for your sip device see sip_custom_post.conf, etc. If you need to adjust sip jitter or something else it will be sip_general_custom.conf (if it is for the general context) or sip_custom.conf. If you do edit this file and place something new in it, it will get overwritten at some point and next time you restart your system you will suddenly wonder why things stopped working.

sip_general_additional.conf - This is where FreePBX places all of its general context settings. If you need to override one of these or add a new one please do so in sip_general_custom.conf.

sip_general_custom.conf - This is the proper location for placing any of the [general] context option lines that you might need to add to your setup. This is also the place to add those lines needed to enable the nat'ing of SIP when you go through a firewall.

Some of the required lines for nat'ing are externip=, nat=, localnet= (you can have more than one occurrence of this line), and optionally fromdomain=. The first three are needed to properly setup a box on protected network behind a firewall that is providing nat to a public IP. If you have a legacy system these lines might have been placed in sip_nat.conf in the past, if so that is ok as long as the lines only exist in one file and not both (or a big debugging mess will occur along with hair loss as you pull it out while tracking it all down). See sip_nat.conf for more info.

configurations with multiple subnets:
For those setups with internal networks that have multiple subnets you will need to add a localnet= line for each subnet that the phone system should have direct access to. If you don't do this the phone system will assume that phones on those other subnets are external and thus provide the External IP of the box in the SIP headers instead of the internal IP. This then becomes a routing problem for the phone as it should not be attempting to talk external IP of the internal box (most firewalls can not handle the looping back of IP traffic).

Example:
Server 192.168.1.2 on a 192.168.1.0/255.255.255.0 network
Phones inside the office are on the 192.168.2.0/255.255.255.0 subnet

Requires these two lines in the either sip_general_custom.conf or sip_nat.conf file
localnet=192.168.1.0/255.255.255.0
localnet=192.168.2.0/255.255.255.0

sip_nat.conf - This is the old common location for placing the lines needed to enable the nat'ing of SIP. The new preferred location is sip_general_custom.conf. If you move the lines from this file to sip_general_custom.conf please remove them from this file or you'll experience hair loss as you spend time debugging why things don't work as you expect.

sip_registrations.conf
General section registrations that are auto-generated by FreePBX.
sip_registrations_custom.conf – This is a custom file just in case there is ever a need to override a general registration that was auto-generated by FreePBX.

sip_custom.conf - This is the first file that is not under the general context. It allows you to define contexts that you need before the contexts that are auto-generated by FreePBX in sip_additional.conf.

**sip_additional.conf**

This is where FreePBX puts all sip extensions, sip trunks, etc. If you need to add a additional parameter to a extension, trunk, etc., see sip_custom_post.conf.

sip_custom_post.conf

This is the file that allows you to add/remove values to those entries found in the auto-generated sip_additional.conf file. So for example you have an extension 1000 that needs an additional parameter added. Create a context line: [1000](+) then on the next line add the item(s) you need to add. To remove use (-) instead followed by the line(s) you want removed.

**sip_notify.conf**

**skinny.conf**

**voicemail.conf**

This file is both editable by you and by FreePBX, so please be careful. The structure of this file is as follows:

```
[general]
#include vm_general.inc
#include vm_email.inc

[default]
```

Once you have configured a system with voicemail there will be values after the context [default]. These lines will be generated by FreePBX every time you add/edit/delete an extension.

If you want to customize the e-mail message that is sent out with a voice mail then edit the vm_email.inc file. If you need to edit the mail sending parameters edit the vm_general.inc file. 99% of the world needs to edit two lines in the vm_general.inc file at the initial build time.

The most common change to this file is to create a context called [zonemessages]. This context allows you to create time zones so that when you have extensions in multiple time zones it can date time stamp recorded messages properly for any given extension. If you create this context it should be placed after the second #include line and before the [default] line.

```
[general]
#include vm_general.inc
#include vm_email.inc

[zonemessages]
eastern =       America/New_York|'vm-received' q 'digits/at' IMp
central =       America/Chicago|'vm-received' q 'digits/at' IMp
mountain =      America/Denver|'vm-received' q 'digits/at' IMp
pacific =       America/Tijuana|'vm-received' q 'digits/at' IMp
eastern24 =     America/New_York|'vm-received' q 'digits/at' R
central24 =     America/Chicago|'vm-received' q 'digits/at' R
```

<table>
<thead>
<tr>
<th>Timezone</th>
<th>Country/Region</th>
<th>Mailbox</th>
<th>Options</th>
</tr>
</thead>
<tbody>
<tr>
<td>mountain24</td>
<td>America/Denver</td>
<td><code>vm-received</code></td>
<td>'digits/at' R</td>
</tr>
<tr>
<td>pacific24</td>
<td>America/Tijuana</td>
<td><code>vm-received</code></td>
<td>'digits/at' R</td>
</tr>
<tr>
<td>deutschland</td>
<td>Europe/Berlin</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' kM</td>
</tr>
<tr>
<td>england</td>
<td>Europe/London</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' R</td>
</tr>
<tr>
<td>germany</td>
<td>Europe/Berlin</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' kM</td>
</tr>
<tr>
<td>alberta</td>
<td>Canada/Mountain</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' HM</td>
</tr>
<tr>
<td>madison</td>
<td>Europe/Paris</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' R</td>
</tr>
<tr>
<td>madrid</td>
<td>Europe/Paris</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' R</td>
</tr>
<tr>
<td>sthlm</td>
<td>Europe/Stockholm</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' R</td>
</tr>
<tr>
<td>europa</td>
<td>Europe/Berlin</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' kM</td>
</tr>
<tr>
<td>italia</td>
<td>Europe/Rome</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' HMP</td>
</tr>
<tr>
<td>military</td>
<td>Zulu</td>
<td><code>vm-received</code></td>
<td>Q 'digits/at' H N 'hours' 'phonetic/z_p'</td>
</tr>
</tbody>
</table>

vm_email.inc - This file contains the e-mail subject line and message body for any voice mails that are e-mailed.

vm_general.inc - This file contains the e-mail / voice mail configuration parameters. The most common change to this file is to edit the `servermail=` line so that it is from a valid worldly e-mail address or any mail server that has spam and/or spoofing protection will reject the voice mail e-mails. Other common lines to edit are:

maxmessage= this is the max message limit,

maxmsg= limits the total number of messages allowed in a mailbox,

operator= if this is set to yes then when a person is leaving a message they can press 0 for the operator (or dial another extension).

zapata.conf
zapata-auto.conf
zapata_additional.conf
zapata_custom_chan_default.conf
On selecting **Config Edit** (marked with arrow), you may need to verify that you have the right to make changes to the configuration. The default username is **maint** and password is **password** unless you have changed it.

You will then see a new screen with a list of all the `.conf` files, which can be edited manually. **Scroll down the page to find the file that needs to be edited.**

**Examples**

These are two problems that may require an edit of the conf file.

**PROBLEM 1**) Need to connect zap channels to specific extensions. A catch all inbound route definition captures the call correctly.

**SOLUTION:** Change `context=from-pstn` to `context=from-zap tel` in the `zapata.conf` and `zapata-channels.conf`.

After that you set up the Zap Channel DID to the extension used by the outside world. Then set up your Inbound Routes to that DID number with the check in CID Priority Route. I also set the CID name prefix to 'Main Office' and of course the Destination Extension.

You must restart asterisk after doing so for this change to take affect. For this use PuTTY to do `amportal restart`
PROBLEM 2)

Voice mail does not detect when someone disconnects without pressing #. As a result most voice mails last three minutes for hang-uppers and even for short messages.

SOLUTION:

zapata.conf
Under [channels] add the following lines:
[channels]
busydetect=yes
busycount=6

You may also need to do this one using the Voicemail Admin option.

maxsilence=10 (How many seconds of silence before we end the recording)
silencethreshold=128 What we consider silence. When using the maxsilence setting, it is sometimes necessary to adjust the silence detection threshold to eliminate false triggering on background noise.

Silencethreshold allows the administrator to do just that. Higher numbers raise the threshold so that more background noise is needed to cause the silence detector to reset. When employing this setting, some experimentation will be necessary to find the best result. The lower the setting is, the more sensitive the trigger. The default silencethreshold value is 128.
### Sys Info

This module provides a quick snapshot of the System Vitals, Network Usage, Hardware Information, Memory Usage, and Mounted Filesystems. The Sys Info provides the following panels:

<table>
<thead>
<tr>
<th>System Vital</th>
</tr>
</thead>
<tbody>
<tr>
<td>Canonical Hostname: pbx.local</td>
</tr>
<tr>
<td>Listening IP: 205.242.138.50</td>
</tr>
<tr>
<td>Kernel Version: 2.6.18-92.1.6.el5 (SMP)</td>
</tr>
<tr>
<td>Distro Name: CentOS release 5.2 (final)</td>
</tr>
<tr>
<td>Uptime: 14 days 3 hours 54 minutes</td>
</tr>
<tr>
<td>Current Users: 0</td>
</tr>
<tr>
<td>Load Averages: 0.50 0.29 0.16</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Network Usage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device</td>
</tr>
<tr>
<td>----------</td>
</tr>
<tr>
<td>lo</td>
</tr>
<tr>
<td>eth0</td>
</tr>
<tr>
<td>stdin</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Hardware Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Processors: 1</td>
</tr>
<tr>
<td>Model: VIA C7 Processor 1500MHz</td>
</tr>
<tr>
<td>CPU Speed: 1.5 GHz</td>
</tr>
<tr>
<td>Cache Size: 128.90 KB</td>
</tr>
<tr>
<td>System: 2094.69</td>
</tr>
<tr>
<td>Busmaster: IDE</td>
</tr>
<tr>
<td>PCI Devices: - Ethernet controller: Digium, Inc. Unknown device 5005</td>
</tr>
<tr>
<td>- Ethernet controller: Realtek Semiconductor Co., Ltd. RTL8110SC/8169SC Gigabit Ethernet</td>
</tr>
<tr>
<td>- (5x) Host bridge: VIA Technologies, Inc. CN700/VPN000/FM800CC/Pro Host Bridge</td>
</tr>
<tr>
<td>- Host bridge: VIA Technologies, Inc. PT880 Host Bridge</td>
</tr>
<tr>
<td>- IDE interface: VIA Technologies, Inc. VIA VT6420 SATA RAID Controller</td>
</tr>
<tr>
<td>- IDE interface: VIA Technologies, Inc. VT8220/VT8237 IDE Controller</td>
</tr>
<tr>
<td>- ISA bridge: VIA Technologies, Inc. VT8237 ISA bridge [KT800/K8T800/K8T800 South]</td>
</tr>
<tr>
<td>- Multimedia audio controller: VIA Technologies, Inc. VT8233/A/8235/8237 AC97 Audio Controller</td>
</tr>
<tr>
<td>- PCI bridge: VIA Technologies, Inc. VT8237 PCI Bridge</td>
</tr>
<tr>
<td>- USB Controller: VIA Technologies, Inc. USB 2.0</td>
</tr>
<tr>
<td>- (4x) USB Controller: VIA Technologies, Inc. VT8200/VT8210/VT8200 UHCI USB 1.1 Controller</td>
</tr>
<tr>
<td>- VGA compatible controller: VIA Technologies, Inc. UniChrome Pro IGP</td>
</tr>
<tr>
<td>IDE Devices: - hda: Flash Card (Capacity: 3.92 GB)</td>
</tr>
<tr>
<td>SCSI Devices: none</td>
</tr>
<tr>
<td>USB Devices: none</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Memory Usage</th>
</tr>
</thead>
<tbody>
<tr>
<td>Type</td>
</tr>
<tr>
<td>-------------</td>
</tr>
<tr>
<td>Physical Memory</td>
</tr>
<tr>
<td>- Kernel + applications</td>
</tr>
<tr>
<td>- Buffers</td>
</tr>
<tr>
<td>- Cached</td>
</tr>
<tr>
<td>Disk Swap</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Mounted Filesystems</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mount</td>
</tr>
<tr>
<td>-------</td>
</tr>
<tr>
<td>/boot</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>/dev/shm</td>
</tr>
<tr>
<td>Totals</td>
</tr>
</tbody>
</table>
phpMyAdmin

phpMyAdmin is a tool written in PHP intended to handle the administration of MySQL over the Web. Currently it can create and drop databases, create/drop/alter tables and views, delete/edit/add fields, execute any SQL statement, manage keys on fields, manage privileges, and export data into various formats.
Support

Asterisk Logfiles

Whenever problem strikes, there is always a reason for it. To assist you in locating the problem you can see the Asterisk Logfiles, under Tools (which will give you the last 2000 entries in the log).

Asterisk Log Files

Display Asterisk Full debug log (last 2000 lines)

/var/log/asterisk/full - last 2000 lines

Redisplay Asterisk Full debug log (last 2000 lines)

```
```

Studying the activities will give you an indication of what has gone wrong.
Online Support

Start IRC

Allows you to contact the FreePBX channel on IRC. As IRC is an un-moderated international medium, AMP, FreePBX, Coalescent Systems, or any other party can not be held responsible for the actions or behaviour of other people on the network. When you connect to IRC, to assist in support, the IRC client will automatically send the following information to everyone in the #freePBX channel:

Your Linux Distribution: (Redhat CentOS release 5.2 (Final))
Your FreePBX version: (2.5.1)

If you do not want this information to be made public, please use another IRC client, or contact a commercial support provider

When you connect, you will be automatically be named ‘FreePBX’ and a random 4 digit number, e.g., FreePBX3486. If you wish to change this to your normal nickname, you can type /nick yournickname, and your nick will change. This is an ENGLISH ONLY support channel.

Online Resources

Takes you to http://www.freepbx.org/
FreePBX Support

FreePBX Support Takes you to http://www.freepbx.org/ also.

I found the documentation link on this page to be the most helpful.
http://www.freepbx.org/support/documentation
## System Administration

### Asterisk SIP Settings

**NAT Settings**

<table>
<thead>
<tr>
<th>NAT</th>
<th>yes</th>
<th>no</th>
<th>never</th>
<th>route</th>
</tr>
</thead>
</table>

**IP Configuration**

- Public IP
- Static IP
- Dynamic IP

**External IP**

- [External IP](#)

**Local Networks**

- [Local Networks](#)

**Add Local Network Field**

For networks with more than 1 LAN subnets, use this button for more fields. Blank fields will be removed upon submitting.

**NAT setting:**

- **yes** = Always ignore info and assume NAT
- **no** = Use NAT mode only according to RFC3581
- **never** = Never attempt NAT mode or RFC3581
- **route** = Assume NAT, don't send report

**IP Configuration** – Indicate whether the box has a public IP or requires NAT settings. Options are: Public IP, Static IP, or Dynamic IP.

**External IP** – This field is shown if you select Static IP. External Static IP or FQDN as seen on the WAN side of the router. (Asterisk: externip)

**Auto Configure** – If Static IP is set and you click Auto Configure button, it will fill in the External IP field with the current entry in sip_nat.conf.

**Dynamic Host** – This field is only shown if you select Dynamic IP. It is the External FQDN as seen on the WAN side of the router and updated dynamically, e.g. mydomain.dyndns.com. (Asterisk: externhost)

**Refresh Rate** – This field is only shown if you select Dynamic IP. How often to lookup and refresh the External Host FQDN, in seconds. (Asterisk: externrefresh)

**Local Networks** – Local network settings (Asterisk: localnet) in the form of IP/mask such as 192.168.1.0/255.255.255.0.

**Add Local Network Field** – For networks with more than 1 LAN subnets, use this button for more fields. Blank fields will be removed upon submitting.
Audio Codecs

**Codecs** – Check the desired codecs, all others will be disabled unless explicitly enabled in a device or trunks configuration. `ulaw` `alaw` `slin` `g726` `gsm` `g729` `ilbc` `g723` `g726aal2` `adpcm` `lpc10` `speex` `g722` `jpeg` `png`

**Non-Standard g726** – Asterisk: `g726nonstandard`. If the peer negotiates G726-32 audio, use AAL2 packing order instead of RFC3551 packing order (this is required for Sipura and Grandstream ATAs, among others). This is contrary to the RFC3551 specification; the peer should be negotiating AAL2-G726-32 instead. Select **Yes** or **No**

**T38 Pass-Through** – Asterisk: `t38pt_udptl`. This option enables T38 pass-through if enabled. This is for SIP channels that support sending/receiving T38 Fax codecs to pass the call. Asterisk cannot process this media. Select **Yes** or **No**

Video Codecs

**Video Support** – Check to enable and then choose allowed codecs. **Enabled** **Disabled**

**Max Bit Rate** — Maximum bit rate for video calls in kb/s

Video Codecs

**Video Support** – Check to enable and then choose allowed codecs. **Enabled** **Disabled**

**Max Bit Rate** — Maximum bit rate for video calls in kb/s
MEDIA & RTP Settings

Reinvite Behavior – This is the Asterisk: canreinvite parameter.

- **yes**: standard reinvites
- **no**: never
- **nonat**: An additional option is to allow media path redirection (reinvite) but only when the peer where the media is being sent is known to not be behind a NAT (as the RTP core can determine it based on the apparent IP address the media arrives from)
- **update**: use UPDATE for media path redirection, instead of INVITE. (yes = update + nonat)

RTP Timers:

- **Rrptimeout** – Terminate call if rrptimeout seconds of no RTP or RTCP activity on the audio channel when we’re not on hold. This is to be able to disconnect a call in the case of a phone disappearing from the net, like a power loss or someone tripping over a cable.
- **Rtpholdtimeout** – Terminate call if rtpholdtimeout seconds of no RTP or RTCP activity on the audio channel when we’re on hold (must be > rrptimeout).
- **Rtpkeepalive** – Send keepalives in the RTP stream to keep NAT open during periods where no RTP stream may be flowing (like on hold).

