CompletePBX v3

Reference Guide

(rev. 3.07)

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1. Overview

Designing CompletePBX

Extensions (or Devices and Users)

Extensions are where you set-up devices (telephones) and users (extensions) on your system. Review Chapter 2, **Devices and Extensions** to get a better understanding of the differences between Extensions and DeviceAndUser modes, and how to choose the correct mode for your system.

More detailed information can be found in Chapter 2, **Devices and Extensions**.

Define your numbering system

You need to decide how many digits to use for extensions – do you want to use 3, 4, or more. You should take into account that most feature codes are 2 digits, so setting a system with 2-digit extensions is not practical. It will help you to navigate your system if you group similar functions together, for example:

- 2000 for extensions
- 8000 for queues
- 9000 for conferences

Create a flow chart

It will help you to plan and visualize your CompletePBX configuration if you create flow-chart of everything that you want.

Trunks

Trunks are the PBX equivalent of a phone line. They are how your system makes calls to the outside world and receives calls from the outside world. Without a trunk, you can't call anyone. You can configure a trunk to connect with any VOIP service provider, with a PSTN/Media Gateway (which allows you to make and receive calls over standard telephone lines from your local telephone company), or to connect directly to another PBX.

Before you start to configure your CompletePBX, it is important that you decide what kind of trunks you will be using – SIP, DAHDI, etc.

Most reputable VOIP providers will give instructions on how to configure a CompletePBX trunk with their service.

The Dialed Number Manipulation Rules section lets you redirect calls to certain numbers to other numbers. For example, if someone dials **411**, CompletePBX can be configured to change that to **1-800-FREE-411.** Or you could make **611** call your grandmother.

More detailed information can be found in Chapter 3, **Trunks.**

Outbound Routes

Outbound Routes are how you tell CompletePBX which Trunks (phone lines) to use when people dial external telephone numbers. A simple installation will tell CompletePBX to send all calls to a single

trunk. However, a complex setup will have an outbound route for emergency calls, another outbound route for local calls, another for long distance calls, and perhaps even another for international calls. You can even create a "dead trunk" and route prohibited calls (such as international and 976 calls) to it.

More detailed information can be found in the *Outbound Routing and Least Cost Routing* section of Chapter 6, **Call Routing**.

Inbound Routes

The Inbound Routes module is where you tell CompletePBX how to handle incoming calls. Typically, you tell CompletePBX the phone number that outside callers have called (DID Number) and then indicate which extension, Ring Group, Voicemail, or other destination the call will go to.

More detailed information can be found in Chapter 6, **Call Routing.**

Follow-me

The follow-me module allows you to create a more complicated method of routing calls that are placed to a specific extension. Using this module, you can cause a call to an extension to ring several other extensions, or even outside phone numbers. You can also make calls to an extension terminate in the voicemail of another extension.

For example, using "follow-me," you could make a call to extension 2009 actually ring extensions 2009, 2010, and extension 2011, and call someone's cellular phone, for 15 seconds, and then, if nobody answers, go the voice mailbox for extension 2003.

More detailed information can be found in the *Follow Me and the VmX Locater* section of Chapter 6, **Call Routing**.

Ring Group

Ring Groups allow you to create a single extension number (the Ring Group number) that will call more than one person. For example, you could make a Ring Group so that when any user dials extension 601, extensions 2009, 2010, and 2011 ring for 15 seconds, and then the call goes to the voicemail for extension 2000.

More detailed information can be found in the *Ring Groups* section of Chapter 4, **Basic Call Targets**.

Parking Lot

A parking lot allows anyone who has received a call to park the call on an extension that anyone else can access. Typically, you receive the call, transfer it to extension 70, and then listen as the system tells you where you can pick up the call (usually extension 71). Then, anyone else on your CompletePBX can dial 71 to pick-up the call.

More detailed information can be found in the *Parking* section of Chapter 5, **Advanced Call Targets**.

Paging and Intercom

The **Paging and Intercom** module allows you to define numbers you can dial to page a group of devices at once. For example, in a small office, you might define a paging group that allows any user to dial 00 to page the entire office. By default, you dial *80 followed by the extension number to intercom a specific user.

More detailed information can be found in the *Paging & Intercom* section of Chapter 5, **Advanced Call Targets**.

Conferences

The **Conferences** function allows you to create an extension number that people can dial into in order to have a conference call. For example, a number of users could dial extension 9000 and participate in a conference call.

More detailed information can be found in the *Conferences* section of Chapter 5, **Advanced Call Targets.**

IVR

IVR allows you configure an auto attendant to answer calls and direct them in response to input from the caller.

It is important that you carefully plan branching options for IVRs and the call flow for your IVRs, while carefully considering the user experience. IVR uses customized Announcements, so you will need to make sure that they are clear and meaningful, and configured to maximize the caller's experience. Factors that you should consider include:

- Handling the timeout when there is no input from the caller
- Controlling the action to take if caller provides invalid user input
- Do you want to take advantage of time-based branching, by defining Time Groups, for normal office hours, that include start and end times, start and end days of the week, and much more.
- You can define a Time Condition, and set one destination if the time matches and a different destination if the time does not match.

More detailed information can be found in the *IVR (Digital Receptionist)* section of Chapter 5, **Advanced Call Targets**.

Queues

A queue consists of incoming calls that need to be answered in an organized manner. Queues allow you to create rules to help you manage the flow of incoming calls.

Call queues are especially useful for scenarios in which the volume of callers is expected to exceed the number of people or agents that are available to answer calls. Scenarios such as a technical support line or a sales line are good examples.

There are many parameters that can be configured to customize queue-handling to meet your specific needs, including:

- Sensitive queues can be password-protected
- Users can be given feedback of their position in the queue and their estimated hold time
- Queues can have both static and dynamic agents
- Multiple ring strategy options are available
- Calls can be redirected to any destination if no agent is available to answer the call

Important note: if you want the Switchboard to track the status of queues and to create statistics for call center reporting, make sure that both **Event When Called** and **Member Status Event** (in the **Events, Stats and Advanced** section of the Queue dialog) are set to **Yes** when defining a new queue

Events, Stats and Advanced

Event When Called: ❷   Yes ▾
Member Status Event: ❷   Yes ▾

More detailed information can be found in the *Queue* section of Chapter 5, **Advanced Call Targets**.

Feature Codes

This module allows you to set the special codes that users dial to access various features. You can also disable features if you don't want users to be able to access them.

More detailed information can be found in the *Customizing feature codes* section of Chapter 8, Personalizing Your PBX. A detailed list of all feature codes can be found in Chapter 13, **Feature Codes.**

General Settings

This module has several important features you may want to consider changing:

The **Dial Voicemail Prefix** setting allows you to directly dial an extension's voice mailbox by dialing the prefix listed on this page. So, for example, if you dial * (the default) and then an extension number, you can skip ringing the extension and go straight to their mailbox. This is useful when you wish to transfer someone directly to voicemail. If you leave this at the default of *, and you use two digit extensions, the direct dial voicemail function will conflict with certain feature codes. You may wish to change the direct dial voicemail prefix from the default of * to something else, such as *86.

Optional Voicemail Recording Gain allows you to control the volume level of voicemail messages. If you find that the voicemail messages you receive are quiet compared to the system recordings, you might want to change this from the default of 0 to 5.

Checking the **Do Not Play** box on will generate a message to the caller to leave a message after the tone. If you want CompletePBX to just play the outgoing message and then beep, then check this box on.

Use **Operator Extension** to specify the extension number or ring group number that people will get transferred to when they dial 0 while leaving a voicemail. If you leave this blank, the caller will return to whatever ring group they came from before reaching voicemail.

System Recordings

System Recordings allows you record the messages for use on your auto-attendant (IVR).

More detailed information can be found in Chapter 8, **Personalizing Your PBX.**

Direct Inward System Access (DISA)

DISA allows you to create a destination that allows people to call in to from an outside line and reach the system dial tone. This is useful if you want people to be able to take advantage of your lower rate for toll calls, or if you want outside callers to be able to use the paging or intercom features of the system. Always *password protect* this feature, if you use it at all.

More detailed information can be found in the **Direct Inward System Access (DISA)** section of Chapter 8, **Personalizing Your PBX.**

Backup and Restore

Backup and Restore allows you to backup and restore the CompletePBX settings as well as recordings made by CompletePBX. After you have created a backup, you can find it by typing the following at the command prompt:

    cd /var/spool/asterisk/backup
    ls -l

More detailed information can be found in the **Backups** section of Chapter 9.

Accessing CompletePBX

Login to the GUI

Using another machine on your network, open a Web browser and enter the IP address of your CompletePBX machine. If you don't know the IP address of your CompletePBX machine, go to the Linux console/command prompt. You can login here using the username **root** and the root password (The password is documented in the Getting Started Guide, which was packed with your CompletePBX hardware). After you login, type **ifconfig** at the command line and determine the IP address of your machine (to the right of eth0).

CompletePBX Launch Pad

The Launch Pad (the main CompletePBX screen) offers you four options:

- **Administration** will allow you to configure your PBX. The default username is admin and the default password should set during the installation.
- **My Extension** allows users to listen to their voicemail messages and change certain features of their extension, such as call forwarding. Users login using their extension number and voicemail password, and can only modify features of their own extension. Users cannot see recordings or setup for any other extension other than their own.
- **My Switchboard** is a screen that allows a call center manager or an office manager to view and manage calls, queues, and many other functions.
- **Call Center Statistics** displays call center statistics, either on the user's screen, or in pdf format.

Module Maintenance

On its own, the core of CompletePBX is just a framework or wrapper which adds functionality, features, bug fixes and stability to the open source PBX software (Asterisk and FreePBX) and CompletePBX. From inbound routing to voicemail, this architecture provides a pluggable module framework that allows expansion of your CompletePBX system while maintaining supportability.

CompletePBX has tested the integration of more than fifty modules, each providing a specific set of functions. When administering a CompletePBX system, Apache serves as a Web interface for each module. The modules store their configuration in a database, which the framework processes and then writes out to the CompletePBX configuration files. The configuration files are read by system, making the configuration from the CompletePBX modules live. This process is shown in the following figure:
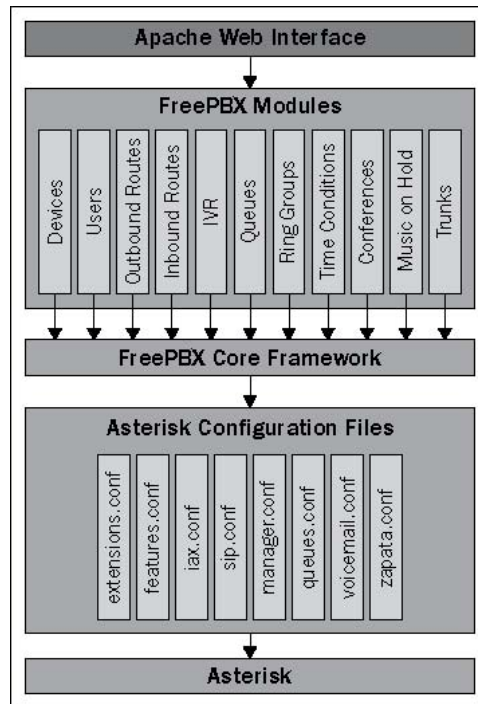
Everything in CompletePBX is a module, including the base components - Core Framework, Localizations, Recording Interface, and the Switchboard (Flash Operator Panel). This modularized structure is how new features are added to the system, as well as how updates are installed. These modules are from the core CompletePBX release or 3rd party modules that add additional functionality.

Splitting functionality into modules also allows each installation of CompletePBX to be built specifically for its deployment requirements. If a PBX simply needs to store voicemail for users, all it will require is the inbound routing and voicemail modules. A PBX that answers calls and then routes them to another PBX may just require the inbound routing and outbound routing modules. Installing just the required modules simplifies administration and improves startup and reload times.

Modules can specify dependencies, allowing each module to be as small and efficient as possible, by building on the shoulders of previously created modules. All of the modules will generally have at least one dependency.

Updating from the online repository

You must not update the software using either the Module Admin dialog or the System Admin dialog. Instead, the update procedure must be performed according to Xorcom instructions published at wiki.xorcom.com.

The next chapter will discuss configuring extensions and voicemail boxes.

2. Devices and Extensions

Much like traditional PBX systems, CompletePBX uses extensions to route calls to individual users. Each person utilizing CompletePBX will typically be assigned a single extension. Users can dial an extension to other users directly. Extensions can have voicemail boxes associated with them. They can be members of ring groups, call queues, and can be targets for direct inbound calls.

Rather than having an extension associated with a particular telephone, CompletePBX separates the actual telephones into a concept called *devices*. An extension may be mapped directly to a device in the traditional style, or a device may be independent of an extension and the extension that rings the phone may change based on who is using it. CompletePBX supports both of these methodologies. In this chapter, we will discuss:

- Operational modes for extensions
- Endpoint types
- Common fields
- Setting up a new extension in Extensions mode
- Setting up a new extension in DeviceAndUser mode
- Voicemail setup

Operational modes for extensions

CompletePBX has the following two different modes that can be used in order to configure extensions:

- Extensions
- DeviceAndUser

In the **Extensions** mode, each numerical extension that is configured within CompletePBX corresponds to a physical telephony endpoint. An endpoint can be a physical telephone, a softphone, an **Analog Telephony Adapter** (**ATA**), or anything else that talks to CompletePBX. In this mode, each endpoint is statically assigned to a particular device. This device never changes.

An example of this would be a typical cubical-office environment. Each cubical has a phone and the extension that rings the phone will never change. Even if the employee who works in a particular cubical is replaced, the extension that rings their phone will remain the same. Most end users find this setup intuitive, as it allows them to dial one number and their call to ring one phone. This mode is the easiest to set up, but offers less flexibility than the DeviceAndUser mode.

In the **DeviceAndUser** mode, the endpoints (devices) and extensions (users) are separate, allowing one numerical extension to correspond to an unlimited number of endpoints. The DeviceAndUser mode also allows a user to roam from endpoint to endpoint, logging in or out of a phone as their location changes.

An example of this might be a warehouse in which a manager needs to receive calls even while he is on the floor. The manager can select a cordless phone, log into it, and begin his rounds. Any calls to his extension will ring the cordless phone. When he has finished, he logs out of the phone and it no longer rings when his extension is called. This mode can be unintuitive to end users as dialing a person's direct extension number can result in several different phones ringing. This mode is very powerful, but it is more complicated to set up.

Each operational mode has its own strengths and drawbacks. As with most complex systems, there are multiple ways to accomplish almost any telephony task with CompletePBX, so ultimately the mode that you choose should be the one that seems to best suit your environment.

Operational modes can be changed any time, but doing so may mean that a significant amount of reconfiguration is required before the system is fully functional again. It is recommended that an operational mode be chosen prior to configuring any extensions in CompletePBX.

The following table lists some common scenarios that may assist in deciding upon the two operational modes for extensions:

Scenario	Operational mode to use
A large business where new employees assume the extension of a person who has left the company. Employees always work at the same telephone	Extensions
A small business where employees generally work at the same desk, but may opt to work from home. Employees should be able to make and receive calls from home	DeviceAndUser
A hotel where each room has a static extension associated with it. Regardless of who occupies the room, calling the room's extension will always ring the phone in the room	Extension
An executive office where users sit down at a desk on a first come, first serve basis. The phone on their desk needs to ring the extension that is assigned to the person	DeviceAndUser
A technical support call center where the phone at each station is only used for inbound calls. Employees are not assigned to a particular station, but selected randomly when they report to work	Extension
A marketing company where the call center is staffed 24 hours per day. Calls are outbound only, but it needs to track who made which calls. Employees are assigned an extension, and they log into their phone when they start their shift	DeviceAndUser

Choose extension numbers wisely

No matter which operational mode you choose, you must choose extension numbers. It is important to choose extension numbers carefully. Several extension numbers are in use by default in CompletePBX. Other extensions are very common and often used as initial weak points for malicious users attempting to gain access to your PBX. It is recommended that you use four-digit extensions if possible, and avoid extensions between 201 and 299 (which is the most common range, and therefore the most targeted). A common practice is to assign the last four digits of a DID as a user's extension number (for example, a DID of 555-555-5432 would ring extension 5432).

Endpoint types

CompletePBX allows four different types of endpoints to be configured:

- Session Initiation Protocol (SIP)
- Inter-CompletePBX eXchange 2 (IAX2)
- DAHDI
- Other (custom)

CompletePBX does not require a single type to be chosen for all of the endpoints; types can be mixed and matched. As long as the endpoint can communicate with CompletePBX using one of these protocols, CompletePBX will bridge calls between the devices.

SIP endpoints

SIP endpoints are the most common type of endpoints. Most VoIP hard-phones are SIP-based devices. Nearly all of the softphone endpoints support SIP. Choosing to utilize SIP endpoints provides the widest endpoint selection.

However, due to the large range of ports required for the **Real-time Transport Protocol** (**RTP**) audio stream, SIP can be problematic when traversing firewalls and other NAT devices. Common SIP configurations require 10,000 open ports to function correctly. Configuration can be particularly troublesome if both the endpoint and the CompletePBX system are behind their own firewalls.

IAX2 endpoints

The IAX2 protocol was designed with the shortcomings of SIP in mind and is most commonly used to link CompletePBX systems together. Several endpoints that support IAX2 are on the market, including several softphones and a few hardphones.

IAX2 uses a single port for call signaling and transmission of a VoIP audio stream. The choice of IAX2 endpoints can be very limiting, but additional firewall configuration is usually very simple (and often not required at all).

DAHDI endpoints

DAHDI devices are adapters for traditional two-wire **Plain Old Telephone Service** (**POTS**) handsets - the same handsets that are found in a home using the traditional telephone service. DAHDI endpoints are provided by Astribank FXS ports.

Custom endpoints

Custom endpoints can be based on any protocol supported by CompletePBX. Custom endpoints are commonly based on the **Media Gateway Control Protocol** (**MGCP**) and the H.323 protocol. As long as CompletePBX can communicate with the endpoint using the *Dial* command, CompletePBX will allow it to be used within its interface.

Endpoint fields

Several common configuration fields exist under both Extensions and DeviceAndUser modes. Regardless of the operational mode that is chosen for CompletePBX to operate in, these fields function in the same way. Use this section as a reference when setting up either operational mode. All of the common configuration fields and their purposes are listed here.

Common general Extension/User fields

User Extension should be a unique numerical value. If an extension that is already in use is entered,

CompletePBX will display a warning when we attempt to save the used extension, and the extension will not be saved. The User Extension number can be dialed from other endpoints attached to the PBX to call this user. This field is required.

Display Name is generally the end user's name or the location of the device (for example, "Alex Robar" or "Server Room"). This value will be displayed as the caller ID text for any calls placed from this user or device to other users or devices on the PBX. This field is required.

CID Num Alias will change the caller ID text displayed when this user calls other users on the PBX. This is commonly used when a user is part of a department in which callbacks should be directed to the department rather than directly to the user (such as a technical support department). This field is not required. If it is left blank, the user's extension will be used to set the **Outbound Caller ID** text.

SIP Alias is used to provide a friendly name for direct SIP calls made to this user. SIP aliases are most often used to distribute an address for contacting a particular user (similar to how email addresses are distributed to allow a user to be sent a message directly). A SIP alias will allow external callers to dial *alias@example.com* to call this particular user directly, without knowing that user's extension. Note that the SIP Alias field is shown for all types of extensions (SIP, IAX2, DAHDI, and Custom). Calls to a SIP alias must come from a SIP device, but CompletePBX will bridge the call to any type of endpoint that it is able to communicate with. This field is not required. If it is left blank, the only way to place a direct SIP call to the user will be the SIP URL *(extension)@example.com*, where (the extension) is replaced by the user's numerical extension.

Outbound CID is used to set the caller ID text for calls to a number of devices outside of the PBX. This option always requires a digital trunk (PRI/BRI, or VoIP) and setting the caller ID must be supported by the trunk service provider. This field is not required. If it is left blank, the default caller ID name for the trunk placing the call will be used to set the caller ID name text.

Ring Time is the number of seconds to ring the endpoint before dropping the caller to the user's voicemail. CompletePBX allows an endpoint to ring between 1 and 120 seconds. If no voicemail is configured for the user, this option will be ignored and the endpoint will ring until the caller hangs up. This field is a dropdown. If it is left at the default value, the extension will ring for the number of seconds configured in the **Ringtime Default** field on CompletePBX's **General Settings** page.

Call Waiting enables or disables the call waiting option on a user's extension by default. With call waiting enabled, if a user is called while their endpoint is already in use, the user will be notified of the new incoming call and will be given the option to answer it. Disabling call waiting will send incoming calls to voicemail if the user's endpoint is already in use. Users are able to enable or disable call waiting from their endpoint by dialing a feature code (**70* to activate, **71* to deactivate). This option is simply the initial setting for call waiting on the user's extension. If the call waiting state is toggled from the user's endpoint, then it will stay that way. It will not be overwritten by CompletePBX on reloads. This field is a drop-down menu.

Internal Auto Answer enables automatic answering of internal calls provided that the endpoint technology allows it. Calls originating from external callers (calls from the trunks) will not be answered automatically, and will be handled in the normal manner.

Call Screening allows users to hear who is calling before picking up a call. With this option enabled, callers will be prompted to say their name before ringing the user's endpoint. Upon answering the call, the user will hear the caller's name being played back to them and will be given the option to accept or reject the call. Rejected calls are sent to the user's voicemail if configured, or disconnected otherwise. Call screening can either be enabled with memory (caller ID text is remembered, and callers who have identified themselves in the past will never be prompted to do so again unless a fresh copy of CompletePBX is installed), or without memory (callers are prompted to identify themselves every time they call in). This field is a dropdown.

Enabling **Pinless Dialing** allows the extension to bypass any pin codes normally required on outbound calls.

The **Assigned DID/CID** group of options allow specific external telephone numbers to be assigned directly to a user's extension. Multiple DIDs can be assigned to a single extension.

DID Description is a description of the number associated with the extension.

Add Inbound DID is the actual number associated with the extension. The number must be in the format that the trunk provider sends it in (often 11 digits or full international dialing standard). **Add Inbound CID** allows only specific caller ID texts on the entered DID to be routed to the user's extension. Caller ID text can be a specific phone number as well as "Private", "Blocked", "Unknown", "Restricted", "Anonymous", or "Unavailable". None of the fields in the Assigned DID/CID group of options are required. If they are left blank, the extension will function as normal, and will only receive calls when an inbound route directs calls to it or another extension calls it directly.

The **Language Code** field will force all of the prompts specific to the user to be played in the specified language (as long as the language is installed and voice prompts for the specified language exist on the PBX). This field is not required. If left blank, prompts will be played back in the default language of the CompletePBX server.

The **Recording Options** group of fields allows a user to record incoming or outgoing calls. The user can either dial a feature code (*1*) to selectively enable recording for the current call, never record calls, or always record calls. Both the Recording Options fields are dropdowns.

The **Dictation Service** field allows the user to dial a feature code (*34*) from their endpoint and have whatever they say recorded. Dictations will be saved in the format selected in the field, and will be emailed to the address specified in the **Email Address** field. Completed dictations are only emailed to the user when they dial the appropriate feature code (*35*). Both the **Dictation Service** and **Dictation Format** fields are drop-down menus. The Email Address field is not required. If it is left blank, dictations cannot be emailed to the user once the user has finished recording.

Voicemail & Directory options are discussed in the **Voicemail setup** section of this chapter.

Common endpoint fields

Regardless of the endpoint technology selected, the Emergency CID will be present. The value entered here will override any other caller ID text set for the extension when the user places a call through a route configured for emergency calls in CompletePBX. This option is useful for allowing outbound calls to send accurate caller ID text to emergency service dispatch centers, where caller ID text would otherwise provide different information (call centers often provide corporate call back numbers in caller ID text that are assigned to buildings in different physical locations from the call centers themselves). This is especially important for VoIP trunks, as calls over VoIP can originate from anywhere in the world, and the trunks do not carry the information related to location that T1 or PSTN lines do. The Emergency CID field is not required. If it is left blank, the default caller ID settings on a trunk will be used to set the caller ID when an emergency call is placed.

Common SIP/IAX2 fields

SIP and IAX2 endpoints will have a **secret** field. This field has the password that the endpoint will use to authenticate with CompletePBX. This field is required.

Choose endpoint passwords carefully

Endpoint passwords can be the weakest link on any externally accessible PBX system, as malicious users will often attempt to locate extensions having weak passwords. Endpoints that authenticate with CompletePBX using passwords such as "1234" stand a good chance of being compromised, allowing an attacker to place calls through your PBX. Pick strong passwords carefully, and ensure that endpoint passwords are not distributed to anyone who does not need to know them

SIP endpoints will also have a dtmfmode field. The DTMF mode for a SIP device specifies how touch tones will be transmitted to the other side of the call. Possible values for this field are rfc2833, inband, info, and auto. The default value of rfc2833 is typically fine and should not be changed unless a trunk provider specifically requires it.