Notification & MWI

- **MWI Polling Freq** – Frequency in seconds to check if MWI state has changed and inform peers.
- **Notify Ringing** – Control whether subscriptions already INUSE get sent RINGING when another call is sent. This setting is useful when using BLF. Select **Yes** or **No**
- **Notify Hold** – Control whether subscriptions INUSE get sent ONHOLD when call is placed on hold. This setting is useful when using BLF. Select **Yes** or **No**
### Registration Settings

<table>
<thead>
<tr>
<th>Registrations</th>
<th>(registertimeout)</th>
<th>(registrationattempts)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration Times</td>
<td>(minexpiry)</td>
<td>(maxexpiry)</td>
</tr>
</tbody>
</table>

#### Registration Settings

**Registrations:**
- **registertimeout** – Retry registration attempts every \(\text{registertimeout}\) seconds until successful or until \(\text{registrationattempts}\) tries have been made.
- **registrationattempts** – Number of times to try and register before giving up. A value of 0 means keep trying forever. Normally this should be set to 0 so that Asterisk will continue to register until successful in the case of network or gateway outages.

**Registration Times:**
- **minexpiry** – Minimum length of registrations/subscriptions.
- **maxexpiry** – Maximum allowed time of incoming registrations
- **defaultexpiry** – Default length of incoming and outgoing registrations.

### Jitter Buffer Settings

<table>
<thead>
<tr>
<th>Jitter Buffer</th>
<th>Enabled</th>
<th>Disabled</th>
</tr>
</thead>
<tbody>
<tr>
<td>Force Jitter Buffer</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Implementation</td>
<td>Fixed</td>
<td>Adaptive</td>
</tr>
<tr>
<td>Jitter Buffer Logging</td>
<td>Enable</td>
<td>Disable</td>
</tr>
<tr>
<td>Jitter Buffer Size</td>
<td>200</td>
<td>(\text{jbmaxsize})</td>
</tr>
</tbody>
</table>

#### Jitter Buffer Settings

- **Jitter Buffer** – Enables the use of a jitter buffer on the receiving side of a SIP channel. An enabled jitter buffer will be used only if the sending side can create and the receiving side can not accept jitter. The SIP channel can accept jitter, thus a jitter buffer on the receive SIP side will be used only if it is forced and enabled. An example is if receiving from a jittery channel to voicemail, the jitter buffer will be used if enabled. However, it will not be used when sending to a SIP endpoint since they usually have their own jitter buffers. See \text{jbforce} to force it’s use always. Select Enabled or Disabled. Asterisk: \text{jbenable}.

- **Force Jitter Buffer** – Forces the use of a jitter buffer on the receive side of a SIP channel. Normally the jitter buffer will not be used if receiving a jittery channel but sending it off to another channel such as another SIP channel to an endpoint, since there is typically a jitter buffer at the far end. This will force the use of the jitter buffer before sending the stream on. This is not typically desired as it adds additional latency into the stream. Select Yes or No. Asterisk: \text{jbforce}.

- **Implementation** – Jitter buffer implementation, used on the receiving side of a SIP channel. Asterisk: \text{jbimpl}. Two implementations are currently available:
  - **fixed**: size always equals to \(\text{jbmaxsize}\);  
  - **adaptive**: with variable size (the new jitter buffer of IAX2).

- **Jitter Buffer Logging** – Enables jitter buffer frame logging. Asterisk: \text{jblog}. Select Enable or Disable.

- **Jitter Buffer Size**: \(\text{jbmaxsize}\) – Max length of the jitter buffer in milliseconds. Asterisk: \text{jbmaxsize}.

- **Jbresyncthreshold** – Jump in the frame timestamps over which the jitter buffer is resynchronized. Useful to improve the quality of the voice, with big jumps in/broken timestamps, usually sent from exotic devices and programs. It can be set to -1 to disable. Asterisk: \text{jbresyncthreshold}.
### Advanced General Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Language</td>
<td></td>
<td>Default Language for a channel, Asterisk: language</td>
</tr>
<tr>
<td>Default Context</td>
<td></td>
<td>Default context for incoming calls if not specified. FreePBX sets this to from-sip-external which is used in conjunction with the Allow Anonymous SIP calls. If you change this you will effect that behavior. It is recommended to leave this blank. Asterisk: context.</td>
</tr>
<tr>
<td>Bind Address</td>
<td></td>
<td>The IP address to bind to and listen for calls on the Bind Port. If set to 0.0.0.0, Asterisk will listen on all addresses. It is recommended to leave this blank. Asterisk: bindaddr.</td>
</tr>
<tr>
<td>Bind Port</td>
<td></td>
<td>Local incoming UDP Port that Asterisk will bind to and listen for SIP messages. The SIP standard is 5060 and in most cases this is what you want. It is recommended to leave this blank. Asterisk: bindport.</td>
</tr>
<tr>
<td>Allow SIP Guests</td>
<td>Yes/No</td>
<td>When set Asterisk will allow Guest SIP calls and send them to the Default SIP context. Turning this off will keep anonymous SIP calls from entering the system. However, the Allow Anonymous SIP calls from the General Settings section will not function. Allowing guest calls but rejecting the Anonymous SIP calls in the General Section will enable you to see the call attempts and debug incoming calls that may be miss-configured and appearing as guests. Asterisk: allowguest. Select Yes or No.</td>
</tr>
<tr>
<td>SRV Lookup</td>
<td>Enabled/Disabled</td>
<td>Enable Asterisk srvlookup. See current version of Asterisk for limitations on SRV functionality. Select Enabled or Disabled.</td>
</tr>
<tr>
<td>Other SIP Settings</td>
<td></td>
<td>You may set any other SIP settings not present here that are allowed to be configured in the General section of sip.conf. There will be no error checking against these settings so check them carefully. They should be entered as: [setting] = [value] in the boxes below. Click the Add Field box to add additional fields. Blank boxes will be deleted when submitted.</td>
</tr>
</tbody>
</table>
Asterisk API

Add Manager

Manager name: 
Manager secret: 
Deny: 
Permit: 

Rights

Read Write
system
call
log
verbose
command
agent
user
ALL

Manager name: Name of the manager without space.
Manager secret: Password for the manager.
Deny: If you want to deny many hosts or networks, use & char as separator.
Example: 192.168.1.0/255.255.255.0
10.0.0.0/255.0.0.0
Permit: If you want to permit many hosts or networks, use & char as separator. Look at deny example.

Rights

All manager API commands registered by various asterisk modules have a privilege group associated to it. In order for a manager API user to be able to issue a command, it has to have read or write privilege to the appropriate group.

Following is a list of commands grouped by privilege groups:

**system:** DBGet, DBPut, SIPpeers, SIPshowpeer

**call:** Hangup, Status, Setvar, Getvar, Redirect, Originate, ExtensionState, AbsoluteTimeout, MailboxStatus, MailboxCount, SetCDRUserField, Monitor, StopMonitor, ChangeMonitor

**log:** logs messages to the asterisk log

**verbose:** logs messages to the asterisk verbose log

**command:** allows access to command line interface.

**agent:** Agents, AgentLogoff, AgentCallbackLogin, QueueAdd, QueueRemove, QueuePause

**user:** generates a user log.

For more information on how to use Linux commands, see the Linux Commands.
Asterisk CLI

The Asterisk CLI module brings up a command line interface to Asterisk.

Asterisk CLI

Command: [Enter]

Asterisk CLI Commands

These are some of the available CLI commands that can be executed from the console when you run: `asterisk -r` (or `--rvv` depending on the level of verbosity you may want).

To stop PIAF – `amportal stop`
To start PIAF – `amportal start`

`! <command>`: Executes a given shell command
`abort halt`: Cancel a running halt
`add extension`: Add new extension into context
`add ignorepat`: Add new ignore pattern
`add indication`: Add the given indication to the country
`amportal start`: Stop PIAF and
`amportal stop`: Restart PIAF.
`debug channel`: Enable debugging on a channel
`dont include`: Remove a specified include from context
`help`: Display help list, or specific help on a command
`help-pbx`: The only command you have to remember
`include context`: Include context in other context
`load`: Load a dynamic module by name
`logger reload`: Reopen log files. Use after rotating the log files.
`no debug channel`: Disable debugging on a channel
`pri debug span`: Enables PRI debugging on a span
`pri intense debug span`: Enables REALLY INTENSE PRI debugging
`pri no debug span`: Disables PRI debugging on a span
`remove extension`: Remove a specified extension
`remove ignorepat`: Remove ignore pattern from context
`remove indication`: Remove the given indication from the country
`save dialplan`: Overwrites your current `extensions.conf` file with an exported version based on the current state of the dialplan. A backup copy of your old `extensions.conf` is not saved. The initial values of global variables defined in the `[globals]` category retain their previous initial values; the current values of global variables are not written into the new `extensions.conf`. Using “save dialplan” will result in losing any comments in your current
extensions.conf.

set verbose: Set level of verboseness

show agents: Show status of agents

show applications: Shows registered applications

show application: Describe a specific application

show channel: Display information on a specific channel

show channels: Display information on channels

show codecs: Display information on codecs

show conferences: Show status of conferences

show dialplan: Show dialplan

show image formats: Displays image formats

show indications: Show a list of all country/indications

show locals: Show status of local channels

show manager command: Show manager commands

show manager connect: Show connected manager users

show parkedcalls: Lists parked calls

show queues: Show status of queues

show switches: Show alternative switches

show translation: Display translation matrix

show voicemail users: List defined voicemail boxes

show voicemail zones: List zone message formats

soft hangup: Request a hangup on a given channel

For more information on how to use Linux commands, see the Linux Commands.
Asterisk Info

Select the Tools tab and click on the Asterisk Info option.

When presented with the info page, select Full Report in the lower right.

You will then be presented with the following screen.
Scroll down to the Subscriptions section and ensure that you are registered to your trunks. You will only be able to make and receive external calls if your extensions are registered.

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

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

This module allows you to add, edit, and delete Phonebook entries. You can export the file to CSV format.

Add or replace entry

<table>
<thead>
<tr>
<th>Name:</th>
<th>Number:</th>
<th>Speed dial code:</th>
<th>Set Speed Dial?</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>✔</td>
</tr>
</tbody>
</table>

Import from CSV

File: Browse...

Import from CSV

File: Import a CSV File formatted as follows:

"Name";Number;Speeddial
Names should be enclosed by '"' and fields separated by ';'

Example:

"John Doe";12345678;123
Backup & Restore

You can configure a regular backup schedule to ensure that you have a copy of your Asterisk and FreePBX configuration, voicemail and CDR records. You can also restore a previous backup, in case of data loss or a major configuration fault. Backups are stored on the file system at /var/lib/asterisk/backups. You should make a point of making an offline copy of important backups.

### Add Backup Schedule

#### System Backup

- **Delete Backup Schedule**: Daily

  - **Schedule Name**: Give this backup a friendly name (e.g. "Daily" or "Voicemail") to accurately identify what you're backing up. This will make future restores easier.
  - **VoiceMail**: Backup the System VoiceMail Boxes. **CAUTION**: Could result in large file.
  - **System Recordings**: Backup the System Recordings (AutoAttendant, Music on Hold, System Recordings)
  - **System Configuration**: Enable this option to backup your Asterisk and FreePBX configuration data, including the MySQL and Asterisk databases. We recommend this be enabled for all backups.
  - **CDR**: Backup the System Call Detail Records (HTML and Database).
  - **Operator Panel**: Backup the Flash Operator Panel (HTML and Database):

#### Run Schedule

- **Run Schedule Below**: Options are: Daily (at Midnight), Weekly (on Sunday at Midnight), Monthly (on the 1st at Midnight) or Yearly (on the 1st January at Midnight).

- **Run Backup**: Options are:
  - Follow Schedule Below, Now, Daily (at midnight), Weekly (on Sunday), Monthly (on the 1st), or Yearly (on 1st Jan).
Restore From Backup

System Restore

Clicking on the Restore from Backup button will list all the backups that are currently on your system (located at /var/lib/asterisk/backups). Click on the backup you wish to restore.

In this example the backup name is Daily. Click on that to bring up the next page.

System Restore

This will delete all of these backups.

Clicking on any of these backups brings up more options:

- **System Restore**
  - Delete File Set
  - Restore Entire Backup Set
  - Restore VoiceMail Files
  - Restore System Recordings Files
  - Restore System Configuration
  - Restore Operator Panel
  - Restore Call Detail Report

System Restore

- **Delete File Set** - Delete this backup set.
- **Restore Entire Backup Set** - Restore your Complete Backup set overwriting all files.
- **Restore VoiceMail Files** - Restore your Voicemail files from this backup set. **NOTE!** This will delete any voicemail currently in the voicemail boxes.
- **Restore System Recordings Files** - Restore your system Voice Recordings including AutoAttendant files from this backup set. **NOTE!** This will **OVERWRITE** any voice recordings currently on the system. It will NOT delete new files not currently in the backup set.
- **Restore System Configuration** - Restore your system configuration from this backup set. **NOTE!** This will **OVERWRITE** any System changes you have made since this backup... ALL items will be reset to what they were at the time of this backup set.
- **Restore Operator Panel** - Restore the Operator Panel from this backup set. **NOTE!** This will **OVERWRITE** any Operator Panel Changes you have made since this backup... ALL Items will be reset to what they were at the time of this backup set.
- **Restore Call Detail Report** - Restore the Call Detail Records from this backup set. **NOTE!** This will **DELETE ALL CALL RECORDS** that have been saved since this backup set.
Delete Backup Set

This is the procedure to delete backup sets to create more disk room.

Under the Tools menu tab, click on Back & Restore.

Click on Restore from Backup.

System Restore

- DELETE ALL THE DATA IN THIS SET

  - 20090811 00 00 01.tar.gz
  - 20090810 00 00 02.tar.gz
  - 20090809 00 00 01.tar.gz
  - 20090808 00 00 01.tar.gz
  - 20090807 00 00 01.tar.gz
  - 20090806 00 00 01.tar.gz
  - 20090805 00 00 01.tar.gz
  - 20090804 00 00 01.tar.gz
  - 20090803 00 00 01.tar.gz
  - 20090802 00 00 01.tar.gz
  - 20090801 00 00 01.tar.gz
  - 20090731 00 00 01.tar.gz
  - 20090730 00 00 01.tar.gz
  - 20090729 00 00 02.tar.gz
  - 20090728 00 00 01.tar.gz

Clicking on any of these backups brings up more options:

- System Restore
  - Delete File Set
  - Restore Entire Backup Set
  - Restore VoiceMail Files
  - Restore System Recordings Files
  - Restore System Configuration
  - Restore Operator Panel
  - Restore Call Detail Report

Choose Delete File Set.

The pop-up will come up:

![Message from webpage](image)

Are you sure you want to delete this File Set?

Click the OK button.

Potential Security Breach

You are attempting to modify settings from a URL that does not appear to have come from a FreePBX page link or button. This can occur if you manually typed in the URL below. This action has been blocked because the HTTP_REFERER does not match your current SERVER. If you require this access, you can set CHECKREFERER=false in amportal.conf to disable this security check.

The suspect URL is listed below. If this action is intended, you can click this link and your action will be processed. Do not proceed with this if you did not intend to execute this command as it may result in changes to your configuration.


Just click on the link and it the data set will be deleted forever.
Custom Destinations

Add Custom Destination

Custom Destination: This is the Custom Destination to be published. It should be formatted exactly as you would put it in a goto statement, with context, extension, priority all included. An example might look like:

mycustom-app,s,1

Destination Quick Pick Choose un-identified destinations on your system to add to the Custom Destination Registry. This will insert the chosen entry into the Custom Destination box above. (Pick destination)

Description: Brief Description that will be published to modules when showing destinations. Example: My Weather App

Notes: More detailed notes about this destination to help document it. This field is not used elsewhere.

At the bottom of the page, click on Submit Changes. Then select Apply Configuration Changes and Continue with reload to apply any changes / updates.
Custom Extensions

Add Custom Extension

Custom Extension: This is the Extension or Feature Code you are using in your dialplan that you want the FreePBX Extension Registry to be aware of.

Description: Brief description that will be published in the Extension Registry about this extension

Notes: More detailed notes about this extension to help document it. This field is not used elsewhere.

At the bottom of the page click on Submit Changes. At the top of the page click Apply Configuration Changes and then Continue with reload to apply any changes / updates.

Custom Extensions provides you with a facility to register any custom extensions or feature codes that you have created in a custom file and FreePBX doesn’t otherwise know about them. This allows the Extension Registry to be aware of your own extensions so that it can detect conflicts or report back information about your custom extensions to other modules that may make use of the information. You should not put extensions that you create in the Misc Apps Module as those are not custom.
DUNDi Lookup

DUNDi Lookup and Extension Registry Proxy

DUNDi Lookup

Lookup Number: [ ] Lookup

This module provides a proxy for the extension registry feature in FreePBX. If you have a DUNDi trunk configured in FreePBX to other branch offices, and a route defined to access it, then this module will proxy and check for extension number duplication in other branch offices when creating new extensions, ringgroups or any other extension that can be dialed.

The module does not consider the outbound dialing rules. It simply checks the DUNDi cloud down all configured DUNDi routes, and if another system indicates they have that extension, then it will consider this a conflict and report it back as such since a change in routing rules would easily expose this conflict.

As a secondary function, the module allows you to easily check for numbers present within the configured DUNDi routes in a simple GUI page.
Java SSH

If you need to edit a file, you should use a proper editor and terminal access to the server. You can use your own SSH client or the javassh provided by FreePBX.

The use of tools like config edit lead to problems with novice users especially that they present configuration files that should never be edited and novice users are not aware of these restrictions. Experienced users typically have no issues using an editor to access configuration files and can further benefit from smart editors that color code Asterisk and other configuration files (such as vim) as well as aid in other syntax related requirements in config files.

Java SSH

When you click on the Java SSH button, it should bring up the mindTerm splash screen with your PBX Server entered for you.

If necessary re-enter the maint user and password. Once the applet has opened enter the IP address of your PBX.

You will then connect to the FreePBX server. You can then login as 'root' or 'asterisk', or whatever user you have created.
**PHP Info**

PHPinfo is a function that returns information, in HTML form, about the PHP environment on your server. It outputs a large amount of information about the current state of PHP. This includes information about PHP compilation options and extensions, the PHP version, server information and environment (if compiled as a module), the PHP environment, OS version information, paths, master and local values of configuration options, HTTP headers, and the PHP License.

Because every system is setup differently, `phpinfo()` is commonly used to check configuration settings and for available predefined variables on a given system.

`phpinfo()` is also a valuable debugging tool as it contains all EGPCS (Environment, GET, POST, Cookie, Server) data.

See [http://us2.php.net/phpinfo](http://us2.php.net/phpinfo) for more information.
**PHPAGI Config**

The PHPAGI is a PHP class for the Asterisk Gateway Interface. This module helps administrators to configure phpagi.conf file. This module lets you configure all the PHP-AGI parameters including debug, errors mail reporting, fast-agi config, text to speech engine and more.

### Main config:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Debug</td>
<td>false</td>
</tr>
<tr>
<td>Error handler</td>
<td>false</td>
</tr>
<tr>
<td>Mail errors to</td>
<td><a href="mailto:admin@example.com">admin@example.com</a></td>
</tr>
<tr>
<td>Hostname of the server</td>
<td>freepbx.example.com</td>
</tr>
<tr>
<td>Temporary directory</td>
<td>/tmp</td>
</tr>
</tbody>
</table>

### Festival config:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Path to text2wave</td>
<td>/usr/bin/text2wave</td>
</tr>
</tbody>
</table>

### Asterisk API settings:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Server</td>
<td>localhost</td>
</tr>
<tr>
<td>Port</td>
<td>5038</td>
</tr>
<tr>
<td>Choose Manager</td>
<td></td>
</tr>
</tbody>
</table>

### Fast AGI config:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>setuid</td>
<td>false</td>
</tr>
<tr>
<td>Basedir</td>
<td>/var/lib/asterisk/agi-bin</td>
</tr>
</tbody>
</table>

### Cepstral config:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Swift path</td>
<td>opt/swift/bin/swift</td>
</tr>
<tr>
<td>Cepstral voice</td>
<td>David</td>
</tr>
</tbody>
</table>

**Submit Changes**

---

**Main config:**

- **Debug**: Set true if you want to enable PHPAGI debugging.
- **Error handler**: Set true if you want to use the internal error handler.
- **Mail errors to**: Email where the errors will be sent.
- **Hostname of the server**: Hostname of this server.
- **Temporary directory**: Temporary directory for storing temporary output.

**Festival config:**

- Festival is a general multi-lingual speech synthesis system developed at CSTR. It offers a full text to speech system with various APIs
- **Path to text2wave**: Path to text2wave binary.

**Asterisk API settings:**

- **Server**: Server to connect to.
- **Port**: Port to connect to manager.
- **Choose Manager**: Choose the user that PHPAGI will use to connect the Asterisk API.

**Fast AGI config:**

- **setuid**: Drop privileges to owner of script.
- **Basedir**: Path to AGI scripts folder.

**Cepstral config:**

- **Swift path**: Path to cepstral TTS binary.
- **Cepstral voice**: TTS Voice used.
Print Extensions

This module will print a list of users and extensions. It is handy for creating a company directory listing quickly.

**Printer Friendly Page**

**PBX Extension Layout**

<table>
<thead>
<tr>
<th>Name</th>
<th>Extension</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Ring Groups</strong></td>
<td></td>
</tr>
<tr>
<td>Ring All</td>
<td>200</td>
</tr>
<tr>
<td><strong>VoiceMail Blasting</strong></td>
<td></td>
</tr>
<tr>
<td>Olivus</td>
<td>500</td>
</tr>
<tr>
<td><strong>Extensions</strong></td>
<td></td>
</tr>
<tr>
<td>Marc Hall</td>
<td>204</td>
</tr>
<tr>
<td>Marty Tremey</td>
<td>208</td>
</tr>
<tr>
<td>Cliff Anderson</td>
<td>212</td>
</tr>
</tbody>
</table>

**Feature Code Admin**

- Blacklist a number: *30
- Blacklist the last caller: *32
- Remove a number from the blacklist: *31
- Call Forward All Activate: *72
- Call Forward All Deactivate: *73
- Call Forward All Prompting Deactivate: *74
- Call Forward Busy Activate: *90
- Call Forward Busy Deactivate: *91
- Call Forward Busy Prompting Deactivate: *92
- Call Forward No Answer/Unavailable Activate: *52
- Call Forward No Answer/Unavailable Deactivate: *53
- Call Waiting Activate: *70
Route Congestion Messages

This panel gives you an option to play either a tone or message if the route is congested. Consider a message instructing callers to find an alternative means of calling emergency services such as a cell phone or alarm system panel.

Route Congestion Messages

**Congested Route Options**

**Standard Routes**

<table>
<thead>
<tr>
<th>Message or Tone</th>
<th>Default Message</th>
</tr>
</thead>
</table>

Select a message or tone to be played if no trunks are available. You can select the Default Message, Congestion Tones, or any of the System Messages.

**Intra-Company Routes**

<table>
<thead>
<tr>
<th>Message or Tone</th>
<th>Default Message</th>
</tr>
</thead>
</table>

Select a message or tone to be played if no trunks are available. You can select the Default Message, Congestion Tones, or any of the System Messages. Used on routes marked as intra-company only.

**Emergency Routes**

<table>
<thead>
<tr>
<th>Message or Tone</th>
<th>Default Message</th>
</tr>
</thead>
</table>

Select a message or tone to be played if no trunks are available. You can select the Default Message, Congestion Tones, or any of the System Messages. Used on all emergency routes.

At the bottom of the page click on **Submit Changes**. At the top of the page click **Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
Weak Password Detection

This module finds any passwords that are weak and points them out to you. You should always use strong passwords to prevent thieves from using your PBX to make outgoing calls. A strong password is one that is difficult to detect by both humans and computer programs, effectively protecting data from unauthorized access. A strong password consists of at least six characters (and the more characters, the stronger the password) that are a combination of letters, numbers and symbols (@, #, $, %, etc.) if allowed. Passwords are typically case-sensitive, so a strong password contains letters in both uppercase and lowercase. Strong passwords also do not contain words that can be found in a dictionary or parts of the user’s own name.

<table>
<thead>
<tr>
<th>Type</th>
<th>Name</th>
<th>Secret</th>
<th>Message</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>No weak secrets detected on this system</td>
</tr>
</tbody>
</table>

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FreePBX 2.6.1 is licensed under GPL
Third Party Add-on

Customer DB

Add a user

Add Customer

Name
Address 1
Address 2
City
State
Zip/Post Code
☐ Sip Account
☐ DID Number
Device
Serial
IP Address
Account
Email
Username
Password
Submit Changes

At the bottom of the page, click on Submit Changes. At the top of the page, click Apply Configuration Changes and then Continue with reload to apply any changes / updates.
**Gabcast**

Gabcast is a hosted service that lets you record your phone or VoIP to create podcasts and then post them to your blog or website. Gabcast is a social broadcasting platform that offers virtual communities, individuals, and organizations an easy way to create and distribute audio content.

You must have a Gabcast account & channel to use this feature. Visit www.gabcast.com to sign up. It's a free service!

This Gabcast module allows you to:

- Link extensions to Gabcast channels. It creates a feature code, which defaults to *422 'gab' - you can change this in Feature Code Admin, which allows you to log directly into your Gabcast account. This is ideal for personal podcasting!

- Define a Gabcast channel as a Destination for other modules. For example, you can direct a DID or IVR menu option directly to Gabcast. This is ideal for group and public podcasting!

The latest episodes across all channels:

Feed URL: [http://www.gabcast.com/casts/feeds/latest.xml](http://www.gabcast.com/casts/feeds/latest.xml)

Click on the Add Gabcast Channel button.

**Gabcast Configuration**

- Gabcast Channel Number:
- Gabcast Channel Password:
- Link to Extension/User Number:

At the bottom of the page, click on Submit Changes. At the top of the page click Apply Configuration Changes and then Continue with reload to apply any changes / updates.
## Inventory

A place to keep the inventory of your phones.

<table>
<thead>
<tr>
<th>Employee #</th>
<th>Employee #</th>
<th>Employee Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>Employee Name</td>
<td>Employee Name</td>
<td>The Name of the Employee</td>
</tr>
<tr>
<td>Building Located</td>
<td>Building Located</td>
<td>Building where the phone is located</td>
</tr>
<tr>
<td>Floor #</td>
<td>Floor #</td>
<td>Floor phone is on</td>
</tr>
<tr>
<td>Room #</td>
<td>Room #</td>
<td>Room phone is in</td>
</tr>
<tr>
<td>Floor Section #</td>
<td>Floor Section #</td>
<td>Floor Section the phone is in</td>
</tr>
<tr>
<td>Cubicle #</td>
<td>Cubicle #</td>
<td>Cubicle phone is in</td>
</tr>
<tr>
<td>Desk #</td>
<td>Desk #</td>
<td>Desk Number phone is on</td>
</tr>
<tr>
<td>Extension #</td>
<td>Extension #</td>
<td>Extension Assigned to the phone</td>
</tr>
<tr>
<td>Phone UserName</td>
<td>Phone User</td>
<td>Name Phone Admin Username</td>
</tr>
<tr>
<td>Phone Password</td>
<td>Phone Password</td>
<td>Phone Admin Password</td>
</tr>
<tr>
<td>MAC Address</td>
<td>MAC Address</td>
<td>MAC Address of the phone</td>
</tr>
<tr>
<td>Serial #</td>
<td>Serial #</td>
<td>Serial Number of the phone</td>
</tr>
<tr>
<td>Phone/Device</td>
<td>Phone/Device</td>
<td>Example...Grandstream GXP2000</td>
</tr>
<tr>
<td>Distributed Date</td>
<td>Distributed Date</td>
<td>Distribution Date</td>
</tr>
<tr>
<td>IP Address</td>
<td>IP Address</td>
<td>IP Address Assigned If not DHCP</td>
</tr>
<tr>
<td>PBX Box Name</td>
<td>PBX Box Name</td>
<td>PBX Box Name</td>
</tr>
<tr>
<td>Extra Info</td>
<td>Extra Info</td>
<td>Extra Information</td>
</tr>
</tbody>
</table>

At the bottom of the page, click on **Submit Changes**. At the top of the page click **Apply Configuration Changes** and then **Continue with reload** to apply any changes / updates.
**Dynamic UI Menus**

Once the PBX has an IP address accessible from your network, you can connect to it using your browser at: `http://ipaddress/` (e.g. `http://192.168.1.100`) to configure PiaF.

You will be presented with the PiaF initial User Mode splash screen as illustrated below.

![Initial Welcome Screen](image)

**Initial Welcome Screen (User Mode)**

This screen (once PiaF has been properly set-up) enables user’s access to the Voicemail & Recordings and the Flash Operator Panel (FOP).

Clicking on any menu will request a password the first time that you access it after a restart of PC accessing the PBX Server.
Voicemail & Recordings

Login

Login: 
Password: 
Submit

Remember Password

Use your Voicemail Mailbox and Password
This is the same password used for the phone

For password maintenance or assistance, contact your Phone System Administrator.

This is the same password used for your phone and voicemail.

Voicemail for Cliff Anderson (212)

These features are self-explanatory.
Flash Operator Panel

From the Main Menu, you can click on the Flash Operator Panel icon.

You can also access FOP from FreePBX Admin, by clicking Panel.

The Timeout is for transferred calls. The call ends after xx minutes.

When you are in the PIAF Flash Operator Panel there is a lock icon: Open Security Code Input Box. When you click on it, it says "Please enter the Security Code". The default password is passw0rd.

<table>
<thead>
<tr>
<th>The following information are displayed on FOP</th>
<th>Functions you can perform on FOP</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Which extensions are busy, ringing or available</td>
<td>• Hang-up a channel</td>
</tr>
<tr>
<td>• Who is talking and to whom (CLID, context, priority)</td>
<td>• Using drag-&amp;-drop to transfer a call</td>
</tr>
<tr>
<td>• SIP and IAX registration status and reachability</td>
<td>• Initiate calls by drag-&amp;-drop</td>
</tr>
<tr>
<td>• MeetMe room status (number of participants)</td>
<td>• Barge in on a call using drag-&amp;-drop</td>
</tr>
<tr>
<td>• Queue status (number of users waiting)</td>
<td>• Set the caller id when transferring or originating a call</td>
</tr>
<tr>
<td>• Message Waiting Indicator and count</td>
<td>• Automatically pop up web page with customer details</td>
</tr>
<tr>
<td>• Parked channels</td>
<td>• Click-to-Dial from a web page</td>
</tr>
<tr>
<td>• Logged in Agents</td>
<td>• Mute/Unmute meet-me participants</td>
</tr>
</tbody>
</table>
Admin

If you have the password and click on the Admin button, you will have access to the Administrator options:

- **FreePBX Administration** – To manage the PBX through FreePBX
- **Linux Webmin** – This is a system utility that allows you to maintain the virtually the entire system. It is best to restrict this to the System Administrator and Root users only.
- **Menu Configuration** – This determines what the User menu shows.

FreePBX Administration brings up the FreePBX Setup and Tools menus as covered previously.
Menu Configuration

If you remove the check from **Enable End-User Menu**, that menu will not be available to users.

If you click on **Update** button without highlighting Voicemail & Recordings or Flash Operator Panel, the User panel will not have these buttons.

Any options that are removed or commented out from the `/var/www/html/welcome/.htindex.cfg` will not show either.

Click on the **Done** button to go back to the Dynamic UI.

Security Alert is a link to Nerd Vittles. This is a very useful blog.

This is a link to http://pbxinaflash.net/

There is also a link to http://pbxinaflash.com/docs/

And most useful of all is the PiaF Forum: [http://pbxinaflash.com/forum/](http://pbxinaflash.com/forum/)
Linux Webmin

Webmin's job is to provide a graphic user interface (GUI) that configures all of the text-based Non Asterisk configuration files. Webmin is the Swiss army knife of Linux. It provides TOTAL access to your system through a web interface. Search Nerd Vittles for webmin if you want more information. Be very careful if you decide to enable it on the public Internet.

Before using Webmin, you need to set up a username and password for access. From the Linux prompt while logged in as root, type the following command where admin is the username you wish to set up and password is the password you've chosen for the administrator account.

**WARNING: Don't use admin and password as your username and password for Webmin.**

```
/usr/libexec/webmin/changepass.pl /etc/webmin admin password
```

To access Webmin on your private network, bring up the PBX in a Flash – Dynamic UI menus in your favorite browser.

Use the PBX root username and password to gain access over port 9001.

To stop Webmin: `/etc/webmin/stop`.
To start Webmin: `/etc/webmin/start`.

For complete documentation, go to: [http://www.webmin.com/intro.html](http://www.webmin.com/intro.html) and [http://doxfer.com/Webmin/Modules](http://doxfer.com/Webmin/Modules)

This is Webmin's main menu screen. It shows the current status of the server and resources used. In the following links, you'll see the powerful tool that Webmin is. Each list item shows the pull-down menus that are available.
Webmin Module

This is the area where you would configure Webmin itself, backup and restore the Webmin configuration and create/modify Webmin users and more.

System Module

The System module is where you would configure the Linux operating system, backup and restore it and create/modify users and more.
Servers Module

The Server module is where you would install and configure the various servers that are needed by Asterisk. Some examples are the mySQL server, web server, ftp server, email server, etc.

- Apache Webserver
- BIND DNS Server
- CVS Server
- DHCP Server
- Dovecot IMAP/POP3 Server
- Fetchmail Mail Retrieval
- Fox FTP Proxy
- Jabber IM Server
- Majordome List Manager
- MySQL Database Server
- OpenSLP Server
- Postfix Configuration
- PostgreSQL Database Server
- ProFTPD Server
- Procmail Mail Filter
- QMail Configuration
- Read User Mail
- SSH Server
- Samba Windows File Sharing
- Sendmail Configuration
- SpamAssassin Mail Filter
- Squid Analysis Report Generator
- Squid Proxy Server
- WU-FTP Server
- Webalizer Logfile Analysis

Networking Module

This is the section that deals with networking issues such as TCP/IP settings, PPP, ADSL, etc.

- Networking
- ADSL Client
- Bandwidth Monitoring
- IPSec VPN Configuration
- Kerberos
- Linux Firewall
- NFS Exports
- NIS Client and Server
- Network Configuration
- PPP Dialin Server
- PPP Dialup Client
- FTP VPN Client
- FTP VPN Server
- SSL Tunnels
- Shoreline Firewall idmapd daemon
Hardware Module

The Hardware module is interested in anything to do with the hard-drives and their management including the voicemail server. Here you’ll find CD burner data, RAID info, LVM info, Partitioning, etc...

- Hardware
- CD Burner
- GRUB Boot Loader
- Linux RAID
- Logical Volume Management
- Partitions on Local Disks
- Printer Administration
- SMART Drive Status
- System Time
- Voicemail Server

Cluster Module

PBX in a Flash is designed to be scalable. The Cluster module gives you the controls to scale up your servers. Here’s the heart of creating and managing those clusters.

- Cluster
  - Cluster Change Passwords
  - Cluster Copy Files
  - Cluster Cron Jobs
  - Cluster Shell Commands
  - Cluster Software Packages
  - Cluster Uservmin Servers
  - Cluster Users and Groups
  - Cluster Webmin Servers
  - Configuration Engine
  - Heartbeat Monitor

Other Module

The Other Module takes on the job of detailing the other stuff that is required to manage a server but just doesn’t fit into the previous categories. You can upload and download files to the server. There is an SSH/Telnet client for connecting to your server through a command line interface, etc.

- Others
  - Command Shell
  - Custom Commands
  - File Manager
  - HTTP Tunnel
  - PHP Configuration
  - Perl Modules
  - Protected Web Directories
  - SSH/Telnet Login
  - System and Server Status
  - Upload and Download

File Manager

Click on the Others menu to open it up. Click on File Manager.

Upload and Download

Click on the Others menu to open it up. Click on Upload and Download. It is fairly intuitive from there.

NOTE: If you do not know the password, but have the root password, use passwd-webmin for users needing Webmin access to your server.
File Manager

Click on the **Others** menu to open it up. Click on **File Manager** and the java splash screen will start.

When the java completes it brings up the file manager:

![Java File Manager](image)

From here you can perform all of the functions on the menu. You can click on files to play them as long as you have the correct player installed on your PC.
Upload and Download

Click on the **Others** menu to open it up. Click on **Upload and Download**.

**Upload and Download**

This form allows you to download files or web pages from HTTP or FTP URLs to the system running Webmin. The download can be done immediately, or scheduled for some time in the future.