It is important to set the DTMF mode correctly based on the trunk. DTMF tones transmitted "inband" on a compressed stream (anything that is not using the ULAW or ALAW codec) will be garbled and will not be recognized on the receiving side. The value rfc2833 will send the tones as data within the RTP payload, and info will send the tones as data in the SIP info headers. Both data methods are more reliable at transmitting understandable tones than inband, but must be supported by the device generating the tones. The **auto** option will attempt to use rfc2833, but will fail over to inband if the receiving host does not indicate support for the data-based tones. The dtmfmode field is required.

Additional SIP fields

Once saved, clicking on a SIP device will show additional options that were not available during the initial setup:

- Device Options

This device uses sip technology.

Field	Value
secret	
dtmfmode	RFC 2833
canreinvite	No
context	from-internal
host	dynamic
trustrpid	Yes
sendrpid	No
type	friend
nat	No - RFC3581
port	5060
qualify	yes
qualifyfreq	60
transport	UDP Only
encryption	No
callgroup	
pickupgroup	
disallow	
allow	
accountcode	
vmexten	
deny	0.0.0.0/0.0.0.0
permit	0.0.0.0/0.0.0.0

The **canreinvite** option specifies whether a SIP endpoint can issue re-invites to other endpoints. By default, the media stream for a SIP call will pass through the CompletePBX server before reaching the other endpoint, even if both endpoints are on the same network. Re-invites allow two SIP endpoints to take CompletePBX out of the media path and speak directly with one another. This can improve latency in certain situations, but for most setups, this will cause dropped calls or unusable call quality. Valid values for canreinvite are **yes** and **no**. This field is not required. If it is left blank, the default value of no will be used.

The **context** option allows the device to be placed into a specific context within CompletePBX. A context should have been manually created in *the /etc/asterisk/extensions_custom.conf* file for this to work properly. Additionally, the CompletePBX **from-internal** context includes all necessary contexts to provide access to all extension features such as voicemail, outbound routes, and feature codes. Specifying a custom context may cause certain CompletePBX features to function incorrectly. This field is required.

The **host** option is used to define where calls for an extension should be sent. Valid values are "dynamic", or a specific IP address. Dynamic extensions will register with CompletePBX and CompletePBX will keep track of where they registered from, hence it knows where to send the call. Endpoints that have dynamic IP addresses should leave this set as "dynamic". Endpoints with static IP addresses can have their IP address entered in this field. These devices are not required to register with CompletePBX because when CompletePBX receives a call for them, it will forward the call to the configured IP. This field is not required. If it is left blank, the default value of "dynamic" will be used.

The **type** option is used to configure what types of calls will be expected from this extension. Valid options for the type field are **peer**, **user**, and **friend**. From the perspective of the CompletePBX

server, calls are sent to peer endpoints, they are received from user endpoints, and can be either sent to or received from "friend" endpoints. A normal office telephone handset will be configured as a "friend" endpoint. A call center that does not allow its agents to make outbound calls may configure all of its endpoints as peer endpoints. A hotel lobby may have a phone that is for outbound calls only (to other hotel guests), which would be configured as a user endpoint. This field is not required. If it is left blank, the default value of "friend" will be used.

The **nat** option specifies whether this endpoint is expected to be placed behind a device performing **Network Address Translation** (**NAT**). This option should only be disabled if a specific setup requires internal IP addresses to be sent to a remote CompletePBX server (for example, a phone that connects to CompletePBX over a VPN and the VPN passes through a NAT device). Valid nat values can only either **yes** or **no**. This field is not required. If it is left blank, the default value of "yes" will be used. Setting nat to "no", CompletePBX will use rport if the remote side says to use it; setting nat to "yes" forces rport to always be on.

The **port** option allows a custom port to be set for the endpoint, SIP control data is sent on this custom port. Note that this port is only used for SIP signaling and not for the actual media stream. The default port is **5060**, and most SIP servers will be listening on this port. This option is commonly changed where endpoints are using a connection on which port 5060 is blocked. Note that if the port value is changed, the CompletePBX server must also be listening for SIP signaling connections on the custom port specified. Valid values for **port** are numeric entries between 1 and 65,000. This field is not required. If it is left blank, the default value of 5060 will be used.

The **qualify** option will keep track of an endpoint causing CompletePBX to ensure that the endpoint is still online. If qualification for an endpoint is enabled, CompletePBX will also track its response time relative to the PBX. This option can be useful for tracking latency between the PBX and its endpoints. Running the command *sip show peers* from the **CompletePBX command line interface** (**CompletePBX CLI**) will show a list of all SIP endpoints along with the number of milliseconds it has been since the endpoint responded to a qualify request. Valid values for the **qualify** option are **no**, **yes**, or a numerical time value in milliseconds. Setting this option as yes will qualify an endpoint every 200 milliseconds. This field is not required. If it is left blank, the default value of no will be used.

The **callgroup** and **pickupgroup** options allow users to pick up calls that are not directed to them by dialing a feature code (*8*). Both values are numeric and range between 0 and 63.

Extensions can be placed into multiple groups by separating numerical values by commas. Calls directed to any phone in a particular call group can be answered by any user who is a member of the corresponding pickup group (for example, a user in *pickupgroup 1* will be able to pick up any call directed to any phone in *callgroup 1*). This can be useful for small office or home setups, where it is easier to simply pick up a call from another phone rather than forward that call to another extension.

Note that a user can be part of a *pickupgroup* without being a member of the associated *callgroup* (for example, a senior staff member may be able to pick up any call directed to anyone in his department, but his department should not be able to pick up calls directed to the senior staff member).

Neither the *callgroup* field nor the *pickupgroup* field is required. If they are left blank, the endpoint will simply not be a member of a *callgroup* or *pickupgroup*, and will not be able to use the call pickup functionality.

The **disallow** option should contain a list of codecs, which the endpoint should not use to communicate with CompletePBX. Multiple codecs can be specified separated by spaces. Valid **codec** values are "all", "g723", "gsm", "ulaw", "alaw", "g726", "adpcm", "slin", "lpc10", "g729", "speex", and "libc".

It is common practice to first disallow all codecs in this line, and then, specifically allow a select few using the **allow** option. This field is not required. If it is left blank, no codecs will be disallowed, and CompletePBX may attempt any codec available to it when establishing a call to the endpoint.

The **allow** option is the opposite of the **disallow** option, and is used to specify which codecs should be used for communication with the endpoint. Codecs are used with preference to the order in which they are specified.

The allow option has the same valid values as the disallow option. This field is not required. If it is left blank, no codec will be preferred by the endpoint, and CompletePBX may use any codec available to it when establishing a call to the endpoint. CompletePBX will give preference to codecs that do not require transcoding.

Transcoding

The codecs that each side of a call supports play the biggest role when choosing a default codec for an endpoint or a trunk. The rule of thumb for codec selection is to try and configure an endpoint to use the same codec as the trunk it will be using for external calls.

When an endpoint uses a different codec than the trunk carrying the endpoint's call, CompletePBX must convert the call's audio stream on the fly. This process is known as **transcoding**. Transcoding uses processor resources and can bog down an CompletePBX server. A system performing transcoding for every call is able to handle a significantly fewer number of concurrent calls than a system that does not.

Transcoding also incurs a latency penalty for the call's audio (that is, the time between when one party says something and the other party hears it is increased). Depending on the original quality of the call before transcoding, this could be enough to make an audible difference in the amount of lag time that each party on the call hears. In extreme cases, it may not be possible to carry on a conversation.

DAHDI trunks (PSTN lines, PRI lines, and T1 lines) will use the G.711 codec. North American and Japanese style trunks should use the G.711 ULAW codec. Trunks in other areas of the world should use G.711 ALAW. VoIP trunks will typically use either G.711 or G.729 and they may support both.

Transcoding costs can be determined by running the command *show translation* in the CompletePBX CLI. CompletePBX will output a matrix of all the codecs it can use and the latency penalty that a call will incur when being transcoded between each available codec.

 Note that certain codecs may need to be purchased (for example, Digium sells the G.729 codec on a per-channel basis). Some codecs may be free, but will need to be installed before they can be used, such as Speex.

The other factor to consider during codec selection is the available bandwidth where the CompletePBX server resides.

The G.711 codecs (*ulaw* and *alaw*) use about 64 kbps for each side of the call (that is, 64 kbps for sending what is said, and 64 kbps for receiving what the other party says, for a total of 128 kbps). The G.729 codec knocks the required bandwidth down to 8 kbps for each side of the call. The iLBC can use about 15 kbps per side of the call, and Speex is a variable bit-rate codec, using between 4 kbps and 48 kbps for each side of the call. Using 128 kbps of bandwidth to connect a phone over a local network to CompletePBX is certainly no problem as there will be plenty of bandwidth to spare. Although, when VoIP trunks or phones outside of the local network come into play, bandwidth is significantly limited. Many broadband providers have a limited upload speed of 768 kbps or less (this is especially common for DSL connections). Factoring in normal connection overhead, this leaves us with space for about 10 active calls if we use G.711, but dozens of calls if we use G.729.

The rule of thumb when selecting a codec is to try to use the same codec for endpoints and trunks as long as the bandwidth required for doing so does not exceed available resources. If matching the codecs is not possible, the next best method is simply to balance the requirement for lowering bandwidth against the requirement for reducing transcoding as much as possible.

Keep in mind that not all of the endpoints must use the same codec. For example, if all calls go out of G.711 trunks, it is always feasible to have all of the endpoints on the network that is local to CompletePBX use G.711, while endpoints outside of the network (that connect to CompletePBX over a broadband connection) use G.729. This method will only use transcoding for those endpoints that are outside of the office. External calls will use 16kbps of bandwidth to connect with CompletePBX.

The **dial** option is the dial string CompletePBX will use to call the endpoint. Any valid CompletePBX dial string is admissible for this option. This field is required.

The **accountcode** option is typically used for billing purposes. A string value of length up to 20 characters may be used for the account code. The value entered here will show up in the CompletePBX CDR for every call placed by this extension. This field is not required. If it is left blank, the CDR will log the extension that placed a call, but will not log any account code associated with the call.

The **mailbox** option is used to specify the voicemail box that the device will be checking. The SIP protocol will return the number of waiting messages in the specified mailbox when this device queries CompletePBX for waiting messages. Most endpoints will provide some kind of indication when their mailboxes have messages in them (such as a flashing light or a stuttering dial tone). Valid values are any mailboxes that have been created using CompletePBX or entered manually into the */etc/asterisk/voicemail.conf* file. This field is not required. If left blank, the endpoint will not check to see if any voicemail messages are waiting.

The **deny** option specifies network addresses from which CompletePBX should not accept traffic for this peer. This option should be in the format of an IP address and subnet, such as 192.168.1.0/255.255.255.0 (to disallow traffic for this extension from the IP range of 192.168.1.1 to 192.168.1.254). It is possible to enter a value of 0.0.0.0/0.0.0.0 to deny all of the networks by default, and, to enter specific networks from which traffic can be accepted in the **permit** option.

This option is commonly used to restrict endpoint usage to a particular network, so that if the endpoint is stolen or otherwise removed from the network, it cannot be used to place calls and will be essentially useless. This field is not required. If it is left blank, CompletePBX will not block traffic for this peer from any IP address.

The **permit** option is the opposite of the **deny** option. Specific IP addresses or networks can be added in this option to allow traffic for this extension from the entered IP/network. This field is not required. If it is left blank, traffic will be allowed from all IP addresses.

Additional IAX2 fields

Once saved, clicking on an IAX2 device will show additional options that were not available during initial setup:

- **Device Options**

This device uses iax2 technology.

secret ❓	
transfer ❓	Yes ▾
context ❓	from-internal
host ❓	dynamic
type ❓	friend ▾
port ❓	4569
qualify ❓	yes
disallow ❓	
allow ❓	
accountcode ❓	
deny ❓	0.0.0.0/0.0.0.0
permit ❓	0.0.0.0/0.0.0.0
requirecalltoken ❓	Yes ▾

The additional options shown for IAX2 endpoints are very similar to the additional options shown for SIP endpoints. The only additional option available to IAX2 endpoints is the **notransfer** option.

The IAX2 **notransfer** option is the equivalent of the SIP **canreinvite** option (although the values are reversed). Setting the notransfer option to **yes** prohibits the media path from being transferred away from CompletePBX, keeping CompletePBX in the media path at all times. If notransfer is set to **no**, IAX2 endpoints will be able to communicate directly with one another, removing CompletePBX from the media path entirely.

All other additional IAX2 options are equivalent in functionality to their SIP option counterparts.

Common DAHDI fields

DAHDI endpoints will have a **channel** field. It refers to the DAHDI channel that the CompletePBX DAHDI driver will use to communicate with the DAHDI endpoint. The channel is specific to each FXS port of the Astribank. Running the *lsdahdi* command from the Linux command line will output a list of all (DAHDI) devices on the system and a list of the channels for each device. The value that is entered into the channel field must be present in the list that the *lsdahdi* command returns.