**Download files to server from URLs**

- **URLs to download**: [Field]
- **Download to file or directory**: [Field]
- **Create directory if needed?**: [Checkbox]
- **Owned by user**: [Field]
- **Owned by group**: [Field]
- **Download mode**: [Radio buttons]
  - Immediately, and show progress
  - In background, at date: [Calendar], and time [Time]
- **Send email when downloads are done?**: [Radio buttons]
  - No
  - Yes, to address: [Field]

**Download URLs**

**Upload and Download**

This page allows you to upload one or more files from the PC on which your web browser runs to the system running Webmin.

**Upload files to server**

- **Files to upload**: [Field]
- **File or directory to upload to**: [Field]
- **Create directory if needed?**: [Checkbox]
- **Owned by user**: [Field]
- **Owned by group**: [Field]
- **Extract ZIP or TAR files?**: [Radio buttons]
  - Yes, then delete
  - Yes
  - No
- **Send email when uploads are done?**: [Radio buttons]
  - No
  - Yes, to address: [Field]

**Upload**
Passwords

There are several places where information is protected by names and passwords. The security of your system is up to you. If your system is open to the Internet and your security is set too low, someone could use your system to make long distance calls at your expense.

Using default FreePBX admin security alone will not protect your system from a web attack and may compromise root access to your entire server, if you do not change the password, before connecting the PBX to the Internet. For this reason, we recommend that you log in as root and immediately run passwd-master. This establishes Apache .htaccess security on your FreePBX web interface. After running this conversion utility, you can only log into the FreePBX admin interface with the username maint and the password which you establish when you run the utility.

For more information on how to use Linux commands, see the Linux Commands.

```
passwd-master
```

To change the main passwords, run `passwd-master` from the command line.

This does the following

1) Changes FreePBX to authtype = none in amportal.conf,

2) Sets up .htaccess on the admin directory which contains FreePBX

3) Sets the wwwadmin, and MeetMe passwords to the same one as maint.

After running the command, to access all areas including FreePBX, the username is maint with the password of whatever you set during the passwd-master script. Maint gets you to admin (FreePBX) maint, FOP, MeetMe.

Other passwords can be set in your system, similarly, see below.

**PASSWORD - LINUX**

If you plug a monitor and keyboard into the PBX and power it up, at the “login as” prompt, the username is root. The default password is 123456. To reset the root password, use the command line `passwd`

**PASSWORD - THE PBX IN A FLASH BROWSER**

When you put the IP Address of the PBX in your browser it brings up the PBX in a Flash Dynamic UI: Menus - users menu with three icons: Voicemail & Recordings, Flash Operator Panel, and MeetMe Conference. The default password is passw0rd.
User Passwords

PASSWORD - VOICEMAIL AND RECORDING

When you click on it, it says: “Use your Voicemail Mailbox and Password...” This is the same password used for the phone.

If you want to access your voicemail through the web client FreePBX, your extension must have voicemail enabled and a password entered. Your username is your extension ID (ex. 1001) and your initial password is the voicemail password configured in the extension. Unless you need more security on your voice mail, configure these passwords to be the same as the two extensions that are setup.

Click on the Main Menu icon in the upper right to get back to the PBX in a Flash - Dynamic UI: Menus - users menu.

PASSWORD - Flash Operator Panel

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

In this example, Putty is used to log in remotely to PIAF. Once logged in, change the directory to /var/www/html/panel
cd /var/www/html/panel

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

nano op_server.cfg

Go to the line that says security code=passw0rd
(In FOP that comes with PIAF, the default password is “passw0rd”)

Replace the “passw0rd” with the password of your choice.

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

From the Main Menu, you can click on the Flash Operator Panel icon. When you are in the PiaF Flash Operator Panel there is a lock icon: Open Security Code Input Box. When you click on it, it says “Please enter the Security Code”. The default password is passw0rd. All this does is let you click on the down arrow next to the extension and bring up a box that shows Call and Queue.

passwd-wwadmin... for users needing FOP and MeetMe access

PASSWORD – MeetMe Conference

MeetMe Conference - Web MeetMe Control comes up without a password.

passwd-meetme... for users needing only MeetMe access.

PASSWORD - FREEPBX – RECORDINGS

When you click on Recordings... It brings up a Login screen asking for Login and password. It says: Use your Voicemail Mailbox and Password. This is the same password used for the phone, for example extension 204 has the password set to 204.
Admin Function Passwords

In the lower left of the Main Menu there is an Admin toggle. Click on it and it changes to Users and brings up a password. The default is 123456. If you have already logged in, it goes directly to the Dynamic UI: Menus - admin menu with six icons: the three from users (Voicemail & Recordings, Flash Operator Panel, and MeetMe Conference) and FreePBX Administration, Linux Webmin, and Menu Configuration. From there you can click on FreePBX Administration. It will bring up the login to the server. The User name is wwwadmin and the default password is passwd.

From there if you click on Administrators. Username: admin Password: admin

PASSWORD - LINUX ADMIN FROM FREEPBX BROWSER

Any Linux Admin uses the same password, whether you log into PBX in a Flash and then click on Linux Admin or if you login using SSH and PuTTY. The username is root. The default password is 123456.

login as: root
root@xxx.xxx.xxx.xxx's password: 123456

PASSWORD - SQL

The Default user = asteriskuser and password = amp109. If you do not change it, the FreePBX System Status will warn you: “Default SQL Password Used”.

PASSWORD – ASTERISK

This is also the password for Sys Info.

# AMPMGRUSER: the user to access the Asterisk manager interface
AMPMGRUSER=admin

# AMPMGRPASS: the password for AMPMGRUSER
AMPMGRPASS=amp111

The Default username is freepbx and the default password is fpbx. If you do not change it, the FreePBX System Status will warn you: “Default Asterisk Manager Password Used”.

PASSWORD - TOOLS - CONFIG EDIT

The default username is maint and password is 123456.

passwd maint... This command sets FreePBX maint password. It covers Config Edit, phpMyAdmin, and Sys Info and everything covered by .htaccess in /var/www/html/maint.

PASSWORD – TOOLS – SYS INF

See PASSWORD ASTERISK

PASSWORD – WEBMIN

Use the command line passwd webmin for users needing Webmin access to your server.

PASSWORD – GRANDSTREAM IP PHONES

To enter the configuration of the phone itself, enter its IP address. The password is admin.
**Fail2Ban**

Fail2Ban is an intrusion monitor and prevention framework for IP security. Fail2Ban is installed and pre-configured with PiaF.

You can make certain it is online by looking at the PBX Server Daemon Status when you first connect to it.

```
********************************************************************
*           PBX in a Flash Version 1.3 Daemon Status               *
*                      Running Asterisk 1.4                        *
********************************************************************
* Asterisk   * ONLINE  * Zaptel    * ONLINE  * MySQL      * ONLINE  *
* SSH       * ONLINE  * Apache    * ONLINE  * Iptables   * ONLINE  *
* Fail2ban  * ONLINE  * Ethernet0 * ONLINE  * IP Connect * ONLINE  *
```

Fail2ban scans log files and bans IP addresses that make repeated, unsuccessful password attempts. It updates iptables rules to reject those IP addresses for a period of time that you can set in `/etc/fail2ban/jail.conf`.

/var/log/fail2ban.log

**QUESTION:**
I locked myself out by trying too many times after forgetting the password. How do I fix this?

**ANSWER:**

```bash
iptables -nvL fail2ban-ASTERISK
gen_list all entries in the Asterisk chain.
```

If you see your IP address listed,

```bash
iptables -D fail2ban-ASTERISK <number>
gen_rule number you want to remove.
```

**QUESTION:**
How do I update Fail2Ban?

**ANSWER:**

These are the commands to run to get the current release:

```
cd /root
mkdir fail2ban
cd fail2ban
wget http://pbxinaflash.net/source/fail2ban/fail2ban-update
chmod +x fail2ban-update
./fail2ban-update
.service fail2ban restart
```

**More Info:**

http://www.voip-info.org/wiki/view/Fail2Ban+(with+iptables)+And+Asterisk
http://www.fail2ban.org/wiki/index.php/Main_Page

For more information on how to use Linux commands, see the [Linux Commands](http://www.fail2ban.org/wiki/index.php/Main_Page).
Apache HTTP Server

The Apache HTTP Server is an open-source HTTP server for modern operating systems. FreePBX chose it as the best possible firewall. PiaF started out in a security configuration, with ACL enabled, but after a security hole was discovered (now fixed), the authentication was moved over to Apache Directory Authentication.

IMPORTANT NOTE:

If your PBX is to be exposed to the internet, you should not change the security model supplied with PiaF. Apart from which, this change will take you out of the update-fixes loop.

Technically speaking, it is possible to change the security model - taking note of all of the above caveats - but this is not recommended or supported.

You should not have to change the Apache Server. For more information on it, please see the Apache HTTP Server Project website:  http://httpd.apache.org/docs/1.3/howto/htaccess.html
All about CallerID

There are CallerID parameters for incoming and outgoing calls. In some countries, in order to get Caller ID, a user must apply to their Telco to have it activated on their line. If caller ID is not activated on their line, they will not get CID.

Outgoing

You can configure CallerID for outgoing calls in the IP Phone, the Trunk, and in Extensions. If the Trunk is configured, it overrides all clients' caller IDs for calls placed out this trunk. The Extensions setting overrides the Trunk. That might be over-ridden by the phone service provider or the incoming PBX.

Trunks

Outbound Caller ID – Caller ID for calls placed out on this trunk.

Format: "caller name" <#######>. Quotes are optional around the caller name, but highly recommended. You can also use the magic string 'hidden' to hide the CallerID sent out over Digital lines ONLY (E1/T1/J1/BRI/SIP/IAX).

Leave this field blank to simply pass client caller IDs.

Never Override CallerID – Some VoIP providers drop the call if you try to send an invalid CallerID. An invalid CallerID is defined as one that you don't 'own'. Use this to never send a CallerID that you haven't explicitly specified in this trunk or in the outbound callerid field of an extension/user. You might notice this problem if you discover that Follow-Me or RingGroups with external numbers don't work properly. Checking this box has the effect of disabling 'foreign' callerids from going out this trunk. You must define an Outbound Caller ID on the trunk when checking this.

Extensions

Display Name – The caller id name for calls from this user will be set to this name. Only enter the name, NOT the number. This is the name that will display on the phone that you call.

CID Num Alias – The CID Number to use for internal calls, if different from the extension number. This is used to masquerade as a different user. A common example is a team of support people who would like their internal callerid to display the general support number (a ring group or queue). There will be no effect on external calls.

SIP Alias – If you want to support direct sip dialing of users internally or through anonymous sip calls, you can supply a friendly name that can be used in addition to the user’s extension to call them.

Outbound CID – Overrides the caller id when dialing out a trunk. Any setting here will override the common outbound caller id set in the Trunks admin.

Format: "caller name" <#######>

Leave this field blank to disable the outbound callerid feature for this user.

Emergency CID – This caller id will always be set when dialing out an Outbound Route flagged as Emergency. The Emergency CID overrides all other caller id settings.

Grandstream phone settings

Account Settings

Name – SIP service subscriber’s name that is used for Caller ID display.

Distinctive Ring Tone – Caller ID must be configured. Select a Distinctive Ring Tone 1 through 3 for a particular Caller ID. The GXP will ONLY use selected ring tones for particular Caller IDs. For all other calls, the GXP will use System Ring Tone. When selected and no Caller ID is configured, the selected ring tone will be used for all incoming calls.
Key Call Features

*30 Block Caller ID (for all subsequent calls)
*31 Send Caller ID (for all subsequent calls)
*67 Block Caller ID (per call)
*82 Send Caller ID (per call)

Incoming

In some countries, in order to get Caller ID, you must apply to your Telco to have it activated on your line. If caller ID is not activated on your line, you will not get CID.

Inbound Routes

Caller ID Number – Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no CID info. In addition to standard dial sequences, you can also put Private, Blocked, Unknown, Restricted, Anonymous and Unavailable in order to catch these special cases if the Telco transmits them.

CID Priority Route – This affects CID ONLY routes where no DID is specified. If checked, calls with this CID will be routed to this route, even if there is a route to the DID that was called. Normal behavior is for the DID route to take the calls. If there is a specific DID/CID route for this CID, that route will still take the call when that DID is called.

CID name prefix – You can optionally prefix the Caller ID name. i.e.: If you prefix with "Sales:", a call from John Doe would display as "Sales:John Doe" on the extensions that ring.

Add Inbound CID – Add a CID for more specific DID + CID routing. A DID must be specified in the above Add DID box. In addition to standard dial sequences, you can also put Private, Blocked, Unknown, Restricted, Anonymous and Unavailable in order to catch these special cases if the Telco transmits them.

Privacy Manager – If no Caller ID is sent, Privacy Manager will ask the caller to enter their 10 digit phone number. The caller is given 3 attempts. The number of digits and attempts can be defined in privacy.conf. If a user has Call Screening enabled, the incoming caller will be asked to enter their CallerID here if enabled, and then to say their name once determined that the called user requires it.
**Caller ID Troubleshooting**

If you have Caller ID activated and still don’t get Caller ID, look at the ZAP configuration file. Typically you may have to look at zapata.conf and/or zapata-auto.conf. You may need to set the following switches in your zapata.conf and zapata-auto.conf.

**zapata.conf**

```plaintext
usecallingpres=yes
callwaitingcallerid=yes
threewaycalling=yes
usecallerid=yes
hidecallerid=no
relaxdtmf=yes
```

**zapata-auto.conf**

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

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

After the above are done, restart PiaF: `amportal restart`
ZapBarge

ZapBarge is a Feature Code that can be enabled in the FreePBX - Setup - Feature Codes.
ZapBarge (channel) lets you listen to the conversation on a specified Zap channel. ZapBarge default code to connect is 888 Enabled Disabled
Multiple people can all use ZapBarge to listen in on the same channel
http://www.voip-info.org/wiki/view/Asterisk+cmd+ZapBarge
If you dial 888, it will ask you for the Zap channel followed by the # key. This will be Zap Channel 1-4.

ChanSpy

ChanSpy is a Feature Code that can be enabled in the FreePBX - Setup - Feature Codes.
ChanSpy default code to connect is 555
Listen in on a call. It is useful in a call center to monitor agents on the phone.
http://www.voip-info.org/wiki/view/Asterisk+cmd+ChanSpy
If you dial 555, it will listen to whatever channel is open. It will just announce what channel it is listening to.
Interactive Voice Response System

Most organizations would like to automate the redirecting of all incoming calls. The Digital Receptionist is very handy for these sorts of things. Unless the calls are non-specific and will need the assistance of a live receptionist the system should allow callers to make the selection.

You use the Digital Receptionist to make IVR's, Interactive Voice Response systems.

When creating a menu option, apart from the standard options of 0-9,* and #, you can also use 'i' and 't' destinations. 'i' is used when the caller pushes an invalid button, and 't' is used when there is no response. If those options aren't supplied, the default 't' is to replay the menu three times and then hang up, and the default 'i' is to say 'Invalid option, please try again' and replay the menu. After three invalid attempts, the line is hung up.

Planning

First, draw out on paper what you intend to achieve. Run it by the other people in your location. Write out word-for-word what all the recordings are going to be.

The proper flow to build a good IVR is:

1) Planning.
2) Agreement with the plan.
3) Record the audio prompts using System Recordings and an extension.
4) Create any destinations that don't currently exist (queues, ring groups, day/night modes or time conditions).
5) Test all of these. One way to do this is use miscellaneous destinations, assigning a * feature code to whatever thing you want to test.
6) Then go create your IVR.
7) Create an Inbound Route to the IVR.
8) Show it to the users, and then make the inevitable changes.
9) Now upgrade the voice prompts to a paid voice or designated employee (the office manager or receptionist, etc.)

Standard IVR Examples:

Office / Light industrial
Welcome to BUSINESSNAME. Please listen carefully as our options have changed. If you know the extension of the person you are trying to reach, you may dial it at any time. Press 1 for sales, press 2 for customer service, press 3 for administration, press 4 for Press inquiries, press 5 for office directions, press # to access the company directory, or press 0 for the operator.

Engineering/Product Company with Direct Sales and Support
Welcome to BUSINESSNAME. Please listen carefully as our options have changed. If you know the extension of the person you are trying to reach, you may dial it at any time. Press 1 for sales, press 2 for customer service, press 3 for technical support, press 4 for administration, press 5 for Press inquiries, press 6 for office directions, press # to access the company directory, or press 0 for the operator.
System Recordings

System recording is a facility available under FreePBX to enable the recording of customized voice prompts etc. The recording in the basic system are generic and will work fine.

To get to the system recordings, select the Setup tab and pick System Recordings.

There are two methods that you can capture your recording:
• By recording directly using your telephone
• Uploading pre-recorded materials

Direct Recording

It is strongly suggest you use an extension connected to the PBX to make your recordings. They'll be quick and in the right format and you can worry about getting everything else right. When everything is all finished, you can come back and replace those temporary recordings with paid or improved versions.

First click on Add Recording to bring up the interface.

System Recordings

Add Recording

Step 1: Record or upload

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

Enter the extension number of the mouthpiece that you will be using to record your message in the appropriate field. You will be making your recording using the telephone on this extension (or the microphone of the SoftPhones).

Click Go.

Follow the prompt on the screen and dial *77 on your phone and make your recording after the beep.

Verify the recording by dialling *99.

Once you are happy with the recording, (and don't obsess here) give the recording a descriptive name so you will know what the recording is for. For lame and silly reasons, spaces are not allowed in the names. Click Save.

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

If you have a recording that you have prepared in any supported Asterisk format, you can simply upload that recording by browsing your local hard drive, locating the file that you want to upload and upload it to PiaF.

**NOTE:** If you are using .wav (typically recorded with Microsoft Recorder) the file must be PCM Encoded, 16 Bits, at 8000 Hz.

Step 1: Upload

Follow the prompt for the upload.

Step 2: Name

Give the recording a descriptive name so you will know what the recording is for. For lame and silly reasons, spaces are not allowed in the names. Click Save.

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

The built-in recordings are found in the `/var/lib/asterisk/sounds/custom` directory. To access them, click on the built-in recordings button.

System Recordings

Edit Recording

To enable a system recording for use with the PBX, select it and click the Go button.

You can select multiple files to be played and then change the order in which they are played.

Click the Save button after adding each sound, and it will offer another line.

From this menu, you may be able to click on the speaker to play messages. If not, use the Webmin – Others – File Manager to play them. Go to the `/var/lib/asterisk/sounds` directory to play them.

There is a text file named `extra-sounds-en.txt` that lists the names all of the Built-in Recordings and a brief description of it. It is found in the `/var/lib/asterisk/sounds` directory.

Once finished, click Save and also click on the Apply Configuration Changes.
Setting up the Digital Receptionist

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

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

Change Name: The short name, visible on the right, of this IVR, and in the drop-down menu of Destinations.

Announcement: A System Recording that is played to users when they enter the IVR. This can be set to 'nothing'. These announcements are great for "today is July 4th and we're closed for the holiday" and then proceeding on to the regular call flow.

Timeout: The amount of time (in seconds) before the 't' option, if specified, is used. 10 is the default.

Enable Directory: Tick to let callers dial '#' to access the directory.

VM Return to IVR: If checked, upon exiting voicemail a caller will be returned to this IVR if they got a users voicemail

Directory Context: When # is selected, this is the voicemail directory context that is used. Advanced users can then use different IVRs to create a multi-tenant installation.

Enable Direct Dial: Tick to let callers into the IVR dial an extension directly.
Loop Before t-dest: If checked, and there is a 't' (timeout) destination defined below, the IVR will loop back to the beginning if no input is provided for the designated loop counts prior to going to the timeout (t) destination.

Timeout Message: If a timeout occurs and a message is selected, it will be played in place of the announcement message when looping back to the top of the IVR. It will not be played if the t destination is the next target. See the “t” option below, also.

Loop Before i-dest: If checked, and there is an 'i' (invalid extension) destination defined below, the IVR will play invalid option and then loop back to the beginning for the designated loop counts prior to going to the invalid (i) destination.

Invalid Message: If an invalid extension is pressed and a message is selected, it will be played in place of the announcement message when looping back to the top of the IVR. It will not be played if the t destination is the next target. If nothing is selected, the system will play a default invalid extension message before going back to the main announcement. See the “i” option below also.

Repeat Loops: The number of times we should loop when invalid input or no input has been entered before going to the defined or default generated ‘i’ or ‘t’ options. If the ‘i’ or ‘t’ options are defined, the above check boxes must be checked in order to loop. 0-9

Options
These are the options that will be given to the caller in the form of ‘dial 1 to reach option 1, dial 2 to reach option 2’, etc.

Use ‘Increase Options’ or ‘Decrease Options’ to alter the number of options available. This won’t let you decrease it to less than the number of options that are currently set.

Options are only displayed if there is at least one entry created. For example, queues will not appear as a possible IVR destination if no queues exist.

Put a number, “t”, or “i” in the block below Return to IVR for each destination option selected.

To delete an option, simply leave the selection blank.

i: This overrides the default invalid choice behavior, which is to play a ‘invalid option’ message and immediately replay the current menu. If you only have 1 2 and 3 defined, and caller pushes 4, it will jump to this destination.

t: This overrides the default timeout behavior, which is to play the menu three times and hangup. A standard configuration is to go the operator, to handle customers that don’t have DTMF-capable phones. The last extension should have the letter “t” for timeout. If the caller does not press any option or, in case the DTMF tone does not register, the call will be sent to the receptionist.

Once finished, click Save and also click on the Apply Configuration Changes.
**Set up a Digital Receptionist Route**

You must configure a Digital Receptionist Route and point it to the IVR.

1. Under the **Setup** tab, select **Inbound Routes**.
2. Click on **Add Incoming Route**.
3. Enter a description.

4. Under **Set Destination** at the bottom of the page, select the IVR for the Digital Receptionist.

5. Once finished, click **Save** and also click on the **Apply Configuration Changes**.

6. Test it by dialling **7777**, (Asterisk will simulate an incoming call) and you will hear your Digital Receptionist in action.
Multi-Language IVR

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

Your first IVR is known as IVR-2 in your extension_additional.conf file. Any subsequent IVR created will be IVR-3, IVR-4 and so on.

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

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

In LanguageChoice, we will have 3 options.
• Select 1 for Australian English
• Select 2 for Italian
• The third option is unannounced as it will be the time out option represented by the letter “t”. This option is the option the IVR will default to if no choice is made or if the caller’s DTMF tone is not recognised by PiaF.

In choice 1, select the Custom App: radio button and enter the following in the box:
Custom App custom-language_au,s,1
In choice 2, select the Custom App: radio button and enter the following in the box:
Custom App custom-language_it,s,1
The third choice, instead of a number, you will enter the letter “t” in the option box, select
Custom App: radio button and enter the following:
Custom App custom-language_au,s,1

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

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

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

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

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

Now you will need to change your inbound route to point to the LanguageChoice IVR.
All calls will then be greeted by the LanguageChoice IVR and when the caller selects 1 (or 2), the IVR will call the appropriate extension. If the caller selected 1, the [customlanguage_au] will be selected and the language will be set to Australian English, an announcement in the appropriate language will be made and the caller will be sent to the MainMenu (or MainMenu_IT) for call options (you must also have recorded the call options announcement in the various languages).
Troubleshooting Tools

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

FreePBX System Status

First click on FreePBX System Status to ensure that Asterisk is up and running. This is the default PiaF screen.

Check the Server Status in the lower right. If all processes are OK then your system is functional and you may have a configuration problem.

If any of the servers aren’t running, something is wrong and your Asterisk will not be working correctly or not working at all. A probable cause can be wrong configuration of your zaptel drivers. If you don’t have a zaptel device, you should try to disable the related .conf files.
Debug Messages and Log Files

Another way to monitor the activities of your PiaF is by using the following command at the command line (which will give you the Asterisk CLI).

Asterisk –rvvv (the “v” depends on the level of verbosity you need)

For more information on how to use Linux commands, see the Linux Commands.
PBX in a Flash Forum

The PBX in a Flash Forum at http://pbxinaflash.com/forum/ is a very helpful place to find solutions and get troubleshooting assistance. Before you make a post to the forum, you should search it first to see if the answer to your problem is already there. When you post a question or problem in this forum, you should include:

- A short paragraph about the problem you are having.
- A copy of the output of the status program. This will help those in the know a lot and it will go a long way to narrowing down your particular version.
- What processor/motherboard/amount of ram/and if you are using a solid state drive.
- PBX in a Flash Version: (valid ones are 1.0, 1.1, and 1.2) Linux status command
- Operating system: (valid are 32bit Centos 5.0, 32bit Centos 5.1, 64bit Centos 5.1)
- Asterisk Version: (valid is Asterisk 1.4 old*, Asterisk 1.4, Asterisk 1.6 BETA)
- Have you run update-scripts, update-fixes, and update-source?
- Relevant logs only! Don’t post full logs into the forum! Post only snippets of long log files. If there is not enough info people will make some suggestions.

To post a message for help, go to the Forum Topics – Help, and click on the New Thread button.

▷ PBX in a Flash Forum » Forum Topics » Help

New Thread
Problems & Solutions

PROBLEM:
I want to connect specific inbound lines to specific extensions. Inbound route definition for a zap channel does not work, but a catch-all inbound route definition captures the call correctly.

CAUSE:
Digital zap lines go into the from-pstn context, which directs the call straight into the inbound route module, as the DID is contained within the signalling of the call.
Analogue lines go into the from-zaptel context, which passes the call into the zap channel DID module to attach a DID to the call, and then passes the call into Inbound Routes.
The zaptel conf definition needs to specify from-zaptel instead of from-pstn.

SOLUTION:
Used Tools / Config Edit from FreePBX Administrator to change zapata.conf and zapata-channels from context=from-pstn to context=from-zaptel.
After that you set up the Zap Channel DIDs to the extension used by the outside world. Then set up your Inbound Routes to that DID number with the check in CID Priority Route.
Use PuTTY to amportal restart.

NOTE:
If you use the genzaptelconf command, you may have to change it back again.
You must restart asterisk or run reload chan_zap.so after doing so.

PROBLEM:
Voice lasts three minutes for hang-uppers and even for short messages. This occurs even when the phone is picked up.

CAUSE:
The channel needs to be told to detect busy and for how long.

SOLUTION:
Use the FreePBX Admin tool to open vm_general.inc
Set silencethreshold = 1000
Then use PuTTY to
amportal restart
zapata.conf
Under [channels] add the following lines:
[channels]
busydetect=yes
busycount=6
PROBLEM:
I am getting the message "Default SQL Password Used" in the FreePBX System Status.
You are using the default SQL password that is widely known, you should set a secure password

SOLUTION:
MySQL password
You must change the password for the database (service), AND for ALL clients to the database, this is FreePBX and asterisk (asterisk and asterisk_cdr).
The user 'maint' is for FreePBX and the other web tools, whereas 'wwwadmin' is for FreePBX only.

1. Run `mysqladmin -u asteriskuser -p password <newpassword>` from the shell (where `<newpassword>` is replaced with the actual new password - mysqladmin will prompt for the _old_ password). This changes the DB service.

2. Edit `/etc/amportal.conf` and change AMPDBPASS to the new password (replace amp109 with the new password). This changes the amp mysql client.

3. Edit `/etc/asterisk/res_mysql.conf` and change the dbpass = line (again, replace amp109 with your new password). This changes the asterisk mysql client

4. Edit `/etc/asterisk/cdr_mysql.conf` and change the password= line (again, replace amp109 with your new password). This changes the asterisk CDR MySQL client

The web interface password (user 'maint')
This is the password used for logging in to the FreePBX interface from a web browser as user 'maint'.
There is no warning about this being set to the default, but you should change it too while you are at it.
The web interface password (user 'wwwadmin'). This is the password used for logging in to the FreePBX interface from a web browser as user 'wwwadmin'.

1. Run `passwd-maint` from the shell
Set password for AMP web GUI and maint GUI
User: maint
2. Run the amportal restart command.

PROBLEM:
When you select Tool and phpMyAdmin it reports: “phpMyAdmin tried to connect to the MySQL server, and the server rejected the connection. You should check the host, username and password in your configuration and make sure that they correspond to the information given by the administrator of the MySQL server.”

Error
MySQL said:
#1045 - Access denied for user 'root'@'localhost' (using password: YES)

SOLUTION:
To change the password in phpMyAdmin, find the line: `-S$cfg['Servers'][$i]['password'] = 'passw0rd';` in /
/var/www/html/maint/phpMyAdmin/config.inc.php and change it to whatever you set your MySQL root password to.
Then run the mysql_install_db script.
**PROBLEM:**
FATAL ERROR DB Error: connect failed

**CAUSE:**
This can occur after changing passwords.

**SOLUTION:**
Make certain that all passwords match. Then use Webmin - MYSQL Database Server to open up the permissions.

---

**PROBLEM:**
In the FreePBX System Status you are getting the message: “Default Asterisk Manager Password Used - You are using the default Asterisk Manager password that is widely known, you should set a secure password”

Added 0 minutes ago
(core.AMPMGRPASS)

**SOLUTION:**
Change the Manager password
You need to update the manager (service) and FreePBX (client).
You change the password in the manager.conf and amportal.conf,
1. Edit /etc/asterisk/manager.conf and change the password in the [admin] section
nano etc/asterisk/manager.conf

```
[admin]
secret = amp111
deny=0.0.0.0/0.0.0.0
permit=127.0.0.1/255.255.255.0
read = system,call,log,verbose,command,agent,user
write = system,call,log,verbose,command,agent,user
```
2. Edit /etc/amportal.conf and change the password for AMPMGRPASS
nano /etc/amportal.conf
Change the amp111 and amp109

```
# AMPDBPASS: the password for AMPDBUSER
AMPDBPASS=amp109

## AMPMGRPASS: the password for AMPMGRUSER
AMPMGRPASS=amp111
```
PROBLEM:
In the FreePBX System Status you are getting the message: “There are 1 bad destinations DEST STATUS: ORPHAN - Daynight: (day) - retrieve_conf.BADDEST

CAUSE:
Unknown Destination ERROR: You have an unknown destination. If this was carried over as a Custom App from an earlier version, you must go register the destination in the Custom Destination tab provided by the Custom Applications module.

SOLUTION:
Check the Day / Night Mode Control.
The problem should be highlighted in red.
Just point to a good destination.

PROBLEM:
When calling out to an outside IVR the DTMF codecs do not seem to work.

CAUSE:
Problem could be related to WIFI connection not properly decompressing the codec or it could be the phone settings.

SOLUTION:
Grandstream Settings:
Disable in-call DTMF display: set to No
DTMF Payload Type: 101
Send DTMF: x in-audio x via RTP (RFC2833) via SIP INFO

PROBLEM:
Voice mail problem. You can leave voice mail on the extension. The light blinks and the user gets an email with the voice mail attached, but the user cannot pick it up from the phone. When you call voice mail, and enter the extension, there is a long delay before it asks for the password, and another delay after entering the password. The IVR finally responds with a "Login incorrect" message.

SOLUTION:
Check the DTMF settings at the phone and at the extension. They should match - usually RFC2833.
First, in the FreePBX Administration, click on Extensions. Under Device Options, you will see dtmfmode. It should be RFC2833.
Second, bring up the Grandstream Device Configuration. Click on ACCOUNT 1 and make certain that there is a check in Send DTMF: "in-audio" and "via RTP (RFC2833)"
If that does not work, reset defaults on phone using the round button:
   1) Press the round button, then up arrow to Config.
   2) Press the round button to select Config.
   3) Press the down arrow to Factory Reset.
   4) Press the round button to select Config.
Re-enter the IP address and set DTMF correctly.
PROBLEM:
Network problems.

SOLUTION:
Verify that you can ping stuff on the internet:
Go to a command prompt and type: "ping www.google.com"
('Ctrl/c' to break out of ping)
If you do not get a good response, do the following to ensure you have a valid IP address:
Type "ifconfig". The entry is next to eth0 that you are interested in.
If the Ethernet addresses are not correct, type system-config-network, and fill in the gaps.
Reboot and verify that you can ping the router.
If not, you need to install the driver for your card.
If you can, you have to add DNS services.
Type "nano /etc/resolv.conf"
Type nameserver <Your DNS Server Address>.
You can get your DNS address from say your windows computer by typing ipconfig /all at the command prompt.
Now you should be able to ping www.google.com.
If you can, reboot. When the server reboots, you should be resisting various urges concerning the Enter key in a short space of time.

PROBLEM:
Can’t web browse to Asterisk

CAUSE:
Fail2ban may have locked you out from the IP address of the computer you are browsing to PIAF from, if you 1. You’ve saved the password in your browser, and changed the password on the PBX (or you have re-installed it) -or- 2. You’ve set up a SoftPhone with the wrong credentials.

SOLUTION:
Make certain that port 9080 is open to access the PIAF server.
You can check /var/log/fail2ban.log to see if this has happened. From the command line, issue:
iptables -nvL fail2ban-ASTERISK  This will list all entries in the Asterisk chain.
If you see your IP address listed, issue the command:
iptables -D fail2ban-ASTERISK <number> where <number> is the rule number you want to remove.
PROBLEM:
The PBX hangs.

SOLUTION:
First, in the FreePBX Administration, click on System Status for clues.
Check the FOP for clues.
Check /var/log/asterisk/full for errors.
PuTTY to it and look for errors.
Check the date and time settings on your machine:
```
root@pbx:~ $ date
```
Delete the trunk registrations and recreate them.
Try removing the ext(s) add a new ext.

PROBLEM:
The Asterisk log is too big or does not have information in it.

SOLUTION:
Set /etc/asterisk/logger.conf as follows:
```
;full => notice,warning,error,debug,verbose (Use verbose and debug only if problems are occurring).
full => notice,warning,error
```
Check the "log rotation" in /etc/logrotate.d/asterisk.

PROBLEM:
No matching peer found. This is seen when you view the log. /var/log/asterisk/full:

```
```

CAUSE:
The second IP address indicates the one that is causing the problem. Something in that IP phone does not line up with any extensions configured in FreePBX.
Problem can be caused by a bad or no DNS configuration.
Problem could be a hacker trying to access the system.
The problem could occur if you have two accounts set up on a phone, and the second one is not configured properly or is only partially configured.

SOLUTION:
On the GrandStream phone clear the parameters on the second extension or configure them correctly.
PROBLEM:
There is a huge delay in audio with calls that usually result in me picking up my mobile and returning the call.

POTENTIAL CAUSES:
1) The network
2) It could be old firmware in the phone.
3) The power supply (they go bad and can cause problems that seem unrelated)
4) The Ethernet cable.
5) The handset cord.
6) The phone audio card could be bad.

SOLUTIONS:
1) Check to see if the same symptoms exist if you move the phone to another network point.
2) Update the firmware and see if that helps. Voip-info has a good page on the various versions: http://www.voip-info.org/wiki/view/G...Firmware+Notes
3) Replace the power supply (about $10)
4) Replace Ethernet cable.
5) Replace the handset cord.
6) Replace the phone.

PROBLEM:
ERROR MESSAGE:

CAUSE:
These settings cannot be changed on a reload, only on a unload/load. It just means that the signalling, switchtype and rxwink is already set up and ignoring it on the reload.

SOLUTION:
Nothing to worry about.
PROBLEM:

ERROR MESSAGE:

[2009-09-11 10:10:34] WARNING[3613] channel.c: Ignoring invalid group 200 (maximum group is 63)

System continues to function just fine.

CAUSE:


A callgroup and pickupgroup was set incorrectly in Extensions.

Call groups and pickup groups

This kind of group is used to allow picking up remotely a ringing phone through *8(#) (by default, it is denied). The call group is what the extension belongs in; the pickup group is which callgroups the extension can remotely pick up.

The basic functionality is this:

•A call is placed in one or several call groups

•If a phone belongs in a pickup group that matches one of the call's call groups, that phone may pickup the incoming call by calling *8# on his phone

You define call and pickupgroup per device (in sip.conf under each extn# section), like

callgroup=1
pickupgroup=1-9,13

PROBLEM: ERROR MESSAGE:

[2009-09-14 09:54:12] WARNING[29633] pbx.c: Context 'from-pstn' tries to include nonexistent context 'from-pstn-custom'

[2009-09-14 09:54:12] WARNING[29633] pbx.c: Context 'from-internal-additional' tries to include nonexistent context 'from-internal-additional-custom'

[2009-09-14 09:54:12] WARNING[29633] pbx.c: Context 'macro-vm' tries to include nonexistent context 'macro-vm-custom'

[2009-09-14 09:54:12] WARNING[29633] pbx.c: Context 'vm-callme' tries to include nonexistent context 'vm-callme-custom'

[2009-09-14 09:54:12] WARNING[29633] pbx.c: Context 'from-zaptel' tries to include nonexistent context 'from-zaptel-custom'

CAUSE:

These messages are warnings only and can be ignored.