Additional DAHDI fields

Once saved, clicking on a DAHDI device will show additional options seen in the following screenshot, which were not available during initial setup:

- Device Options

This device uses zap technology. (Via DAHDi compatibility mode)

channel ❓	
context ❓	from-internal
immediate ❓	No
signalling ❓	fxo_ks
echocancel ❓	yes
echocancelwhenbridged ❓	No
echotraining ❓	800
busydetect ❓	No
busycount ❓	7
callprogress ❓	No
accountcode ❓	
callgroup ❓	
pickupgroup ❓	

Several additional DAHDI configuration options are equivalent to their SIP or IAX2 configuration option counterparts.

The **immediate** option will cause CompletePBX to start executing its dialplan for this extension as soon as the handset goes "off hook", without the caller ever having to dial anything. This option is commonly used for emergency phones, such as phones in an elevator, so that the emergency personnel are contacted as soon as someone picks up the phone.

This option is useful only if a custom context has been built to support it, and that context is specified in the context option. Custom contexts must be manually written into the */etc/asterisk/extensions_custom.conf* file.

The **signalling** option is used to specify the type of signaling used by the channel. Signaling is specified using a string, and must correspond to the DAHDI settings for the channel. For example, if an FXS channel is defined in the DAHDI configuration file *(/etc/dahdi/system.conf)* as 'fxoks=1' then the 'signalling' parameter must be defined as fxo_ks

The **echocancel** option enables *software* echo cancellation for the endpoint. This is not to be confused with hardware echo cancellation that some devices have as an option. Hardware echo cancellation must be configured on the device itself. The *echocancel* option can be set to one of these values—**no**, **yes**, **16**, **32**, **64**, **128**, or **256**.

Setting the option to "no" disables echo cancellation. Setting the option to "yes" enables echo cancellation with a value of 128. The numerical value specified in the *echocancel* option is the number of taps that CompletePBX will use to sample the media stream to determine the best way to cancel an echo on the call. One tap is one sample from the data stream.

Note that if this option is set to anything other than one of the valid values specified, CompletePBX will default the option to yes (128 taps) without warning.

The **echocancelwhenbridged** option allows echo cancellation to be enabled or disabled for calls that are bridged between two DAHDI channels. As most of the time the calls between two DAHDI endpoints will not have any echo, this option is not required. It is best practice to begin adjusting for echo only after the echo is experienced, rather than trying to adjust the echo cancellation properties based on echoes that might occur. Valid values for the *echocancelwhenbridged* field are **yes** or **no**.

The **echotraining** option allows a DAHDI channel to train itself for the proper echo cancellation

requirements for the call, before the call begins. Valid values for echo training are "yes", or they may be numbers between 10 and 4,000. When enabled, CompletePBX will mute the channel at the beginning of any call made to or from this device for the number of milliseconds entered as a value. During this time, a pulse will be sent to the other side of the call, and the information gathered from whatever echo of this pulse is returned, which will be used to fine tune the echo cancellers on the channel. This can often result in echo-free calls from the very beginning of the conversation, instead of forcing the echo trainer to begin training after the call has already started. As with the *echowhenbridged* field, it is a good idea to start adjusting the *echotraining* field only after the echo is experienced. Note that *echotraining* is not required for OSLEC software echo canceller or for the Hardware Echo Canceller

The **busydetect** option is used to detect if a called DAHDI channel returns a busy tone. As DAHDI endpoints are typically analog devices and can therefore not return their status to CompletePBX, enabling the *busydetect* option allows CompletePBX to know that the called endpoint is in use and enables calls to be routed appropriately based on that information. Valid values for this option are "yes" and "no".

The **busycount** option works in conjunction with the *busydetect* option. If the *busydetect* option is set to "yes", the value set in the *busycount* option is used as the number of busy tones for which CompletePBX will wait before it determines that a channel is busy.

Valid values are whole numbers greater than 1. The higher this value is, the longer a busy channel will take to disconnect. However, lower values may cause repetitive sounds in a conversation to be treated as busy/disconnect tones, and that call will be terminated. The default value of **7** is recommended.

When enabled, the **callprogress** option will force CompletePBX to try to recognize various analog tones, which are types of **Special Information Tone** (**SIT**), in order to determine how a call is progressing. CompletePBX will attempt to recognize ringing tones, busy tones, disconnection tones, or sounds indicating that a call has been answered. Although this option typically provides few benefits, it will usually result in quicker termination of disconnected channels.

Common custom fields

Custom endpoints will have a **dial** field. This field is for specifying a custom *Dial* command that CompletePBX will use to communicate with the endpoint (using a protocol that is not SIP, IAX2, or DAHDI).

Common custom dial strings are *H323/${EXTEN}@(IPADDRESS)* for an H.323 device, and *MGCP/${EXTEN}@(IPADDRESS)* for an MGCP device. The *IPADDRESS* token should be replaced with the IP address of the device being added.

Setting up extensions in Extensions mode

In Extensions mode, each user who is defined in CompletePBX has a single endpoint associated with them. Extensions mode is a good choice for simple setups where each person configured on the PBX is always expected to be using the same phone. If your CompletePBX system is configured in the *DeviceAndUser* mode, you can skip this section.
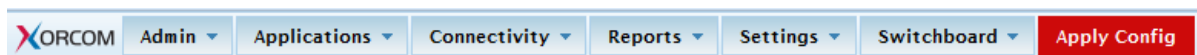
Configuring CompletePBX for Extensions mode

Switching CompletePBX between operational modes is done from the Administration GUI by entering the **Advanced Settings** dialog.

If CompletePBX was configured to use Extensions mode during initial installation, no additional configuration is required to begin creating extensions. These instructions only need to be followed if CompletePBX was set up to use the DeviceAndUser mode during the initial installation

Log in to the Administration interface, navigate to the **Settings** menu, and select the **Advanced Settings** dialog. Navigate down to the **System Setup** section, and modify the **User & Devices Mode** parameter to the default value of **extensions**, and click on the arrow within the green circle to the right. The configuration changes must be applied to CompletePBX by clicking on the red-colored **Apply Config** bar at the top of the screen.

CompletePBX will now be running in **extensions** mode.

Configuring extensions

Now that CompletePBX is running in Extensions mode, we can begin creating extensions. Each endpoint used will require a unique extension. To configure extensions, navigate to the CompletePBX Administration interface and click on the **Connectivity** menu on the navigation bar, and select the **Extensions** dialog. The **Add an Extension** dialog will be displayed:

From the **Device** drop-down menu, select the type of endpoint that is being configured (SIP, IAX2, DAHDI, or other). Click on the **Submit** button. Additional form fields will be displayed like those in the following screenshot:

Add SIP Extension

- **Add Extension**

User Extension❓ [   ]
Display Name❓ [   ]
CID Num Alias❓ [   ]
SIP Alias❓ [   ]
- **Extension Options**

Outbound CID❓ [   ]
Ring Time❓ [Default ▾]
Call Forward Ring Time❓ [Default ▾]
Outbound Concurrency Limit❓ [No Limit ▾]
Call Waiting❓ [Enable ▾]
Internal Auto Answer❓ [Disable ▾]
Call Screening❓ [Disable   ▾]
Pinless Dialing❓ [Disable ▾]
Emergency CID❓ [   ]
Queue State Detection❓ [Use State ▾]
- **Assigned DID/CID**

DID Description❓ [   ]
Add Inbound DID❓ [   ]
Add Inbound CID❓ [   ]

Fill out the form fields with the appropriate information. Information on most fields can be found in the *Common fields* section covered earlier in this chapter. Information regarding the **Voicemail & Directory** group of fields can be found in the **Voicemail setup** section of this chapter. Information regarding the **VmX Locater** group of fields can be found in the *Follow Me and the VmX Locater* section of Chapter 6, **Call Routing**.

Note that only **User Extension**, **Display Name,** and if present, **secret** are required fields. All other fields can be left at their default values and the extension will still be created successfully. In addition, all fields except for the User Extension field can be altered after the extension has been saved.

Note that a list of all of the user extensions that already exist is present on the right side of the screen. This list can be used to easily determine the next available unique extension number.

Once all fields have been filled out, click on the **Submit** button at the bottom of the page to save the extension. The extension that was just saved will be in the extension list on the right side of the screen. Before the extension can be used, the configuration changes must be applied to CompletePBX. Click on the red-colored **Apply Config** bar at the top of the screen:

XORCOM Admin ▾ Applications ▾ Connectivity ▾ Reports ▾ Settings ▾ Switchboard ▾ **Apply Config**
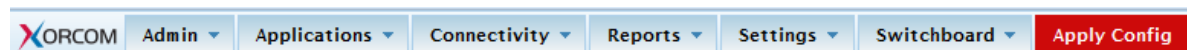
Once a device is saved, clicking on the device name in the active extensions list will provide additional options that were not accessible during the extensions initial setup for SIP, IAX2, and DAHDI devices. (Note that custom devices do not have any additional options.) CompletePBX assigns the most common values to these additional options. Typical setups will not require any changes.

Setting up a new extension in DeviceAndUser mode

In the DeviceAndUser mode, each user defined in CompletePBX can have multiple endpoints associated with them. DeviceAndUser mode is a good choice for more complex setups, where each person configured on the PBX is expected to work from multiple, different phones or multiple locations. If your CompletePBX system is configured in the Extensions mode, this section can be skipped.
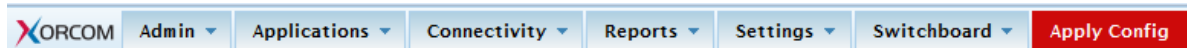
Configuring CompletePBX for DeviceAndUser mode

Switching CompletePBX between operational modes is done from the Administration interface in the Advanced Settings dialog.

If CompletePBX was configured to use **DeviceAndUser** mode during initial installation, no additional configuration is required to begin creating devices and users. These instructions only need to be followed if CompletePBX was set up to use the Extensions mode during initial installation

Log in to the Administration interface, navigate to the **Settings** menu, and select the **Advanced Settings** dialog. Navigate down to the **System Setup** section, and modify the **User & Devices Mode** parameter from the default value of **extensions** to **deviceanduser**, and click on the arrow within the green circle to the right. The configuration changes must be applied to CompletePBX by clicking on the red-colored **Apply Config** bar at the top of the screen.

CompletePBX will now be running in **deviceanduser** mode.

Configuring users

Now that CompletePBX is running in DeviceAndUser mode, we can begin creating users on the system. Each user will require a unique numerical extension. To configure users, navigate to the **Connectivity** menu and select the Users dialog. Click on the **Add Users**: the **Add User/Extension** screen will be displayed:

Add User/Extension

- **Add User**

User Extension❓
User Password❓
Display Name❓
CID Num Alias❓
SIP Alias❓

- **Extension Options**

Outbound CID❓
Ring Time❓   [Default ▾]
Call Forward Ring Time❓   [Default ▾]
Outbound Concurrency Limit❓ [2   ▾]
Call Waiting❓   [Enable ▾]
Internal Auto Answer❓   [Disable ▾]
Call Screening❓   [Disable   ▾]
Pinless Dialing❓   [Disable ▾]
Queue State Detection❓   [Use State ▾]

- **Assigned DID/CID**

DID Description❓
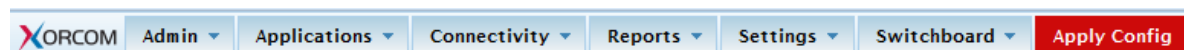Add Inbound DID❓
Add Inbound CID❓

Fill out the form fields with the appropriate information. Besides the common configuration fields, there will be a **User Password** field. This field is a numerical password that this user can use to log into any endpoint attached to CompletePBX.

Information on most fields can be found in the *Common fields* section earlier in this chapter. Information regarding the **Voicemail & Directory** group of fields can be found in the **Voicemail setup** section later in this chapter. Information regarding the **VmX Locater** group of fields can be found found in the *Follow Me and the VmX Locater* section of Chapter 6, **Call Routing**.

Note that only **User Extension**, **User Password**, and **Display Name** are required fields. All of the other fields can be left at default values and the extension will still be created successfully. In addition, all fields except for the User Extension field can be altered after the extension has been saved.

Once all of the fields have been filled out, click on the **Submit** button at the bottom of the page to save the extension. The user that was just saved will now be shown in the active user list on the right side of the screen.

Before the extension can be used, the configuration changes must be applied to CompletePBX. Click on the red-colored **Apply Config** bar at the top of the screen as shown in the following screenshot:

XORCOM | Admin ▾ | Applications ▾ | Connectivity ▾ | Reports ▾ | Settings ▾ | Switchboard ▾ | Apply Config

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Once a user is saved, clicking on the user name in the active users list will load the user settings to be edited. Certain modules such as **Follow Me** will add additional options that were not accessible during the initial setup of the user.