SOLUTION:

If they bother you, you can add empty context for each one. Example:

For this one:

[2009-06-10 20:16:44] WARNING[2862] pbx.c: Context 'ext-group' tries to include nonexistent context 'ext-group-custom'

add [ext-group-custom] to extension-custom
PROBLEM:
WARNING[27450] chan_zap.c: Unable to request echo training on channel 1

CAUSE:
In zapata.conf
  echocancel=yes
  echocancelwhenbridged=no
  echotraining=800

The reason that Asterisk is returning this error about echo training is that echo training is software based, and since a hardware echo can is in place, Asterisk is smart enough NOT to try to implement the software echo can as well.

The ONLY parameter required for echo cancellation to work is the echocancel=yes.

SOLUTION:
Simply remove echotraining=yes or set it to no, from your zapata.conf and the error will go away, and you won't lose any functionality.
  echocancel=yes
  echocancelwhenbridged=no
  ;(if your phones are all on the local network: echocancelwhenbridged=no)
  ;(if you access FreePBX from remote: echocancelwhenbridged=yes)
  echotraining=no

PROBLEM:
NO AUDIO ON ANALOG LINE Call on an analog line seems to ring and connect correctly but no audio

SOLUTION:
PiaF uses the OSLEC echo canceller which has provided significant improvement on echo problems. In order for this to operate correctly echotraining needs to be commented out in zapata.conf. The installation does not always comment it out so it needs to be checked. The same parameter needs done for every FXS extension, but this is done in FreePBX GUI. After defining the zap extension go back and change the echotraining parameter from its default 800 to "blank".

PROBLEM: ERROR:
[2009-09-14 09:54:12] WARNING[29633] pbx_config.c: The use of '_.' for an extension is strongly discouraged and can have unexpected behavior. Please use '_X.' instead at line 1527

CAUSE:
Something in extensions.conf

SOLUTION:
This is normal. The errant entry is commented on in extensions.conf, I think with the words "Yes I know what I'm doing"
PROBLEM:
I have two phones set up: Marc at 204 and Marty at 208. I can leave a voice mail for Marc, the light on the phone blinks, I can call *98, put in extension 204 and the same for password, and get his voice mails. I also enabled the email feature and that works as well.

Marty at 208 is another story. I can leave voice mail on extension 208. The light blinks and he gets an email with the voice mail attached, but he cannot pick it up from his phone. When you call voice mail, and enter the extension, there is a long delay, and a delay after entering the password. The IVR finally responds with a "Login incorrect" message.

You can check Marty's email from Marc's phone, with Marty's phone extension and password.

The debug logs a couple of interesting items:


SOLUTION:
Check the DTMF settings at the phone and at the extension. They should match - usually RFC2833.

First bring up the FreePBX Administration.

Click on Extensions
Under Device Options, you will see dtmfmode. It should be RFC2833.

Second, bring up the Grandstream Device Configuration.

Click on ACCOUNT 1 and make certain that there is a check in Send DTMF: "in-audio" and "via RTP (RFC2833)"

If that does not work, reset defaults on phone using the round button:
1) Press the round button, then up arrow to Config.
2) Press the round button to select Config.
3) Press the down arrow to Factory Reset.
4) Press the round button to select Config.

Re-enter the IP address and set DTMF correctly.

PROBLEM:
Seeing yellow warnings in FreePBX.

CAUSE:
Various.

SOLUTION:
Hover your mouse over them, and see if it gives you a clue to the error.

For more information on how to use Linux commands, see the Linux Commands.
Call Failed - Reason codes

If the phone displays Call Failed - Reason code ###, this guide will help you to interpreting SIP Reason Headers.

In some cases, SIP servers may be able to generate their own TDM reason codes using a feature called the 'SIP Reason Header', sometimes also referred to as 'Q.850'. Here the SIP header of the error response (e.g. '503 - Service Unavailable') also contains a Reason field that can be explicitly coded with a TDM reason code, for example:

Reason: Q.850; cause=34; text="no circuit"

Code Meaning
1xx Informational -- request received, continuing to process the request
2xx Success -- the action was successfully received, understood, and accepted
3xx Redirection -- further action needs to be taken in order to complete the request
4xx Client Error -- the request contains bad syntax or cannot be fulfilled at this server
5xx Server Error -- the server failed to fulfill an apparently valid request
6xx Global Failure -- the request cannot be fulfilled at any server.

Code Meaning
1xx Informational
100 Trying
180 Ringing
181 Call Is Being Forwarded
182 Queued

2xx Success
200 OK

3xx Redirection
300 Multiple Choices
301 Moved Permanently
302 Moved Temporarily
303 See Other
305 Use Proxy
380 Alternative Service
4xx Client Error
400 Bad Request
401 Unauthorized
402 Payment Required
403 Forbidden
404 Not Found
405 Method Not Allowed
406 Not Acceptable
407 Proxy Authentication Required
408 Request Timeout
409 Conflict
410 Gone
411 Length Required
413 Request Entity Too Large
414 Request-URI Too Large
415 Unsupported Media Type
420 Bad Extension
480 Temporarily not available
481 Call Leg/Transaction Does Not Exist
482 Loop Detected
483 Too Many Hops
484 Address Incomplete
485 Ambiguous
486 Busy Here

5xx Server Error
500 Internal Server Error
501 Not Implemented
502 Bad Gateway
503 Service Unavailable
504 Gateway Time-out
505 SIP Version not supported
6xx General Error
600 Busy Everywhere
603 Decline (9)
604 Does not exist anywhere
606 Does not exist anywhere

http://en.wikipedia.org/wiki/SIP_Responses
FreePBX Modules

The modules included in freepbx-core would be all those in the 'Basic' category in the FreePBX module system, with the exception of the language packs, which would get their own package:

* System Dashboard (dashboard),
* Feature Code Admin (featurecodeadmin),
* FreePBX ARI Framework (fw_ari),
* FreePBX FOP Framework (fw_fop),
* Voicemail (voicemail),
* Core (core), and
* FreePBX Framework (framework).

* freepbx-modules-asterisk:
  * Asterisk Logfiles (logfiles),
  * Asterisk Info (asteriskinfo),
  * Asterisk CLI (asterisk-cli), and
  * Asterisk API (manager)

* freepbx-modules-callcontrol:
  * Ring Groups (ringgroups),
  * Blacklist (blacklist),
  * Caller ID Lookup (cidlookup),
  * Day Night Mode (daynight),
  * Time Conditions (timeconditions),
  * Announcements (announcement), and
  * IVR (ivr)

* freepbx-modules-directory:
  * Phonebook (phonebook),
  * Print Extensions (printextensions),
  * Speed Dial Functions (speeddial), and
  * Phonebook Directory (pbdirectory)

* freepbx-modules-langpacks:
  * FreePBX Localization Updates (fw_langpacks)

* freepbx-modules-queues:
  * Queue Priorities (queueprio), and
  * Queues (queues)
* freepbx-modules-features:*
  * Music on Hold (music),
  * DISA (disa),
  * Languages (languages),
  * Dictation (dictate),
  * VoiceMail Blasting (vblast),
  * Parking Lot (parking),
  * Call Forward (callforward),
  * Paging and Intercom (paging),
  * PIN Sets (pinsets),
  * Misc Applications (miscapps),
  * Callback (callback),
  * Recordings (recordings),
  * Misc Destinations (miscdests),
  * Do-Not-Disturb (DND) (donotdisturb),
  * Conferences (conferences),
  * Info Services (infoservices), and
  * Call Waiting (callwaiting)

* freepbx-modules-sysadmin:*
  * PHPAGI Config (phpagiconf),
  * PHP Info (phpinfo),
  * Java SSH (javassh),
  * Backup & Restore (backup),
  * Online Support (irc),
  * DUNDi Lookup Registry (dundicheck), and
  * Custom Applications (customapppreg)

* freepbx-modules-thirdparty:*
  * Customer DB (customerdb),
  * Inventory (inventorydb), and
  * Gabcast (gabcast)
Common Asterisk file locations within Linux

Most configuration files reside in the etc directory and subdirectories.

/etc/ stores zaptel.conf file and mostly OS related files
/etc/rc.d/init.d/zaptel file that sets zaptel module loading order
/etc/sysconfig/zaptel file that sets zaptel module loading order
/etc/modprobe.conf file that sets module loading order
/etc/logrotate.d/asterisk file that sets log rotation of asterisk logs
/etc/hosts the host name for the server
/etc/asterisk/ stores all the Asterisk config files, like dialplan and users
/var/log/ log directory – message log and asterisk full logs are useful
/tftpboot/ store configs for phones to download
/usr/local/sbin/ the config files like genzaptelconf for setting up cards
/var/www/html/ directory where web server files are located
**Linux Commands**

**Command Line Interfaces**

You can get access to a CLI to run Linux commands using these methods:

- Using the browser to the **PBX IP Address**, then **Webmin** – **Others** - **Command Shell**.
- Using a SSH application like **PuTTY** to the **PBX IP Address**.
- Using a keyboard and monitor connected directly to the PBX.

Once you are at the Linux Command Line Interface, you can use standard Linux tools to view and edit (at your own risk) logs and conf files. Beware that FreePBX overwrites most conf files. You should only edit the custom files.

**Some Text Editors:**

- **nano** – A cursor based text editor. If you type a long line, nano will wrap it. You can join the wrapped line back to the original by backspacing on the wrapped line. `-w` disables wrapping.

  **Example:** `nano -w /etc/hosts`

- **tail** – is a nano command that lists the last 10 lines of a file. There are switches to change this value and show the log in realtime.

  **Example:** `tail --lines=500 /var/log/asterisk/full --display the last 500 lines of the log.`

- **grep** – this switch searches the contents of the files for the word stated.

  **Example:** `grep "AUTO FXO" /var/log/messages`

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>grep 'AUTO FXO' /var/log/messages</code></td>
<td>Tells you what mode your TDM400 is running on.</td>
</tr>
<tr>
<td><code>/etc/init.d/ntpd stop</code></td>
<td>To set the NTP time manually from the command line.</td>
</tr>
<tr>
<td><code>ntpdate timeserver URL</code></td>
<td></td>
</tr>
<tr>
<td><code>/etc/init.d/ntpd start</code></td>
<td></td>
</tr>
<tr>
<td><code>ifconfig</code></td>
<td>display/edit current network interface settings</td>
</tr>
<tr>
<td><code>modprobe -r zaptel</code></td>
<td>unloads a module (aka driver), substitute zaptel for any module name i.e. wctdm is the 4 port telephony card driver, if “service restart zaptel” fails, use this command to unload modules listed</td>
</tr>
<tr>
<td><code>reboot</code></td>
<td>Reboots the system</td>
</tr>
<tr>
<td>`rpm -qa</td>
<td>grep asterisk`</td>
</tr>
<tr>
<td>`rpm -qa</td>
<td>grep zaptel-modules`</td>
</tr>
<tr>
<td><code>service network restart</code></td>
<td>To restart network service if connection goes down and did not come up again even though internet has been restored.</td>
</tr>
<tr>
<td><code>service zaptel restart</code></td>
<td>Reloads the Zaptel telephony drivers</td>
</tr>
<tr>
<td><code>date</code></td>
<td>To see what the current local time is</td>
</tr>
<tr>
<td><code>date 051222152007.30 [ MMDDhhmm[[CC]YY][.ss] ]</code></td>
<td>To set the system clock under Linux, you need to use the “date” command. Example: To set the current time and date to May 12, 2007:10:15.30 seconds pm, (The time, in bold, is in 24 hour notation).</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>-------------</td>
</tr>
<tr>
<td><code>chown &lt;newowner&gt; &lt;filename&gt;</code></td>
<td>To change file ownership to the new owner</td>
</tr>
<tr>
<td><code>cd</code></td>
<td>Change directories</td>
</tr>
<tr>
<td><code>ls -l</code></td>
<td>View details such as the file permissions, owner and dates.</td>
</tr>
<tr>
<td><code>ls -l f*</code></td>
<td>List files that start with the letter f (or your choice of letters).</td>
</tr>
<tr>
<td><code>find / -name f*</code></td>
<td>Search for files starting with the letter f recursively starting from the / root directory.</td>
</tr>
<tr>
<td><code>man “command”</code></td>
<td>Displays the manual for a command/program: man nano</td>
</tr>
<tr>
<td>`cat file.txt</td>
<td>more`</td>
</tr>
<tr>
<td><code>grep &quot;phrase&quot; file.txt</code></td>
<td>Find a phrase in a file or folder.</td>
</tr>
<tr>
<td>`‘command’</td>
<td>grep “word”`</td>
</tr>
<tr>
<td>`cat “file”</td>
<td>grep “word”`</td>
</tr>
<tr>
<td><code>./filename</code></td>
<td>Executes a file.</td>
</tr>
<tr>
<td><code>touch filename</code></td>
<td>Create a file.</td>
</tr>
<tr>
<td><code>chmod +x filename</code></td>
<td>Change the permission of a file to be executable.</td>
</tr>
<tr>
<td><code>cp, mv, rm, mkdir</code></td>
<td>Copy, Move, Remove a file/dir and make a dir.</td>
</tr>
<tr>
<td><code>rm -r dirname</code></td>
<td>Deletes a directory with files in it. (use –rf to remove without asking)</td>
</tr>
<tr>
<td><code>chmod “owner” filename</code></td>
<td>Change the owner of a file.</td>
</tr>
<tr>
<td><code>chgrp “group” filename</code></td>
<td>Change the group ownership of a file.</td>
</tr>
<tr>
<td><code>unzip file.zip</code></td>
<td>Unzip a zip</td>
</tr>
<tr>
<td><code>wget http://file</code></td>
<td>Download a file (FTP also)</td>
</tr>
<tr>
<td><code>rpm -ihv “package.rpm”</code></td>
<td>Install a package.</td>
</tr>
<tr>
<td><code>rpm -e “package”</code></td>
<td>Remove a package, name without .rpm extension.</td>
</tr>
<tr>
<td><code>rpm -e --nodeps “package”</code></td>
<td>same as remove, but ignores/doesnt remove dependencies.</td>
</tr>
<tr>
<td><code>yum -y install “packagename”</code></td>
<td>Install a package.</td>
</tr>
<tr>
<td><code>yum remove “packagename”</code></td>
<td>Remove a package.</td>
</tr>
<tr>
<td><code>tar -zxvf file.tar.gz</code></td>
<td>Unpack a compressed archive file.</td>
</tr>
<tr>
<td><code>tar -Pcz -f “destinationTAR.tar.gz” “/sourcedir”</code></td>
<td>Create a compressed tar file from a folder.</td>
</tr>
</tbody>
</table>
**Mail**

You may see message that says “you have mail at /var/spool/mail/root”. To read the email, go to /var/spool/mail by using the command: `cd /var/spool/mail`. And then to read the email use the command: `nano root` or `tail root`. The file is appended to so you have to go to the end. You can use Ctrl/w and enter the current date in the format: 1 Sep 2009.

**Example:** `grep -i tom` would find all files with the word “tom” in them. It is case sensitive.

**Linux Commands:**

http://pbxinaflash.net/Conversational_Linux.pdf
http://www.ss64.com/bash/
http://www.linuxdevcenter.com/linux/cmd/
http://www.linuxcommand.org/learning_the_shell.php

The Linux Documentation Project: http://tldp.org/

**PBX in a Flash Help Menu**

The only command that you have to remember is the `help-pbx` command. It brings this list up.

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>* setup</td>
<td>Configure the Linux Operating System options</td>
</tr>
<tr>
<td>* status</td>
<td>Provides information about your PIAF System</td>
</tr>
<tr>
<td>* netconfig</td>
<td>Configure ethernet interface</td>
</tr>
<tr>
<td>* install-netconfig</td>
<td>Reinstalls netconfig which is deleted by Centos 5.1</td>
</tr>
<tr>
<td>* passwd-master</td>
<td>Fixes fpbx/webmin and sets passwds maint/amp/meetme</td>
</tr>
<tr>
<td>* passwd-maint</td>
<td>Set password for maint - web GUI</td>
</tr>
<tr>
<td>* passwd-wwwadmin</td>
<td>Set password for wwwadmin</td>
</tr>
<tr>
<td>* passwd-meetme</td>
<td>Set password for Web MeetMe</td>
</tr>
<tr>
<td>* passwd-webmin</td>
<td>Set password for Webmin</td>
</tr>
<tr>
<td>* passwd</td>
<td>Set root password for console login</td>
</tr>
<tr>
<td>* enable-iptables</td>
<td>Turns on IPTABLES (You may need to configure it!)</td>
</tr>
<tr>
<td>* disable-iptables</td>
<td>Turns off IPTABLES (for Freepbx problems)</td>
</tr>
<tr>
<td>* enable-fail2ban</td>
<td>Turns on Fail2ban IP security monitor</td>
</tr>
<tr>
<td>* disable-fail2ban</td>
<td>Turns off Fail2ban IP security monitor</td>
</tr>
<tr>
<td>* setup-astra</td>
<td>Create a aastrta.cfg in /tftpboot</td>
</tr>
<tr>
<td>* setup-cisco</td>
<td>Create a SIPDefault.cnf in /tftpboot</td>
</tr>
<tr>
<td>* setup-grandstream</td>
<td>Setup for autoconfiguration of Grandstream</td>
</tr>
<tr>
<td>* setup-_linksys</td>
<td>Create Linksys default files in /tftpboot</td>
</tr>
<tr>
<td>* setup-polycom</td>
<td>Create Polycom default files in /tftpboot</td>
</tr>
<tr>
<td>* setup-samba</td>
<td>Setup samba windows &lt;-&gt; linux file sharing</td>
</tr>
<tr>
<td>* genzaptelconf</td>
<td>Auto-configure Zaptel cards</td>
</tr>
</tbody>
</table>
* install-a2billing  * Set-up A2Billing <> VoIP & TDM billing platform  *
* update-source    * Updates all of the digium source, don't do        *
* update-scripts  * Installs latest scripts, ok every so often     *
* update-fixes    * Fixes minor PIAF probs until next ver, every so oft *
* install-key     * This lets you generate a new secure server key   *
* install-munin   * This will install munin reports                  *
* install-ZAPHFC  * Install ISDN/RDSI - QuadBRI,DUOBRI & HFC Chipsets  *
* setup-mail      * Configure sendmail                             *
* setup-tftp      * Install and set up tftp server                     *
* asterisk -rvv   * Asterisk CLI Starts the asterisk CLI              *

(add v's after the r to turn up the verbose output (5 v's = 5 times the logging)
**Asterisk Command Line Interface**

The Asterisk CLI only supports Asterisk commands. You can get access to the Asterisk CLI to run Asterisk commands using these methods:

- Using the browser to the **PBX IP Address**, then **FreePBX Administration – Tools – Asterisk CLI**.
- Using the browser to the **PBX IP Address**, then **FreePBX Administration – Tools – Config Edit**.
- Using the browser to the **PBX IP Address**, then **Webmin – Others - Command Shell** and then run the `asterisk -rvvv` command.