Follow Me Settings are discussed in Chapter 6. Once a user is configured, we can proceed with configuring devices.

Configuring devices

With a user configured, it is now possible to configure devices. To configure devices, navigate to the CompletePBX Administration interface, click on the **Connectivity** menu on the navigation bar and select **Devices** dialog:

Add Device

Please select your Device below then click Submit

- Device

Device Generic SIP Device ▼

Submit

From the Devices drop-down menu, select the type of endpoint that is being configured (SIP, IAX2, DAHDI, or other). Click on the **Submit** button.

Add Device

Please select your Device below then click Submit

- Device

Device Generic SIP Device ▼

| Generic SIP Device |
| Generic IAX2 Device |
| Generic ZAP Device |
| Generic DAHDi Device |
| Other (Custom) Device |

Submit

Fill in the appropriate fields in the dialog. The fields in the dialog will vary depending on the type of endpoint that will be in use. All endpoint types will have fields for **Device ID**, **Description**, **Emergency CID**, **Device Type**, and **Default User**.

The **Device ID** field is a unique numerical value that is used by CompletePBX to dial the device. SIP and IAX2 devices will require this value when the devices themselves are being configured.

Description is a text description of the purpose of the device. It is common to use this field to describe the location of the device or its MAC address.

The **Device Type** field can be either "fixed" or "adhoc". A **fixed** device will always have the same user logged into it (it will not be possible for the user to log out of the device). An **adhoc** device allows users to log in and out of itself at will. When a user is logged into an ad-hoc endpoint, any calls to their extension will ring that endpoint. Users can be simultaneously logged into as many endpoints as they wish.

The user selected in the Default User dropdown is the user that will be logged into the device by default. Fixed devices require this value to be set. Adhoc devices can have this value set to "none" so

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that no user is logged into the device by default.

Once all of the required fields are filled in, click on the **Submit** button at the bottom of the page to save the device configuration.

Once a device is saved, clicking on the device name in the active devices list will load additional device configuration options that were not available during the initial setup for SIP, IAX2, and DAHDI devices.

CompletePBX assigns the most common values to these additional options. Typical setups will not require any changes.

Voicemail setup

The voicemail capabilities built into CompletePBX compare well against enterprise-level voicemail solutions. CompletePBX makes voicemail boxes a breeze to set up.

Voicemail delivery options

Voicemail options are located either on the **Extensions** page (if CompletePBX is operating in Extensions mode), or on the **Users** page (if CompletePBX is operating in DeviceAndUser mode). From the CompletePBX **Connectivity** menu, either click on **Extensions** or on **Users**, click on the name of the user for whom you want to configure voicemail settings in the active users list, and scroll to the bottom of the page to view voicemail settings:

The **Status** field can be set to either "Enabled" or "Disabled". If voicemail is disabled, messages cannot be left for the user and calls directed to them will continue to ring until the caller hangs up. All other voicemail options are disabled, unless this field is set to "Enabled".

The **Voicemail Password** field should be set to a numerical password that the user will enter to access their message center. This setting can also be changed by the user from their VoIP endpoint.

If a valid email address is entered in the **Email Address** field, any time a voicemail is left for the user an email message will be sent to the address entered here. By default, the message will simply notify the user that a new message has been left for them. Only one address can be entered into this field.

The **Pager Email Address** field can be used to have CompletePBX send a short message to a mobile device whenever a new message is left in the user's voicemail. The message will be under 140 characters, and is typically used in conjunction with SMS or pager mobile services. Note that this field is *not* for a mobile phone number, but rather an email address that will reach a mobile device. Most mobile carriers provide an email-to-SMS gateway service that can be used to send emails to

mobile telephones. Only one address can be entered into this field.

If the **Email Attachment** field is set to **yes**, the email message sent to the address in the Email Address field will have a copy of the voicemail attached to it. This message can then be played through the speakers of the computer on which it was received. Messages are not attached to email notifications sent to the address in the Pager Email Address field.

If the **Play CID** field is set to **yes**, the system will play back the caller ID number of the person who left the message prior to the message being played.

If the **Play Envelope** field is set to **yes**, the user will hear the date and time that the message was left prior to hearing the message being played.

If **Delete Voicemail** field is set to **yes**, new messages will be deleted from the user's messaging center as soon as the notification email is delivered to the address specified in the Email Address field. Be very careful with this option, because CompletePBX will allow you to delete the message without having a copy of it attached to the email notification. This could mean that a message is left and a notification email is sent to the user, but the actual voicemail that was left will not be accessible at all.

The **VM Options** field can contain any option that would normally be specified in the */etc/asterisk/voicemail.conf* file. These options can enable other options such as forcing a user to record their name and greeting before listening to messages, increasing the volume on voicemail messages as they are recorded, or enabling a warning for the user if they are using a temporary greeting as their outgoing message. Details on these options can be found in Appendix C, *Voicemail.conf Options*, or in the CompletePBX documentation for the *voicemail.conf* file. Multiple options can be entered into this field by separating the options with commas.

The **Voicemail Context** field is a string value that can be set to any valid context configured within the */etc/asterisk/voicemail.conf* file. Voicemail contexts can be used to apply one collection of settings to a particular group of mailboxes. If you have not manually edited the */etc/asterisk/voicemail.conf* file to create your own voicemail contexts, leave it as "default".

When configuring voicemail settings is complete, click on the **Submit** button at the bottom of the page followed by the red-colored **Apply Config** bar at the top of the page.
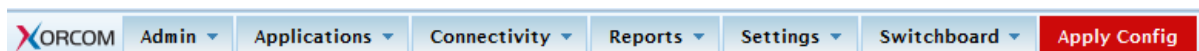
Managing voicemail messages

There are two primary ways for a user to manage their voicemail messages:

- The CompletePBX messaging center
- The CompletePBX **My Extension** Web site built into CompletePBX

Users can dial *97* to access their own mailbox when dialing from their own extension (Extensions mode), or from the device where their extension is logged in. They will be prompted for a password only. If the user is calling from someone else's phone on the PBX, they can dial *98*. They will be prompted for their extension and their password. Once a valid password has been entered, the user will hear voice prompts that direct which buttons to press on their touch tone telephone or softphone in order to listen to and manage their voicemail messages.

Once logged into the messaging center, the following options are valid:

1: Plays the oldest unheard message (if any), or plays the oldest message that is still stored. When listening to a message, the following options are valid:

> **3**: Advanced options, allows user to send a voice reply, call the caller who left the message,

and hear date and time information about the voicemail

4: Play the previous message

5: Repeat the current message from the beginning

6: Play the next message

7: Delete the current message

8: Forward the message to another user on the PBX

9: Save the message to another folder

*: Rewind the message five seconds

#: Fast forward the message five seconds

2: Change folders

3: Advanced mailbox options

0: Mailbox options. Once in mailbox options, the following options are valid:

1: Record unavailable message

2: Record busy message

3: Record name

4: Change voicemail password

*: Return to the main menu

If the user would like to have a visual overview of their messages, voicemail messages can be viewed and administered from **My Extension** Web page that is accessible from the CompletePBX Launch Pad. To access **My Extension**, visit the IP address of your CompletePBX server in the recordings directory (for example, http://192.168.0.64/recordings*)*. The **My Extension** login screen will be shown:

Users can log into **My Extension** using their extension number for the **Login** field and their voicemail password for the **Password** field. Once logged in, users will see a visual representation of their voicemail messages:

From **My Extension**, users will be able to play messages by clicking on the **play** link in the **Playback** column.

A message can be deleted by clicking on the checkbox next to the message date and clicking on the **delete** button.

A message can be moved into another folder by clicking on the checkbox next to the message date, selecting a new folder in the **Folder** dropdown, and then clicking on the **move_to** button.

A message can be forwarded to another user by clicking on the checkbox next to the message date, selecting a user in the empty second-row dropdown, and then clicking on the **forward_to** button.

Summary

In this chapter, we learned how to create users and endpoints in either Extensions or DeviceAndUser mode. We also discussed the distinctions between devices and users when CompletePBX is run in the DeviceAndUser mode.

After this chapter, we should have a CompletePBX system configured with user accounts and device options that are appropriate for our network. We also learned how to create a voicemail box for each user, and how to manage voicemail boxes using both the CompletePBX messaging center and **My Extension**. Each user account on our PBX should now have a voicemail box and be listed in the directory.

In the next chapter, we will learn how to connect our PBX with the outside world or other PBX servers using trunks.

3. Trunks

A trunk in the simplest of terms is a pathway into or out of a telephone system. A trunk connects CompletePBX to outside resources, such as PSTN telephone lines, or additional PBX systems to perform inter-system transfers. Trunks can be physical, such as a PRI or PSTN line, or they can be virtual by routing calls to another endpoint using **Internet Protocol** (**IP**) links. In this chapter, we will discuss the following:

- Various types of trunks allowed by CompletePBX
- Methods for setting up each type of trunk
- Checking the status of any configured trunk

Trunk types

CompletePBX allows the creation of six different types of trunks as follows:

- DAHDI
- IAX2
- SIP
- ENUM
- DUNDi
- Custom

DAHDI, IAX2, and SIP trunks utilize the technologies of their namesake. These trunks have the same highlights and pitfalls that extensions and devices using the same technology do. DAHDI trunks require physical hardware cards for incoming lines to plug into.

SIP trunks are the most widely adopted and compatible, but have difficulties traversing firewalls. IAX2 trunks are able to traverse most firewalls easily, but are limited to adoption mainly on CompletePBX-based systems.

In terms of VoIP, **ENUM** (**E.164 NUmber Mapping**) is a method for unifying E.164 (the international telecommunication numbering plan) with VoIP routing. The ENUM system can be considered very similar to the way that the Internet DNS system works. In the DNS system, when a domain name is looked up an IP address is returned. The IP address allows a PC to traverse the Internet and find the server that belongs to that IP address. The ENUM system provides VoIP routes back when queried for a phone number. The route that is returned is usually a SIP or IAX2 route.

An ENUM trunk allows CompletePBX to send the dialed phone number to the public *e164.org* ENUM server. If the called party has listed their phone number in the *e164.org* directory, a VoIP route will be returned and the call will be connected using that route. A VoIP route contains the VoIP protocol, the server name or IP address, the port, and the extension to use in order to contact the dialed phone number.

For example, a SIP route for dialing the number 555-555-1234 might appear as SIP:1234@pbx.example.com:5060. This is advantageous in several ways. It is important to note that indirect routes to another telephony system are often costly. Calling a PSTN telephone number typically requires that call to route through a third-party provider's phone lines and switching equipment (a service they will happily charge for). If a number is listed in the ENUM directory, the returned route will bridge the call directly to the called party (or their provider), bypassing the cost of routing through a third party.

ENUM also benefits the called party, allowing them to redirect inbound calls to wherever they would

like. Service disruptions that would otherwise render a particular phone number useless can be bypassed by directing the phone number to a different VoIP route in the ENUM system.

More information on ENUM can be found at the following Web sites:

- The ENUM home page: http://www.enum.org/
- The e164.org home page: http://www.e164.org/
- The Internet Engineering Task Force ENUM charter:
- http://www.ietf.org/html.charters/enum-charter.html

DUNDi (**Distributed Universal Number Discovery**) is a routing protocol technology similar to ENUM. In order to query another CompletePBX or CompletePBX system using DUNDi, that system must be "peered" with your own CompletePBX system. Peering requires generating and exchanging key files with the other peer.

DUNDi is a decentralized way of accomplishing ENUM-style lookups. By peering with one system you are effectively peering with any other system that your peer is connected to. If system A peers with system B, and system B peers with system C, then system C will be able to see the routes provided by system A. In peer-to-peer fashion, system B will simply pass the request along to system A, even though system C has no direct connection to system A.

DUNDi is not limited to E.164 numbering schemes like ENUM and it allows a PBX to advertise individual extensions, or route patterns, instead of whole phone numbers. Therefore, it is a good candidate for distributed office setups, where a central PBX can be peered with several satellite PBX systems. The extensions on each system will be able to call one another directly without having to statically set up routes on each individual PBX.

More information on DUNDi can be found at the following Web sites:

- DUNDi home page: http://www.dundi.com/
- Example DUNDi SIP configuration: http://www.voip-info.org/wiki/view/DUNDi+Enterprise+Configuration+SIP
- Example DUNDi IAX2 configuration: http://dundiglobal.org/documentation/tying-two-pbxs-together-using-dundi-and-iax2

Custom trunks work in the same fashion as custom extensions do. Any valid CompletePBX *Dial* command can be used as a custom trunk by CompletePBX. Custom trunks typically use additional VoIP protocols such as H.323 and MGCP.