- Using a SSH application like **PuTTY** to the **PBX IP Address** and then run the `asterisk -rvvv` command.
- Using a keyboard and monitor connected directly to the PBX and then run the `asterisk -rvvv` command.

<table>
<thead>
<tr>
<th>Asterisk Commands</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>amportal restart</td>
<td>To restart PiaF after configuration change without rebooting. However certain changes will need rebooting.</td>
</tr>
<tr>
<td>amportal stop / start</td>
<td>Hard reset of Asterisk</td>
</tr>
<tr>
<td>asterisk -rvvv</td>
<td>To get to asterisk CLI</td>
</tr>
</tbody>
</table>
| Disable default voicemail message | This works in custom context e.g.  
|                           |   • Voicemail(s2000@default) - will not play default message  
|                           |   • Voicemail (su2000@default) - will not play default message and instead play your unavailable custom message. |
| cat /pathname/filename    | Displays the contents of any text file                                     |
Commands for troubleshooting telephony hardware

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ztool</td>
<td>Shows the config status of all telephony cards in the system</td>
</tr>
<tr>
<td>ztcfg –v</td>
<td>Configures zaptel cards after modules load –v shows results</td>
</tr>
<tr>
<td>zttest</td>
<td>Used to test the PCI bus: results should be 99.97% or better</td>
</tr>
<tr>
<td>lszaptel</td>
<td>Shows the zaptel cards and channels</td>
</tr>
<tr>
<td>fxotune -i 1</td>
<td>Tune out echo on a 4 port analog card</td>
</tr>
<tr>
<td>ztmonitor 3 –v</td>
<td>Real time channel monitor for PSTN echo tuning (3=channel 3)</td>
</tr>
<tr>
<td>cat /proc/interrupts</td>
<td>list what hardware is using what IRQ (for troubleshooting hardware detection)</td>
</tr>
<tr>
<td>lspci</td>
<td>Shows all hardware found on the PCI bus</td>
</tr>
<tr>
<td>lsmod</td>
<td>Shows all modules (aka drivers in Windows) that are loaded</td>
</tr>
<tr>
<td>top</td>
<td>Live Process monitor - show CPU/memory usage (Ctrl/z to end)</td>
</tr>
</tbody>
</table>

Asterisk CLI Commands

You have to run these commands from the Asterisk CLI.

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>!</td>
<td>Execute a shell command</td>
</tr>
<tr>
<td>abort halt</td>
<td>Cancel a running halt</td>
</tr>
<tr>
<td>dialplan add extension</td>
<td>Add new extension into context</td>
</tr>
<tr>
<td>add ignorepat</td>
<td>Add new ignore pattern</td>
</tr>
<tr>
<td>add queue member</td>
<td>Add a channel to a specified queue</td>
</tr>
<tr>
<td>ael debug contexts</td>
<td>Enable AEL contexts debug</td>
</tr>
<tr>
<td>ael debug macros</td>
<td>Enable AEL macros debug</td>
</tr>
<tr>
<td>ael debug read</td>
<td>Enable AEL read debug</td>
</tr>
<tr>
<td>ael debug tokens</td>
<td>Enable AEL tokens debug</td>
</tr>
<tr>
<td>ael no debug</td>
<td>Disable AEL debug messages</td>
</tr>
<tr>
<td>ael reload</td>
<td>Reload AEL configuration</td>
</tr>
<tr>
<td>agent logoff</td>
<td>Sets an agent offline</td>
</tr>
<tr>
<td>agi debug</td>
<td>Enable AGI debugging</td>
</tr>
<tr>
<td>agi no debug</td>
<td>Disable AGI debugging</td>
</tr>
<tr>
<td>cdr status</td>
<td>Display the CDR status</td>
</tr>
<tr>
<td>database del</td>
<td>Removes database key/value</td>
</tr>
<tr>
<td>database deltree</td>
<td>Removes database keytree/values</td>
</tr>
<tr>
<td>database get</td>
<td>Gets database value</td>
</tr>
<tr>
<td>database put</td>
<td>Adds/updates database value</td>
</tr>
<tr>
<td>database show</td>
<td>Shows database contents</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>---------------</td>
<td>-------------------------------------------------------</td>
</tr>
<tr>
<td>database showkey</td>
<td>Shows database contents</td>
</tr>
<tr>
<td>debug channel</td>
<td>Enable debugging on a channel</td>
</tr>
<tr>
<td>debug level</td>
<td>Set global debug level</td>
</tr>
<tr>
<td>dnsmgr reload</td>
<td>Reloads the DNS manager configuration</td>
</tr>
<tr>
<td>dnsmgr status</td>
<td>Display the DNS manager status</td>
</tr>
<tr>
<td>dont include</td>
<td>Remove a specified include from context</td>
</tr>
<tr>
<td>dump agihtml</td>
<td>Dumps a list of agi command in html format</td>
</tr>
<tr>
<td>dundi debug</td>
<td>Enable DUNDi debugging</td>
</tr>
<tr>
<td>dundi flush</td>
<td>Flush DUNDi cache</td>
</tr>
<tr>
<td>dundi lookup</td>
<td>Lookup a number in DUNDi</td>
</tr>
<tr>
<td>dundi no debug</td>
<td>Disable DUNDi debugging</td>
</tr>
<tr>
<td>dundi no store history</td>
<td>Disable DUNDi historic records</td>
</tr>
<tr>
<td>dundi precache</td>
<td>Precache a number in DUNDi</td>
</tr>
<tr>
<td>dundi query</td>
<td>Query a DUNDi EID</td>
</tr>
<tr>
<td>dundi show entityid</td>
<td>Display Global Entity ID</td>
</tr>
<tr>
<td>dundi show mappings</td>
<td>Show DUNDi mappings</td>
</tr>
<tr>
<td>dundi show peers</td>
<td>Show defined DUNDi peers</td>
</tr>
<tr>
<td>dundi show peer</td>
<td>Show info on a specific DUNDi peer</td>
</tr>
<tr>
<td>dundi show precache</td>
<td>Show DUNDi precache</td>
</tr>
<tr>
<td>dundi show requests</td>
<td>Show DUNDi requests</td>
</tr>
<tr>
<td>dundi show trans</td>
<td>Show active DUNDi transactions</td>
</tr>
<tr>
<td>dundi store history</td>
<td>Enable DUNDi historic records</td>
</tr>
<tr>
<td>extensions reload</td>
<td>Reload extensions and &quot;only&quot; extensions</td>
</tr>
<tr>
<td>feature show channels</td>
<td>Show status of feature channels</td>
</tr>
<tr>
<td>group show channels</td>
<td>Show active channels with group(s)</td>
</tr>
<tr>
<td>help</td>
<td>Display help list, or specific help on a command</td>
</tr>
<tr>
<td>include context</td>
<td>Include context in other context</td>
</tr>
<tr>
<td>indication add</td>
<td>Add the given indication to the country</td>
</tr>
<tr>
<td>indication remove</td>
<td>Remove the given indication from the country</td>
</tr>
<tr>
<td>init keys</td>
<td>Initialize RSA key passcodes</td>
</tr>
<tr>
<td>load</td>
<td>Load a dynamic module by name</td>
</tr>
<tr>
<td>local show channels</td>
<td>Show status of local channels</td>
</tr>
<tr>
<td>logger reload</td>
<td>Reopens the log files</td>
</tr>
<tr>
<td>logger rotate</td>
<td>Rotates and reopens the log files</td>
</tr>
<tr>
<td>logger show channels</td>
<td>List configured log channels</td>
</tr>
<tr>
<td>Command</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------</td>
<td>----------------------------------------------------------------</td>
</tr>
<tr>
<td>meetme</td>
<td>Execute a command on a conference or conferee</td>
</tr>
<tr>
<td>mgcp audit endpoint</td>
<td>Audit specified MGCP endpoint</td>
</tr>
<tr>
<td>mgcp debug</td>
<td>Enable MGCP debugging</td>
</tr>
<tr>
<td>mgcp no debug</td>
<td>Disable MGCP debugging</td>
</tr>
<tr>
<td>mgcp reload</td>
<td>Reload MGCP configuration</td>
</tr>
<tr>
<td>mgcp show endpoints</td>
<td>Show defined MGCP endpoints</td>
</tr>
<tr>
<td>mixmonitor</td>
<td>Execute a MixMonitor command</td>
</tr>
<tr>
<td>moh classes show</td>
<td>List MOH classes</td>
</tr>
<tr>
<td>moh files show</td>
<td>List MOH file-based classes</td>
</tr>
<tr>
<td>moh reload</td>
<td>Music On Hold</td>
</tr>
<tr>
<td>no debug channel</td>
<td>Disable debugging on a channel</td>
</tr>
<tr>
<td>pri debug span</td>
<td>Enables PRI debugging on a span</td>
</tr>
<tr>
<td>pri intense debug span</td>
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FreePBX Phone System Users Guide

From the Grandstream TCP/IP phones to other Grandstream phones

___ rings all on the PBX

xxx rings xxx extension

*80xxx rings xxx extension – AutoAnswer, where that extension’s speaker phone is activated

From the Grandstream TCP/IP phones, to make an outside call

The FreePBX system is connected to a line on the old PBX, so the first 9 dialed gets you a line on old PBX and second 9 is required to get an actual outside line. You do not wait for the dial tone, just enter the numbers and press SEND or wait.

Dial 9, and the number

NOTE: After dialing the number, press the SEND button or wait.

To transfer a call to another extension

Blind: Press TRNF, then dial the extension and press the SEND button.

Attended: Press HOLD, Dial the other extension, once the call is established, press “TRNF”

To transfer a call to an outside line

Blind: Press TRNF, then dial 9 and the extension and press the SEND button.

Attended: Press HOLD, Dial 9 and the extension, once the call is established, press “TRNF”
Conferencing

1. Initiate a Conference Call:
   - Establish a connection with two or more parties
   - Press CONF button
   - Choose the desired line to join the conference by pressing the corresponding LINE button.
   - Repeat step 2 and 3 for all parties that you want to join the conference.

   If after pressing the “CONF” button, a user decides not to conference anyone, press CONF again or the original LINE button. This will resume two-way conversation.

3. End Conference:
   Press HOLD to end the conference call and put all parties on hold;
   To speak with an individual party, select the corresponding blinking LINE.

   **NOTE:** The party that starts the conference call has to remain in the conference for its entire duration, you can put the party on mute but it must remain in the conversation.

Grandstream TCP/IP phone functions

Press the MUTE/DEL button to go to Do Not Disturb mode. The Do Not Disturb icon flashes in the display.

Press SEND to redial the last number.

To transfer a call, just press TRNF, dial the number and press SEND.
Voice Mail

The red light on the right of the phone will flash when you have voice mail. I can arrange to have it emailed to you to play over your PC speaker(s).

To check voice mail from your own extension:

*97 (or *98 and * when it answers) then your extension and then press # or SEND.

To check voice mail for another extension:

*98 and when it answers enter the extension press SEND, and then the extension password and then press # or SEND.

When in Voice Mail

0 = mailbox options
   1 = record your “unavailable message”
   2 = record your “busy message”
   3 = record your name
   4 = record your temporary message
   5 = change your password
   * = return to main options

1 = listen to new message

2 = change to another voice mail extension

3 = advanced options
   1 = send a reply
   3 = listen to message envelope
   * = return to main options

4 = previous message

5 = repeat

6 = next message

7 = delete message

8 = forward message

9 = save message

* = help

# = exit
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<td><strong>Connectivity Status / SIP Proxy/Server Icon:</strong></td>
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<td><strong>Solid</strong> – connected to SIP Server/IP address received</td>
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<td><strong>OFF</strong> when the speakerphone is off</td>
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<td><strong>DND Icon:</strong></td>
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Key Call Features

*30 Block Caller ID (for all subsequent calls)
*31 Send Caller ID (for all subsequent calls)
*67 Block Caller ID (per call)
*82 Send Caller ID (per call)
*50 Disable Call Waiting (for all subsequent calls)
*51 Enable Call Waiting (for all subsequent calls)
*70 Disable Call Waiting (per Call)
*71 Enable Call Waiting (per Call)
*72 Unconditional Call Forward
   Dial "*72" for a dial tone.
   Dial the forwarding number followed by "#". Wait for dial tone.
   LCD will display “Call FWD Activated”.
*73 Cancel Unconditional Call Forward
   Dial "*73" and get the dial tone, then hang up.
   LCD will display “Call FWD Activated”.
*90 Busy Call Forward
   Dial "*90" for a dial tone.
   Dial the forwarding number followed by "#".
   Wait for a dial tone. Hang up.
*91 Cancel Busy Call Forward
   Dial “*91”. Wait for dial tone. Hang up.
*92 Delayed Call Forward
   Dial "*92" for a dial tone.
   Dial the forwarding number followed by "#".
   Wait for a dial tone. Hang up.
   LCD will display “Call FWD Activated”.
*93 Cancel Delayed Call Forward
   Dial "*93" for a dial tone, then hang up.
Glossary

**A2Billing** – A2Billing is an add-on to Asterisk project that allows calling card services and billing. It is not supported by Ringdale, but you can use it with the Ringdale PBX.

**ACD** – Automatic Call Distribution – The module that routes incoming calls.

**ADSI** – Analog Display Services Interface is a complex set of standards for the telecom industry. Built off of FSK keying used by Caller ID, ADSI is capable of remotely controlling a screenphone with softkeys.

Asterisk has ADSI support, but it is not well documented. It is not supported unless you purchase the "supported" phone from Ringdale.

**AEL** – Asterisk Extension Language

**AGI** – Asterisk Gateway Interface is an advanced scripting interface that allows you to do thing like query an external database. AGI scripting, like CGI scripts, can be done in almost any language. Commonly used AGI scripting languages include C, Perl, Python, and PHP.

**Alaw** – One of two different file format variants of the G.711 codec used for telephony. It is used mainly on European E1 connections. See ULAW.

**AMI** – The Asterisk Manager Interface allows a client program to connect to an Asterisk instance and issue commands or read events over a TCP/IP stream. Integrators will find this particularly useful when trying to track the state of a telephony client inside Asterisk, and directing that client based on custom (and possibly dynamic) rules.

**AMP** – Asterisk Management Portal – Used to configure the asterisk server. AMP was renamed to FreePBX and introduced the first generation of modular architecture present in the 2.x version.

**Amporal** – the original name for Asterisk

**Apache** – The Apache HTTP Server Project is an effort to develop and maintain an open-source HTTP server for modern operating systems including UNIX and Windows NT. The goal of this project is to provide a secure, efficient and extensible server that provides HTTP services in sync with the current HTTP standards. http://httpd.apache.org/

**ARI** – Asterisk Recording Interface is used to monitor the recorded conversations. Note that in some places these may be illegal.

**AsteriDex** – This is a free web-based Asterisk database speed-dialer utility.

**Asterisk** – Asterisk™ is a Linux based IPBX application developed by Mark Spencer of Digium™, the company behind Asterisk. Asterisk™ is a trademark of Digium, Inc. Digium™ is a trademark of Digium Inc.

**Authenticate Password** – This parameter on the Grandstream GXP2000 phone must match the extension secret.

**BLF** – Busy Lamp Field

**BRI** – Basic Rate Interface is a kind of Integrated Services Digital Network channel consisting of two 64 Kbit per second "bearer" (B) channels for user-data transfer plus one 16 Kbps "delta" (D) channel for control and signalling information.

**CAPI** – Computer Assisted Personal Interviewing.

**CentOS** – CentOS® is a version of Linux related to a very well known Enterprise Linux (but without the branding and support).

**CDR** – Call Detail Records

**CFB** – Call Forward Busy
CFU – Call Forward Unavailable

ChanSpy – Listen in on a call. It is primarily useful in a call center to monitor agents on the phone.

CID – Caller Identification or Caller ID.

CLI – Command Line Interface

Codec – Short for COder/DECoder. A codec is a device or computer program capable of encoding and/or decoding a digital data stream or signal.

Comedian – The Asterisk Voice Mail System.

Context – A context is just a collection of extensions. In Asterisk, outgoing numbers are divided in groups called contexts in order to separate/define different needs for different user types. All calls begin in a certain context. For example, a context for local calls, another for within the city, and another for international calls and so on. Each context has one or more extension associated with it. Contexts also have priorities and applications (such as answer, playback, and hangup).

See http://www.voip-info.org/wiki/view/Asterisk+Dialplan+Introduction

Core module – The core covers your basic 'Extensions' and 'Trunks' etc. It is always enabled during normal operation.

CPC --Calling Party Control – is a signal sent from most modern electronic COs to indicate that the “Calling Party” has hung up. It’s usually called "Open Loop Disconnect" when you’re programming telephone equipment.

CRM – Customer Relationship Management

CVS – Concurrent Versions System this is a central repository used to control the source code.

Daemon – A daemon is a type of program on Unix-like operating systems that runs unobtrusively in the background, rather than under the direct control of a user, waiting to be activated by the occurrence of a specific event or condition.

DAHDI – Digium Asterisk Hardware Device Interface. Formerly called Zaptel, it is the hardware driver for the Digium Interface Card.

DID – Direct Inward Dialing – This is a service offered by telephone companies which allows the last 3 or 4 digits of a phone number to be transmitted to the destination exchange.