Setting up a new trunk

Setting up a trunk in CompletePBX is very similar to setting up an extension. All of the trunks share eight common setup fields, followed by fields that are specific to the technology that trunk will be using.

Trunk configurations for common VoIP providers are listed in Chapter 16, **Trunk Configurations.**

In order to begin setting up a trunk, navigate to the **Connectivity** menu and select the **Trunks** dialog:
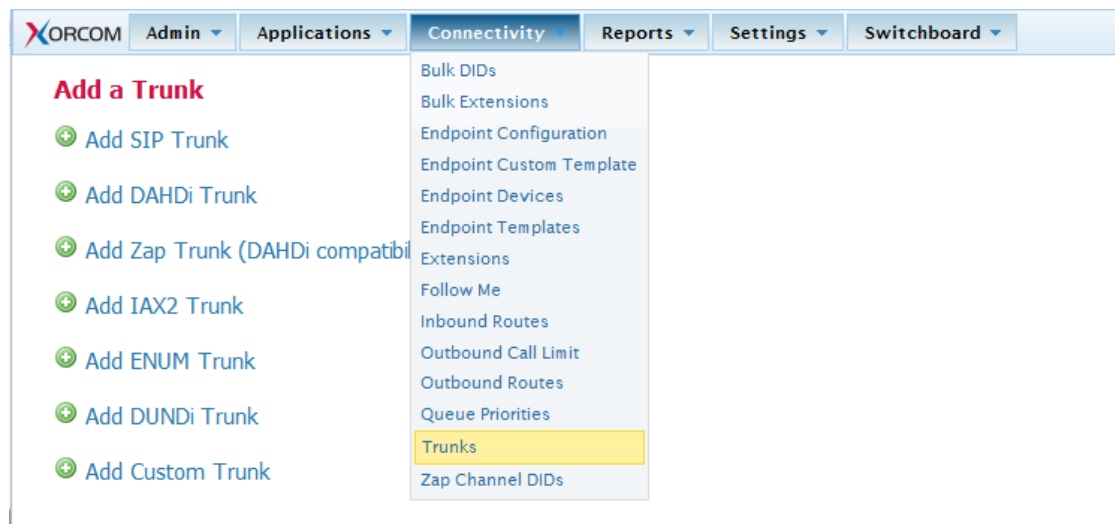
From the **Add a Trunk** screen, click on the name of the technology that the trunk will be using (for example, if a SIP trunk will be used, click on **Add SIP Trunk** as shown in the following screenshot:

Common trunk setup fields

No matter which technology a trunk will be used, the same common fields will appear first on the configuration page as shown in the following screenshot:

General Settings

Trunk Name ❓ :

Outbound CallerID ❓ :

CID Options ❓ :   Allow Any CID ▼

Maximum Channels ❓ :

Disable Trunk ❓ :   ☐ Disable

Monitor Trunk Failures ❓ :   ☐ Enable

Dialed Number Manipulation Rules ❓

(prepend) + prefix | match pattern   ➕ 🗑

+ Add More Dial Pattern Fields   Clear all Fields

Dial Rules Wizards ❓ :   (pick one) ▼

Outbound Dial Prefix ❓ :

Outbound Caller ID sets the caller ID name and number that will be displayed to the called party. Caller ID should be in the format of "Name" <##########>, where "Name" is the name that will be displayed, and <##########> will be the telephone number displayed. Be sure to include the quotes and angle brackets to send both name and number information. In order to send just a telephone number without a name, simply put the telephone number in the **Outbound Caller ID** field without quotes or angle brackets. Note that setting outbound caller ID only works on digital lines (T1/E1/J1/PRI/BRI/SIP/IAX2), not POTS lines. The ability to set outbound caller ID must also be supported by your provider.

Warning

CompletePBX will ignore the Outbound Caller ID field if it is not in the proper format. Be sure that a caller ID with a name and number contains a name in quotes and a telephone number in angle brackets. A caller ID with just a telephone number must only contain the number and nothing else. CompletePBX will not set the caller ID on outbound calls that do not match one of these two formats

Maximum Channels is the maximum number of *outbound* calls that this trunk can support. Note that CompletePBX does not count inbound calls against the channel limit. Whole numbers greater than one are the only valid values for this field. If no limit should be imposed, leave this field blank.

The **Disable Trunk** checkbox allows a trunk to be temporarily disabled in any outbound route where the trunk is in use. If a trunk is experiencing difficulties, it can be disabled here in order to have outbound calls skip the trunk entirely in their routing sequence. Under normal circumstances, a downed trunk will simply refuse a call, but in scenarios in which a trunk accepts a call but fails to connect it properly, this option is especially useful. Without disabling the trunk in this scenario, all of the calls will be dropped. Disabling the trunk allows outbound calls to bypass the problematic trunk while it is repaired, instead of having to remove the trunk entirely and recreate it when the problem is resolved.

The **Monitor Trunk Failures** field allows an AGI script to be called when a trunk fails. CompletePBX determines that a trunk has failed when calls cannot be completed and the returned status from CompletePBX is *not BUSY*, *NOANSWER*, or *CANCEL*. Scripts specified in this field can perform any action that can be scripted, including reloading CompletePBX configuration to force trunk re-registration, or notifying a system administrator of the failure through email. Failure scripts must be

located in the *var/lib/asterisk/agi-bin* directory. The full name of the failure script should be specified.

Dialed Number Manipulation Rules allow dialed numbers to be manipulated before they are passed to the trunk. Dialed numbers are altered based on the patterns entered into this field. There should only be one pattern per line. Only the first matched rule will be acted upon (other patterns will be ignored, even if they match). Patterns are matched based upon the order they are listed in, that is, from top to bottom. Patterns can consist of the following items:

Pattern	Description
X	Any whole number from 0-9
Y	Any whole number from 1-9
N	Any whole number from 2-9
[#]	Any whole number or letter in brackets. Note that multiple numbers can be separated by commas, and a range of numbers can be specified with a dash, so [1,3,6-8] would match the numbers 1, 3, 6, 7, and 8
.	Matches one or more characters (acts as a wildcard)
		Removes a prefix from the dialed number (555|1234567 would take 5551234567 and pass 1234567 to the trunk)
+	Add a prefix to the dialed number (555+1234567 would take 1234567 and pass 5551234567 to the trunk)

Note that wildcards are not valid before a "+" or "|". Also, "+" and "|" can be used in the same pattern (0|01+15551234567 would take 015551234567 and pass 0115551234567 to the trunk). Dial rules are very useful for allowing users to continue dialing numbers the way they are accustomed to, even if the provider is expecting them to be delivered in a different way.

A few real-world examples of dial pattern usage are given below:

Scenario	Dial Pattern(s)
Users are used to dialing 9 to access an outside line. A CompletePBX system replaces their existing PBX, but should accommodate this dialing pattern instead of forcing users to learn a new one. The provider is expecting all of the numbers to be sent in standard North American 10 digit format	9|1+NXXNXXXXXX 9|1NXXNXXXXXX
The provider requires an account code to be sent in front of all of the dialed calls (in this example, the account code is 654321). Calls must be sent in standard North American 10 digit format	5432101+NXXNXXXXXX 543210+1NXXNXXXXXX

Scenario	Dial Pattern(s)
The provider requires all of the numbers to be sent in standard North American 10 digit format except for toll free calls. Toll free calls should never send the leading 1. In this example, note the order that items are listed in. If the last item were listed first and a user were to dial *800-555-1234*, a 1 would be appended and sent to the provider, and the call would fail.	1|888NXXXXXX 1|866|NXXXXXX 1|877|NXXXXXX 1|800|NXXXXXX 1+NXXNXXXXXX

Dial Rules Wizards are pre-constructed dial patterns for use in the **Dial Rules** field. Selecting the name of one of the pre-made patterns will populate the **Dialed Number Manipulation Rules** box with the appropriate pattern.

Outbound Dial Prefix can be used in order to add a prefix to every single call dialed through this trunk. This option can be useful if accessing an outside line requires dialing an extra number (usually 9). This option can also be useful for certain DAHDI cards that are slow to pick up the line and therefore, start dialing before the line is listening for the digits. Adding a "w" as the outbound dial prefix will cause the system to wait for 50 milliseconds before dialing (multiple "w" characters can be strung together to create a longer wait). Anything that is valid in a CompletePBX dial string command can be entered into this field.

DAHDI trunks

Beyond the common trunk fields, DAHDI trunks only have one additional field to fill in as shown in the following screenshot:

Outgoing Settings

DAHDi Identifier ❓ :   g0

Submit Changes   Duplicate Trunk

DAHDI Identifier should match the group or the channel number that is set up for this DAHDI trunk in the */etc/asterisk/dahdi-channels.conf* file. *dahdi-channels.conf* will have a section like the ones below for each DAHDI channel that is present in the system:

```
context=from-zaptel
group=0
signalling = fxs_ks
channel => 1
context=from-zaptel
group=0
signalling = fxs_ks
channel => 2
```

Entering a value of **1** in the **DAHDI Identifier** field will use channel 1 for this trunk.

There are various ways to define the channel groups for trunks: gN, GN, rN, RN where N is a channel group number. The group number prefix has the following meaning:

- **g** – use the next free channel, starting with the lowest free number in the group
- **G** – use the next free channel, starting with the highest free number in the group
- **r** – select the next channel in the group, using round-robin method, starting from the lowest channel number
- **R** – select the next channel in the group, using round robin method, starting from the highest channel number

When all of the required values have been entered, click on the **Submit** button to write the changes to the database. As with all of the changes, click the red-colored **Apply Config** bar to load the new configuration changes into memory.
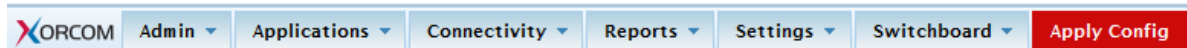
IAX2 and SIP trunks

IAX2 and SIP trunks share the same additional configuration settings as shown in the following screenshot:

Trunk Name is a "friendly" name. This value is not parsed by CompletePBX in any way; it is simply

---

used to refer to the trunk in other parts of the CompletePBX interface.

PEER Details are the outbound configuration details that would normally be placed into *iax_additional.conf* or *sip_additional.conf* for IAX2 or SIP trunks, respectively. These details will vary widely from trunk to trunk, and should be supplied by the trunk provider. By default, CompletePBX populates this field with the most commonly required values.

In the **PEER Details** field, **host** is the IP address or DNS hostname of the provider. This is the destination server or network that CompletePBX will send calls to when they use this trunk. The **username** and **secret** lines are for the credentials used to authenticate this trunk against the provider. The **type** line should almost always be set to **peer**, as this designates that this set of details is for a destination that CompletePBX is sending calls to (not receiving calls from).

The **USER Context** field is the telephone number, account number, or account name that your provider is sending inbound calls to. Some providers will simply send a call to the registered IP address, in which case this field is not required.

The **USER Details** field is for the inbound configuration details that would normally be placed into the *iax_additional.conf* or *sip_additional.conf* files for IAX2 or SIP trunks, respectively. As with the **PEER Details** field, these details can vary widely, but will likely be supplied by the trunk provider.

In the **USER Details** field, the **secret** line is the **password** that the provider will use to authenticate incoming calls against. The **type** line should almost always be set to **user**, as this designates that this set of details is for a peer that CompletePBX is receiving calls from (not sending calls to). The **context** line indicates the context that inbound calls from this trunk should be sent to. Unless you have defined custom contexts, leave this field set to **from-trunk** to allow CompletePBX to process the calls.

Order is important

The order in which the parameters are specified for both the **PEER Details** and **USER Details** fields is important. Details are set from top to bottom. If a certain parameter appears twice in one of these fields, the second instance of it will be used and the first will be ignored.

If **allow=ulaw** is followed by **deny=all**, all of the codecs will be denied and the **allow** statement will be ignored. Pay close attention to how these parameters are ordered. It can often save hours of troubleshooting later on when a trunk seems to be refusing calls for no reason at all

The **Register String** is also a setting that should be supplied by the trunk provider. The value in this field will cause CompletePBX to attempt to register with the trunk provider. Registration lets the provider know where to send calls when they come in for your telephone number. The register string is usually in the format of [username]:[password]@[voipserver] (for example, 5551234567:password@sip.example1.com). If a register string is not provided then the provider must be set up in order to send calls to a specific IP address, and the IP address of your VoIP server must not change.

ENUM trunks

ENUM trunks require absolutely no additional configuration beyond the common configuration fields. The only item worth noting for ENUM trunks is that all of the calls are queried against the *e164.org* database. *e164.org* expects calls to be in the format of [CountryCode][PhoneNumber], so adjust dial patterns appropriately for your location.