For example, a company could have 10 incoming lines, all with the number 234 000. If a caller dials 234 697, the call is sent to 234 000 (the company's exchange), and the digits 697 are transmitted. The company's exchange then routes the call to extension 697. This gives the impression of 1000 direct dial lines, whereas in fact there are only 10. Obviously, only 10 at a time can be used.

DISA – Direct Inward System Access – allows someone calling in from outside the telephone switch (PBX) to obtain an "internal" system dial tone and dial calls.

Distro – A large collection of Linux software applications built on top of the Linux kernel.

DND – Do Not Disturb

DOVM - Data Over Voice Multiplexer

DTMF – Dual-Tone Multi-Frequency – Push-button or touch tone dialing. Dual-tone multi-frequency (DTMF) signalling is used for telecommunication signalling over analog telephone lines in the voice-frequency band between telephone handsets and other communications devices and the switching center.

DUNDi – Distributed Universal Number Discovery is a VoIP routing protocol that provides directory services similar to what is provided by ENUM.
**ENUM – E.164 NUmber Mapping** is the most prominent facility for telephone number mapping. It uses special DNS record types to translate a telephone number into a Uniform Resource Identifier or IP address that can be used in Internet communications.

**E.164** – An ITU-T recommendation which defines the international public telecommunication numbering plan used in the PSTN and some other data networks. It also defines the format of telephone numbers.

**Fail2Ban** – An intrusion monitor and prevention framework for IP security.

**Feature Code Admin** – For configuration of ‘call features’, such as DND and Call Forwarding.

**Follow Me** – This is a function where you can forward calls to other pre-setup extensions if you do not answer. For example you can forward an unanswered call to your cell phone.

**FOP – Flash Operator Panel** is a switchboard type application for the Asterisk PBX. It runs on a web browser. It is able to display information about your PBX activity in real time. [http://www.asternic.org/](http://www.asternic.org/)

**FreePBX™** – FreePBX is a web-based GUI interface and configuration file generator that PiaF uses to manage Asterisk, the IP PBX. It is a registered trademark of Atengo LLC. The original name was AMP.

**FSK – Frequency Shift Keying** is a simple digital modulation technique that uses two frequencies for 0 and 1.

**FXO – Foreign eXchange Office**

When a customer receives phone service from a central office other than the one that would normally serve them, the line between the customer and the “Foreign” office is called a “Foreign Exchange” line and FXS (Foreign eXchange Station) is the station end. FXO (Foreign eXchange Office) is the office end of the line. FXO is also used to refer to the type of interface on phone equipment. An FXO interface receives power and ring signals. An FXS interface provides power and ring signals. If you want to connect your phone line to your computer so that it can make and answer calls, you need to add an FXO interface to your computer. If you want to connect an ordinary telephone to a computer, you need a card in the computer with an FXS interface.

An FXO device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like one. They should also operate the same way when connected to an FXS interface.

An FXO interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.

**FXO** is complimentary to **FXS** (and the PSTN).

**FXS – Foreign eXchange Station** An FXS device has hardware to generate the ring signal to the FXO extension (usually an analog phone).

An FXS device will allow any FXO device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.

The FXS Interface connects to FXO devices (by an FXO interface, of course).

**Gabcast** – A hosted service that lets you record your phone or VoIP to create podcasts and then post them to your blog or website.

**GSM – Global System for Mobil Telecommunications** – GSM codec recording. As with WAV49 calls, the quality of GSM recordings is less than that of ULAW/ALAW or WAV calls, but is generally acceptable for most purposes. GSM recordings weigh in at around 100 kilobytes per minute.

**G.729** – An audio data compression algorithm for voice that compresses digital voice in packets of 10 milliseconds duration. It is officially described as Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP). Because of its low bandwidth requirements, it is mostly used in Voice over Internet Protocol (VoIP) applications where bandwidth must be conserved.
GUI – Graphical User Interface.

H.323 – H.323 is an umbrella Recommendation from the ITU Telecommunication Standardization Sector (ITU-T) that defines the protocols to provide audio-visual communication sessions on any packet network. The H.323 standard addresses call signalling and control, multimedia transport and control, and bandwidth control for point-to-point and multi-point conferences.

.htaccess (hypertext access) is the default name of a directory-level configuration file that allows for decentralized management of web server configuration.

IAX – Inter-Asterisk eXchange is a protocol native to Asterisk PBX and supported by a number of other softswitches and PBXs. It was developed as an alternative to SIP and H.323. There are two versions of IAX, where IAX2 is the most common used. IAX is not submitted by any standards group, but it is being adopted by different manufacturers for soft- and hard-phones. The biggest advantage to IAX is that it uses only one UDP port and thus works very well behind NAT firewalls.

IPBX – stands for IP Private Branch Exchange or International Private Branch Exchange (ITU-T) also iPBX

Ip6tables – IPv6 Firewall for Linux -- Ip6tables is used to set up, maintain, and inspect the tables of IPv6 packet filter rules in the Linux kernel.

IRC – Internet Relay Chat is a form of real-time Internet text messaging (chat) or synchronous conferencing.

ISDN – Integrated Services Digital Network – ISDN is a circuit-switched methodology using digital facilities. This system allows data to be transmitted simultaneously across the world using end-to-end digital connectivity.

ITU-T – The Telecommunication Standardization Sector coordinates standards for telecommunications on behalf of the International Telecommunication Union (ITU)

IVR – Interactive voice response is a dialog system technology that allows a computer to detect voice and keypad inputs. IVR technology is used extensively in telecommunications, but is also being introduced into automobile systems for hands-free operation.

Java SSH – This application sets up an SSH session to the PBX Server. It is the same as using PuTTY.

Kernel – The kernel is the essential center of a computer operating system, the core that provides basic services for all other parts of the operating system. A synonym is nucleus. A kernel can be contrasted with a shell, the outermost part of an operating system that interacts with user commands.

KSU – Key System Unit is a PBX that acts as a central control unit, providing features and functions.

LAMPA – Linux, Apache, MySQL, PHP and Asterisk – FreePBX is built on the LAMPA™ stack.

Linux – Linux is a Unix-like operating system that was designed to provide personal computer users a free or very low-cost operating system.

Meet Me – Web Meet Me Control, a meet me conferencing control application.

MGCP – Media Gateway Control Protocol is used for controlling Media Gateways on Internet Protocol (IP) networks and the public switched telephone network (PSTN).

MPLS – Multiprotocol Label Switching provides traffic isolation and differentiation without substantial overhead.

MPLS VPN is a family of methods for harnessing the power of Multiprotocol Label Switching (MPLS) to create Virtual Private Networks (VPNs).

MP3 – Moving Picture Experts Group Layer-3 – Audio (audio file format/extension)
MSN – Multiple Subscriber Line – This is a telephone number associated with an ETS 300 BRI line. Providers of ETS 300 often give you three MSNs with a BRI, although additional MSNs can be purchased. An ISDN terminal will “ring” (provide an alerting signal) only when calls are made to the MSN (or MSNs) entered in that terminal. If a terminal has no MSNs entered it will “ring” whenever there is a call to any of the MSN.s on that BRI.

MWI – Message Waiting Indicator

MySQL – This a database server that supports standard SQL and is ideal for both small and large applications

NAT – Network Address Translation – A firewall process of modifying network address information in datagram packet headers while in transit across a traffic routing device for the purpose of remapping a given address space into another.

OSLEC – Oslec is an open source high performance line echo canceller. When used with Asterisk it works well on lines where the built-in Zaptel echo canceller fails. No tweaks like rxgain/txgain or fxotrain are required. Oslec is supplied as GPL licensed C source code and is free as in speech.

PBX – Private Branch Exchange is a smaller version of a phone company’s large central switching office.

PCM – Pulsed Code Modulation

PHP – Personal Home Page

PHP – PHP: Hypertext Preprocessor

PHPAGI – This the PHP class for the Asterisk Gateway Interface.

PHPInfo – A function that returns information, in HTML form, about the PHP environment on your server (see http://us2.php.net/phpinfo for more information)

phpMyAdmin – This is a tool written in PHP intended to handle the administration of MySQL over the Web. Currently it can create and drop databases, create/drop/alter tables and views, delete/edit/add fields, execute any SQL statement, manage keys on fields, manage privileges, and export data into various formats.

PiaF – PBX in a Flash is an integration of Asterisk and a collection of telecommunication utilities and tools compiled together to become an integrated IP PBX. PiaF uses FreePBX as the GUI for Asterisk, Webmin as the GUI for the OS, and Fail2Ban is preconfigured, all you need to do is simply download, install, update and configure.

POTS – Plain Old Telephone System is sometimes used as a synonym for PSTN (public switched telephone network). However, the latter usually has a broader meaning: it refers to the worldwide collection of interconnected public telephone networks that were designed primarily for voice traffic but which now use high speed digital links for nearly all of their trunk and intermediate lines and which are carrying an increasingly large share of non-voice traffic.

PRI – Primary Rate Interface

PRI is a standard network interface consisting of 1 D channel and 23 B channels on a T1, or 1 D channel and 30 B channels on a E1.

PSTN – Public Switched Telephone Network

PuTTY – PuTTY is a free implementation of Telnet and SSH for Win32, UNIX, and Linux platforms, along with an xterm terminal emulator. It is written and maintained primarily by Simon Tatham. You can download a free copy of PuTTY from http://www.putty.nl/download.html.

regex – regular expressions are strings of text, particular words, or patterns of characters.

Ring Group – The Ring Group defines what extensions will ring when an incoming call comes in. The Ring Group can be defined to ring all defined extensions or in a particular order. The options are:
Ringall, Hunt, MemoryHunt, Ringall-Prim, Hunt-Prim, MemoryHunt-Prim, FirstAvailable, FirstNotOnPhone.

**RTCP – The RTP Control Protocol** is a sister protocol of the Real-time Transport Protocol (RTP). Its basic functionality and packet structure is defined in the RTP specification RFC 3550.

**RTP – Real-Time Transport Protocol**, an Internet protocol for transmitting real-time data such as audio and video. RTP itself does not guarantee real-time delivery of data, but it does provide mechanisms for the sending and receiving applications to support streaming data. Typically, RTP runs on top of the UDP protocol, although the specification is general enough to support other transport protocols.

**Screenphone** – Combined voice and textphone, which allows the user to speak and then read the reply on a screen.

**SIP – Session Initiation Protocol** is an IETF proposed standard for setting up sessions between one or more clients. It uses port 5060 over UDP and TCP, but may use other ports. It is currently the leading signalling protocol for Voice over IP, gradually replacing H.323. Most VoIP devices support it. Other protocols that Asterisk supports are IAX, H.323, and CAPI.

**sln – Signed Linear** – This is the audio format that is native to Asterisk. It is Raw Signed Linear Audio. Recordings that are in SLN format will have the same quality and file size as WAV recordings. SLN recordings are raw WAV, little endian 16-bit signed linear (PCM) format recordings. Most computers will play these files, although some software packages refuse to play them unless the extension is renamed to .wav from .sln.

**Softphone** – This is a software program for making telephone calls over the Internet using a general purpose computer, rather than using dedicated hardware. There are a number of softphones available for use with PiaF. Some of them are free.

**Softswitch** – A softswitch is a central device in a telecommunications network which connects calls from one phone line to another, entirely by means of software running on a computer system. This work was formerly carried out by hardware, with physical switchboards to route the calls.

**strong password** – A password that is difficult to detect by both humans and computer programs, effectively protecting data from unauthorized access. A strong password consists of at least six characters (and the more characters, the stronger the password) that are a combination of letters, numbers and symbols (@, #, $, %, etc.) if allowed. Passwords are typically case-sensitive, so a strong password contains letters in both uppercase and lowercase. Strong passwords also do not contain words that can be found in a dictionary or parts of the user’s own name.

**SQL – Structured Query Language** is a database computer language designed for managing data in relational database management.

**SRV – An SRV record or Service record** is a category of data in the Internet Domain Name System specifying information on available services. It is defined in RFC 2782. Newer internet protocols such as SIP often require SRV support from clients.

**STUN – Session Traversal Utilities** for NAT

**SugarCRM** – This is an open-source software-solution vendor which produces the Sugar Customer Relationship Management (CRM) system. It is not supported by Ringdale, but you can use it with the Ringdale PBX.

**System Dashboard** – System Status Dashboard is a user interface that displays information from the operating system. It shows FreePBX System Status, FreePBX Notices, Statistics, and processor information.

**Tango** – The red-eyed tree frog is the FreePBX logo image.

**TDM – Time Division Multiplexer** is a device that supports simultaneous transmission.
of multiple data streams into a single high-speed data stream. TDM separates signals by interleaving bits one after the other.

**Trunk** – A circuit which connects the PBX to the local telephone company's switching center. A trunk consists of two channels. So one BRI or ISDN line = one trunk, which is made up of two channels. So 4 BRI lines = 4 Trunks, total 8 channels. The first call to go out, will take trunk one channel 1, and the second call to go out will take trunk one channel 2, etc.

**TTS – Text To Speech**

**ULAW** – This is one of two different variants of the G.711 codec. This is the International standard CCITT version. See Alaw. G.711 codec recording. The recording quality is excellent, and should sound exactly like the call did to all of the parties who were on the original call. File size is very large (similar to the WAV format at about 1 megabyte per minute). ULAW and ALAW recordings are very difficult to play on most computers. There are very few computers that will play the recording without additional software that understands the G.711 codec.

**VMB – Voice Mail Blasting** – An option to send a voice mail message to a group of pre-defined extensions.

**VmX – Voice Message Exchange**

**VoIP – Voice over IP.** In more common terms, phone service over the Internet. VoIP encompasses many protocols. All the protocols do some form of signalling of call capabilities and transport of voice data from one point to another. Examples are SIP, H.323, IAX, and IAX2.

**VSP – VOP Service Provider**

**WAV – Waveform Audio File Format** - Uncompressed WAV format recording. Sound quality will be very good, but the file will be very large in size (roughly 1 megabyte per minute of the recording). WAV format recordings are natively playable on nearly all of the computers without additional software.

**WAV49** – WAV format recorded using the GSM codec. As GSM is a compressed codec, the sound quality is compromised. Sound quality on a GSM recording is usually equivalent to the quality that is achieved during mobile telephone calls. File size is much smaller than a standard WAV (roughly 100 kilobytes per minute of the recording); WAV49 files are often difficult to play on computers without additional software that understands the GSM codec.

**Webmin – Web-based administration** toolkit is a web-based interface for system administration for Linux.

**ZAP channels** – The Asterisk Zap Channel Module provides an interface layer between Asterisk on the one side, and the Zaptel interface drivers on the other side. These drivers, in turn, provide the ability to use interface cards to connect your PBX to traditional digital and analog telephone equipment:

Asterisk <-> chan_zap.so <-> zaptel.ko (kernel) <-> device driver <-> Zaptel device <-> Phone/switch/PSTN

**ZapBarge** – ZapBarge(channel) Lets you listen to the conversation on a specified Zap channel. Multiple people can all use ZapBarge to listen in on the same channel.

**Zaptel** – This is the hardware driver for the interface card that connects telephone lines and/or telephone handsets to your PBX. The Ringdale PBX uses the Digium Digital Interface Cards. The Zaptel project has been renamed ‘DAHDI’.
References

The Ringdale PBX is based on a 1.5 GHz CentOS release 5.2 (Final) - 32 Bit ** Kernel: 2.6.18-92.1.6.el5 Linux platform with FreePBX software & hardware custom-installed. It has a Digium 4-port Analog card installed for access to the outside world.

The following reading is highly recommended:

PBX in a Flash


PBX in a Flash® without Tears: http://dumbme.mbit.com.au/PiaF/PiaF_without_tears.pdf

PBX in a Flash Forum: http://pxxinflash.com/forum

A short dynamic presentation on the PBX in a Flash System Summary. Basically how Linux and Asterisk fit together, config files and misc things. Consider it an introduction to PiaF.
http://prezi.com/6v_swrcu4dh4/

FreePBX


FreePBX documentation: http://freepbx.org/support/documentation

FreePBX module documentation: www.freepbx.org/support/documentation/module-documentation


Asterisk

Theory outline for Asterisk beginners: voip-info.org/wiki/view/Asterisk+Primer

The little Asterisk handbook: http://www.automated.it/asterisk/lah-3-6-05_1.html

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

For text file asterisk users: www.asterisk.org

Video tutorials for Asterisk: http://www.asterisktutorials.com

VOIP

VOIP Wiki - a reference guide to all things VOIP: http://www.voip-info.org/wiki/

VoIP Glossary: http://www.voxgratia.org/docs/glossary.html

Flash Operator Panel: http://www.asternic.org/


On testing Zap: http://www.cadvision.com/blanchas/Asterisk/TestingZapte1HW.html
On passwords:
Changing the MySQL password
http://www.freepbx.org/support/documentation/faq/changing-the-mysql-password
Asterisk Handbook:
    http://www.voip-info.org/wiki/view/Asterisk@Home+Handbook+Wiki+Chapter+3
PBX in a Flash for Newbies - Users and Passwords
http://www.cadvision.com/blanchas/Asterisk/Passwords.html
Trixbox Open Platform for Business Telephony
http://www.trixbox.org/forums/trixbox-forums/open-discussion/freepbx-password-reset

Grandstream phone support:
http://www.grandstream.com/faqs.html
http://www.asteriskguru.com/tutorials/gxp2000_grandstream_hardphone.html

Support Forums:
Very useful forum: http://pbxinaflash.com/forum
The FreePBX Forum http://www.freepbx.org/forums
Very good site with definitive guide to getting a system going: http://nerdvittles.com/

IMPORTANT NOTE: When you post a question or problem in a forum, you should include:

✓ A short paragraph about the problem you are having.
✓ A copy of the output of the Linux `status` command. This will help those in the know a lot and it will go a long way to narrowing down your particular version. It provides:
    PBX in a Flash Version 1.3
    Operating system (CentOS release 5.2)
    Asterisk Version 1.4.21.2
✓ FreePBX Core Version (from Module Administration)
✓ What processor/motherboard/amount of ram:
    CPU & Companion Chips VIA C7 1.5GHz + CN700
    Memory 1GB DDRII (SO-DIMM)
✓ Have you run update-scripts, update-fixes, and update-source?
✓ Relevant logs only! Don't post full logs into the forum! Post only snippets of long log files. If there is not enough info people will make some suggestions.
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