DUNDi trunks

DUNDi trunks require a bit of prep work before they can be successfully used to route calls. Before a DUNDi trunk can be set up, your CompletePBX must be peered with one or more nodes. Peering DUNDi nodes are not set up within the CompletePBX interface. Peering is completed by editing the */etc/asterisk/dundi.conf* file on both systems.

Once the PBX systems have been peered, CompletePBX can be used to create a DUNDi trunk. Beyond the common fields, DUNDi trunks only require one additional piece of information as shown in the following screenshot:

The **DUNDi Mapping** field tells CompletePBX which DUNDi context to query for results. It is possible to have various DUNDi contexts for sharing different types of VoIP routes. One might have one DUNDi context for sharing local extensions between branch offices, and another context for advertising which external phone numbers they own. Each DUNDi context will require its own DUNDi trunk within CompletePBX.

To find the name of the DUNDi context, open the */etc/asterisk/dundi.conf* file for editing, and look for the *[mappings]* section. The section should look like the following:

```
[mappings]
priv => dundi-
localextensions,0,IAX,priv:${SECRET}@pbx.examplehost.com/${NUMBER},nopartial pub =>
dundi
publicnumbers,0,IAX,priv:${SECRET}@pbx.examplehost.com/${NUMBER},nopartial
```

In this example, *priv* is the name of the DUNDi context that advertises local extensions. CompletePBX will advertise any extension that exists under the *dundi-localextensions* dialplan context if a peer queries it using this context. Likewise, *pub* is the public DUNDi context that will advertise any telephone route listed in the *dundi-publicnumbers* dialplan context. Type the name of the DUNDi context that this trunk will query into the **DUNDi Mapping** field and click on the **Submit** button. Once the changes have been saved, click on the red-colored **Apply Config** bar at the top of the screen.


Custom trunks

Custom trunks are very similar to custom extensions and devices. There is only one additional field for setting up custom trunks as shown in the following screenshot:

The **Custom Dial String** can be any valid dial command that would normally be used in the CompletePBX dialplan, with one notable exception: the called number is inserted into the dial string using the *$OUTNUM$* variable. For example, the string to call an H.323 device at IP address 192.168.1.2 might be *H323/192.168.1.2/$OUTNUM$*.

As with any changes made to CompletePBX, make sure to click on the **Submit Changes** button, followed by the red-colored **Apply Config** bar at the top of the screen to save and reload all of the settings.
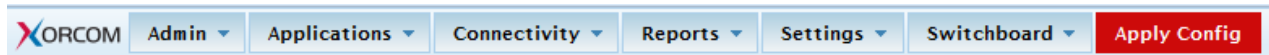
Summary

By now we should have a CompletePBX system with some fully-functional trunks configured. We should be able to configure a new trunk of any type, and be able to check the status of our trunks using the CompletePBX command line interface (CompletePBX CLI), if necessary. We should also have a good idea about how to monitor the health of each trunk, and how to alert an administrator on failure.

Now that we have operational trunks, the next chapter will walk through setting up various call targets so that incoming calls coming in through our brand new trunks have somewhere to go.

4. Basic Call Targets

A call target in CompletePBX is any kind of destination for a phone call that enters the system. A target might be a voice-response menu that prompts the caller to press various digits to route their call, a ring group to call a number of endpoints, or virtually any other type of processor to route the call in whatever way is desired. A call may have several targets throughout its lifespan.

The following types of call targets are available in CompletePBX:

- Terminate Call
- Extension
- Voicemail
- Ring Groups
- Conferences
- Day Night Mode
- Phonebook Directory
- Queues
- Time Conditions
- IVR (Digital Receptionist)

This chapter will cover the first seven targets (the "basic" call targets) of the above list. The more advanced call targets will be covered in the next chapter. Basic call targets generally have a single, easy-to-configure function, whereas advanced call targets tend to have several functions and are more complicated to configure.

Chapters 4 and 5 deal with understanding and configuring the different types of call targets. These call targets are put to use through the configuration of inbound routes and users' follow-me settings. Once all the required call targets have been configured, please see Chapter 6, **Call Routing** for information on routing inbound calls to the new call targets.

Terminating Calls

CompletePBX provides several clever ways to terminate calls. We may wish to terminate a call because the calling number is a known telemarketer, or perhaps the caller has attempted to reach an invalid extension too many times. Whichever the case, the following options are available for the **Terminate Call** target:

- **Hangup**: Terminates the call immediately without any further call processing or signaling.
- **Congestion**: Sends a signal indicating that there are no channels available to complete the call before terminating it. This is used only for digital channels (such as SIP, IAX, or PRI). Some providers will use this signal to locate an alternate route or process the call in another way.
- **Busy**: Relays a busy tone to the caller. The call terminates when the calling party hangs up (CompletePBX will play a busy tone indefinitely).
- **Play SIT Tone** (Zapateller): Plays a SIT known as an operator intercept to the calling party. An operator intercept tone consists of two short tones and one long tone. It is designed to indicate that the called number is not in service. SIT tones are recognized by a large portion of automated dialing equipment, so this tone is often useful for playback to known telemarketing numbers. Automated dialers who understand this tone will usually remove the called number from their dialing lists. Once the intercept tone is played, the call will be terminated immediately.
- **Put caller on hold forever**: As one might imagine, it places the calling party on hold until they terminate the call themselves.

- **Play ringtones to caller until they hangup**: It continually plays back a ringing tone to the calling party until they terminate the call themselves.

Be aware of indefinite termination options!

The **Busy**, **Put caller on hold forever**, and **Play ringtones to caller until they hangup** options open up the possibility of a call never terminating until the PBX or the calling party loses connectivity and the call is dropped.

This could possibly mean a call that lasts for weeks. With VoIP providers charging by the minute, such calls can get expensive!. The latter two options are typically used to annoy unwanted callers.

Indefinite ringtones or hold music are relatively safe for these kinds of calls as the caller will terminate the call themselves when they do not receive a human on the other end of the line.

Nevertheless, be aware of the activity on your PBX to avoid being billed for a week-long session of hold music. When in doubt, terminate the call rather than opting to leave it on hold indefinitely.

Extensions and Voicemail

The **Extension** target will route calls directly to a particular extension. The only option available for the Extension target is to which extension to send a call.

If CompletePBX is in Extensions mode, an extension must have been previously created for this target to be available. If CompletePBX is in DeviceAndUser mode, a user must have been previously created for this target to be available.

The **Voicemail** target will route calls directly to an extension's voicemail box without ringing the extension first. The Voicemail target provides options of selection such as **busy**, **unavail**, or **no- msg**. These options control the outgoing message that will be played prior to playing the standard instructional message asking the caller to leave a message. If the selected message does not exist, a standard message stating that the person at the called extension is either on the phone or unavailable will be played. For the voicemail target to be available, the **Voicemail & Directory** option must be enabled when setting up an extension or user.

Ring Groups

A ring group allows one call to ring any number of endpoints. It is typically used for a particular department or section of a building. A company might use a ring group to ring all phones in the sales department. A home might use a ring group to ring all phones on a particular floor. Ring groups must be set up prior to selecting them as a call target. To set up a ring group, navigate to the **Applications** menu and select the **Ring Groups** dialog. The **Add Ring Group** screen will be displayed as shown in the next screenshot:

Add Ring Group

Ring-Group Number: ❓   `600`
Group Description: ❓   `_____`
Ring Strategy: ❓   `ringall ▾`
Ring Time (max 300 sec) ❓   `20`
Extension List: ❓   `[          ]`

Extension Quick Pick ❓   `(pick extension) ▾`
Announcement: ❓   `None ▾`
Play Music On Hold? ❓   `Ring ▾`
CID Name Prefix: ❓   `_____`
Alert Info ❓ :   `_____`
Ignore CF Settings ❓ :   ☐
Skip Busy Agent ❓ :   ☐
Enable Call Pickup ❓ :   ☐
Confirm Calls ❓ :   ☐
Remote Announce: ❓   `Default ▾`
Too-Late Announce: ❓   `Default ▾`

Change External CID Configuration

Mode: ❓   `Default ▾`
Fixed CID Value: ❓   `_____`

Call Recording

Record Calls ❓   `Always` `On Demand` `Never`

Destination if no answer:

`== choose one == ▾`

Each field or dropdown should be configured to ensure the ring group behaves in the desired manner.

The **Ring-Group Number** is a number that can be dialed from internal endpoints to access this ring group. Users can either dial this number directly, or transfer a caller they have on the line to this number. The Ring-Group Number can be thought of as the extension number of the ring group.

The **Group Description** field is just used to identify this ring group when selecting it as a call target. The contents of this field are not visible to callers in any way.

The **Ring Strategy** dropdown defines the way in which the endpoints in this ring group will be rung. Endpoints can be rung using the following methods:

- **ringall**: Ring all group members listed in the Extension List field at the same time.
- **ringall-prim**: If the first extension listed in the Extension List is not in use, all group members will be rung at the same time. If the first extension is in use or is in the do-not- disturb mode, none of the extensions will be rung and the call will immediately fail over to the target selected in the **Destination if no answer** section.

- **hunt**: Ring each group member one at a time in the order that they are listed in the Extension List field.
- **hunt-prim**: Functions similar to the ringall-prim method. If the first extension is in use, no group members will be rung. Otherwise, each extension is rung in the same fashion as in the hunt method.
- **memoryhunt**: Rings the first extension listed in the Extension List field, then rings the first and second extensions listed in the Extension List field, then rings the first, second, and third extensions listed in the Extension List field (and so on until all extensions are ringing or a timeout is reached).
- **memoryhunt-prim**: If the first extension is in use no group members will be rung. Otherwise, each extension is rung in the same fashion as in the **memoryhunt** method.
- **firstavailable**: Rings the first extension in the Extension List that is available to ring. If an extension has call waiting enabled, it will be considered "available", even if it is already on a call.
- **firstnotonphone**: Rings the first extension in the Extension List that is completely unoccupied. Call waiting settings are not taken into account. If an extension is already on a call, it will not be rung whether or not call waiting is enabled.

The **Ring Time (max 300 sec**) field is for defining the number of seconds that the ring group will be rung before failing over to the **Destination if no answer** field. For the hunt type of strategies, the ring time is applied to each stage of the hunt sequence.

For example, if the Ring Time (max 300 sec) field is set to "30" and the Ring Strategy field is set to "memoryhunt", the first extension in the Extension List will be rung for 30 seconds. Following this, the first and second extensions will be rung for 30 seconds, then the first, second, and third extensions will be rung for 30 seconds, and so on until the end of the Extension List is reached.

Extension List contains a list of all extensions to ring. Extensions should be listed one per line. Available internal extensions can also be selected from the **Extension Quick Pick** dropdown to insert them into the Extension List. Any extension listed here will be rung directly and Follow Me Settings will not apply. To force the ring group to respect Follow Me Settings, append the pound (#) symbol to the extension number (for example, 5001#).

External phone numbers or extensions can be added to a ring group by suffixing them with a pound symbol as well (for example, 11234567890# would dial 1-123-456-7890 whenever the ring group is called). As long as a trunk exists to route the number, it will be dialed as part of the ring group. Note that external numbers will take longer to connect, so ring groups that include them are likely to have longer ring times.

The pound symbol is also used to include internal resources that are not extensions to the group. In this fashion, it is possible to ring other ring groups or queues simply by adding their number to the Extension List and suffixing it with a pound.

The message selected in the **Announcement** dropdown will be played to the caller entering the ring group prior to any extensions being rung.

The music on hold category selected in the **Play Music On Hold?** dropdown will be played to the caller while the ring group members are being rung. Selecting the **Ring** option will cause the caller to hear the ringing instead.

The **CID Name Prefix** field will prefix the caller ID of all calls to this ring group with the value specified here. This is useful for the person receiving the call as they can determine if a caller is attempting to call them directly or is calling a larger group.

The **Alert Info** field is used to transmit some information in the *ALERT_INFO* SIP headers to a SIP endpoint. Some SIP endpoints can be set up to play alternate ringtones depending on the *ALERT_INFO* data they receive. If the endpoint being rung does not use SIP technology, this field is

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ignored for that endpoint.

When the **Ignore CF Settings** option is checked, any extension in the Extension List that has call forwarding enabled will not be rung.

When the **Skip Busy Agent** option is checked, any extension in the Extension List that is in use will not be rung. This option will ignore call waiting settings, so multi-line or single- line phones with call waiting enabled will not be rung if they are in use at all.

The **Confirm Calls** option will prompt the person who answers the call to press *1* on their phone before the call is actually bridged to their extension. This is often useful for ring groups that contain external phone numbers in the Extension List. If one of the external phone numbers goes to voicemail, the PBX will see that the call has been "answered" and will bridge the caller to the external number even though a real person did not answer the call. If the call is not confirmed by the external party, the call will continue ringing the rest of the Extension List as normal. The Confirm Calls option will only work with the ringall ring strategy.

The **Remote Announce** and **Too-Late Announce** dropdowns are only applicable if the Confirm Calls option is enabled. The message in the Remote Announce field is played to the person who picks up the call (this should be a message asking them to press *1* to confirm the call). The message in the Too-Late Announce field is played to the person who picks up the call if someone else in the ring group picked up the call before they were able to confirm it. Default messages for these two fields do not exist and must be recorded. More information on recording custom voice prompts can be found in the *Custom Voice Prompts* section of Chapter 8,

Personalizing Your PBX

The **Destination if no answer** section is used when nobody in a ring group picks up the call within the designated ring time – see some of the available options in the following screenshot:
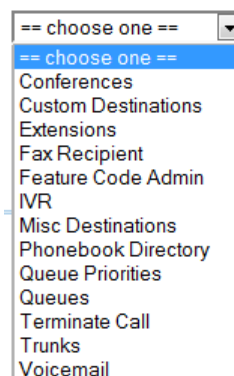
The destination chosen is a normal call target. The options available in this section will vary depending on which types of targets have already been configured in CompletePBX.

Once all options have been configured, click on the **Submit Changes** button, followed by the red-colored **Apply Config** bar.
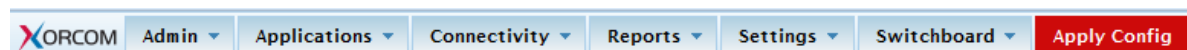
Conferences

Conferences allow two or more calls to be joined together so that all parties on the call can hear one another. At least one conference room must be configured before this target option is available. To

configure a conference room, navigate to the **Applications** menu and select the **Conferences** dialog.

The first section of the Add Conference screen requires a number and description for the conference room that is being created:

Add Conference

Add Conference

Conference Number: ❓ [   ]
Conference Name: ❓ [   ]
User PIN: ❓ [   ]
Admin PIN: ❓ [   ]

The **Conference Number** is a number that internal endpoints can dial to reach this conference. Like the ring groups, this can be thought of as the extension number of the conference.

The **Conference Name** is meant to identify the purpose of the conference room. The name is used only when selecting the room as a call target, and is not visible by the caller in any way.

The **User PIN** is a numeric passcode that is used to enter the conference room. If a PIN is entered in this field, no one is able to join the conference room without entering the PIN.

The **Admin PIN** functions in the same way as the **User PIN**. The Admin PIN and User PIN should not be set to the same value. The Admin PIN is used in conjunction with the **Leader Wait** option explained further in this chapter, in order to identify the administrator or leader of the conference.

The second section of the Add Conference page sets up general conference options that determine how the conference room behaves when in use:

Conference Options

Join Message: ❓ [None ▾]
Leader Wait: ❓ [No ▾]
Talker Optimization: ❓ [No ▾]
Talker Detection: ❓ [No ▾]
Quiet Mode: ❓ [No ▾]
User Count: ❓ [No ▾]
User join/leave: ❓ [No ▾]
Music on Hold: ❓ [No ▾]
Music on Hold Class: ❓ [inherit ▾]
Allow Menu: ❓ [No ▾]
Record Conference: ❓ [No ▾]
Maximum Participants: ❓ [No Limit ▾]
Mute on Join: ❓ [No ▾]

[Submit Changes]

The **Join Message** option is a message that will be played to anyone attempting to join the conference. The caller will hear this message before their channel is bridged into the conference room.

If the **Leader Wait** option is set to "Yes", the conference will not begin until the conference administrator joins the conference room. The administrator is identified by the Admin PIN option. If

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other callers join the conference room before the leader does, they will hear on-hold music or silence until the conference begins (what they hear depends on the **Music on Hold** option explained later in this section). If this option is set to "No", the callers will be bridged into the conference as soon as they call the conference room number.

If the **Quiet Mode** option is set to "No", a tone will be played whenever a caller enters or exits the conference room.

If **User Count** option is set to "Yes", the number of users currently in the conference room will be announced to the caller before they are bridged into the conference.

If **User join/leave** option is set to "Yes", the caller will be prompted to say their name and press the pound (#) key before joining the conference. The existing room occupants will hear the name of the new caller being played back before the user is bridged into the conference room.

If the **Music on Hold** option is set to "Yes", on-hold music will be played if there is only one caller in the conference room or the conference has not started yet (because the leader has not arrived). If this option is set to "No**"**, no sound will be played during these situations.

If **Allow Menu** is set to "Yes", conference participants are able to press the CompletePBX (*) key on their phones to be presented with an audio menu that only they can hear. The menu gives them the ability to mute/unmute their channel, increase/decrease their transmit volume, and increase/decrease their receive volume. If this option is set to **No**, pressing the CompletePBX key will do nothing. The available menu options and their functions are listed in the following table:

Menu Option	Action
1	Toggles mute for the user. When enabled, anything the user says is not transmitted to the rest of conference members. If the conference is being recorded, anything said by a muted user is not part of the recording
2	Toggles the conference lock. When a conference is locked, no more callers may join. A locked conference must be unlocked for any new users to join. This option is only available to a conference administrator. If the conference does not have an admin PIN configured or the user has joined the conference as a user instead of an admin, this option is not available
3	Ejects the last user who joined the conference from the conference room. The user will hear a message informing them that they have been ejected from the conference and that their call will be terminated. Note that if a conference is unlocked, the user may rejoin. The best way to remove an abusive conference user is to eject them and then immediately lock the conference. This option is only available to a conference administrator. If the conference does not have an admin PIN configured, or the user has joined the conference as a user instead of as an admin, this option is not available
4	Decreases receive volume. The user can tap this option to decrease the volume of what they are hearing. This does not affect what any other conference member hears. If a user is finding other conference members too loud, they can press *4* a few times while in the conference menu to make the conference quieter for themselves

Menu Option	Action
5	Increases receive volume. The user can tap this option to increase the volume of what they are hearing. This does not affect what any other conference member hears. If a user is having trouble hearing other members of the conference, they can press *6* a few times while in the conference menu to make the conference louder for themselves
6	Decreases transmit volume. The user can tap this option to decrease the volume of what they are transmitting to the rest of the conference members. When this option is used, the user will sound quieter to all other conference members. If a user is much louder than the other members of a conference room, they can tap *7* a few times while in the conference menu to make their transmit volume quieter.
7	Increases transmit volume. The user can tap this option to increase the volume of what they are transmitting to the rest of the conference members. When this option is used, the user will sound louder to all other conference members. If the conference members are having trouble hearing a particular user, that user can tap *9* a few times while in the conference menu to make their transmit volume louder.

If the **Record Conference** option is set to **Yes**, all the channels that are bridged together for the conference will be recorded into a single WAV file. The file will be stored in the */var/spool/asterisk/monitor* folder, and the filename will have a time and date stamp of when the conference ended.

Summary

By this point, we should have a good understanding of each of the different types of basic call targets, and should likely have several targets set up and ready to accept calls. We should be able to create call targets in order to do the following:

- Terminate calls
- Ring an extension directly
- Drop a call directly to voicemail (without ringing an extension first)
- Ring a group of phones using various ring strategies
- Conference multiple users together
- Route calls between two different destinations using Day Night Mode controls
- Search the phonebook directory for all users with a voicemail box on the system

In the next chapter, we will cover the advanced call targets of queues, time conditions, and IVRs.

5. Advanced Call Targets

Advanced call targets in CompletePBX require a bit of additional set up before they can be used. They often generate complicated call flows. However, for the most part, CompletePBX developers have encapsulated this complexity into a straightforward interface. Advanced call targets allow callers to be directed to queues (so that a caller hears music until all of the calls in front of them have been cleared and someone is available to take their call), or to digital receptionists routing calls based on input from the caller. Advanced call targets also allow calls to take different paths depending on the date, day of the week, or time of the day.

In this chapter, we will be discussing advanced call targets in CompletePBX, which are as follows:

- Queues
- Time Conditions
- IVR (Digital Receptionist)

Queue

A queue is a "line up" or "stack" of calls that need to be answered. By default, CompletePBX queues work in a First In, First Out (FIFO) fashion (the first caller who enters the queue is the first caller to be picked up out of the queue). Call queues are useful for scenarios in which the volume of callers is expected to exceed the number of people available to answer calls. Scenarios such as a technical support line or a sales line are good examples. Callers who are placed into a queue will hear music while on hold until someone is available to answer their call.

It is often helpful to think of queues in terms of a physical store, wherein, there are a small number of employees to serve a large number of people. Imagine you are in a line at the bank with ten other people and there are only three tellers. The three tellers are unable to serve all of the ten people at once, and it would be rude for them to help someone from the middle of the line before the person who is next in the line.

Each person is called up to an available teller in the order that they entered the line. In North America, this is generally called "waiting in line", but in some countries this is actually referred to as a queue. Such is the case with a telephone queue. If you are the fifth caller to enter a queue, then you will be the fifth caller to be picked up by an available user.

Queue Priorities

By default, any queue created in CompletePBX will treat callers with a FIFO methodology. However, there may be situations, when certain callers should be prioritized in order to enable shorter wait times. Going back to the analogy of a physical bank, those ten customers waiting in line may see a new customer enter the bank and walk straight past the line to be serviced by an available teller. The customer may be a business customer instead of a personal banking customer, or they may simply be paying for a higher level of service. Whatever the case may be, they are allowed to bypass all of the other customers in order to receive faster service. Queue priorities allow particular callers to be weighted differently than others, in order to enable this kind of prioritization.

The default setting for all of the callers is to have a priority of zero. Callers with a priority above zero will be placed in front of priority zero callers. The highest priority call will be placed first when ordering calls in the queue, and the order will be decided by descending priority. Queue priorities are often used when **service level agreements (SLAs)** are present, which stipulate that certain callers must have their call answered within a certain timeframe. Setting those callers up with a higher

priority will ensure they are placed at the front of the queue and that their call will be answered more quickly than calls from those without SLAs. CompletePBX allows call priorities to be set between zero and twenty.

Queue priorities act as their own call target. A caller is directed to a queue priority target and is then immediately redirected to another call target. Once passing through a queue priority target, a call will hold its newly assigned priority for any queue that it enters (a caller only needs to be prioritized once, after which they can enter as many queues as they need to without fear of losing their priority number).

In order to set up a new queue priority target, navigate to the **Connectivity** menu and select the **Queue Priorities** dialog. The queue priorities edit screen will be displayed as follows:

The edit screen has two main sections as follows:

- Edit Queue Priority Instance
- Destination

The first section allows for a description (used when selecting this priority as a target) and the actual priority itself. In this example, the queue priority is being set up in order to prioritize bronze level callers. Callers are given a priority of **5**, placing them above callers with no priority, but leaving plenty of room for callers with a higher-level SLA.

The **Destination** section is for selecting the call target that the caller will be sent to, now that they are prioritized. Queue priorities that do not have queues as their eventual destinations are essentially useless. So, be sure that the caller will eventually wind up in a queue if they are sent through a queue priority target. This usually means sending the caller directly to a queue or an IVR where a queue is a possible option.

When the **Description**, **Priority**, and **Destination** fields have been set, click on the **Submit Changes** button in order to save the priority, and then click on the red-colored **Apply Config** bar in order to have the changes take effect.
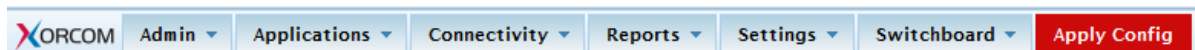
Queue priorities are usually set up as the first call target for inbound calls. Inbound routes can be set up in order to recognize a particular caller's caller ID and then route them to the appropriate call priority. Alternatively, if customers are given different phone numbers to call based upon their service level, then an inbound route can send calls to those numbers to specific queue priorities.

Queues

Queues can be very simple to set up for basic functionality, but advanced options and behaviors

come with a bit of a learning curve. Queues have over thirty options, making them the most complicated call target available in CompletePBX. In order to set up a new queue, navigate to the **Applications** menu and select the **Queues** dialog.

Important note

if you want the Switchboard to track the status of queues and to create statistics for call center reporting, make sure that both **Event When Called** and **Member Status Event** (in the **Events, Stats and Advanced** section of the Queue dialog) are set to **Yes** when defining a new queue.

Important note: if you want the Switchboard to track the status of queues and to create statistics for call center reporting, make sure that both **Event When Called** and **Member Status Event** (in the **Events, Stats and Advanced** section of the Queue dialog) are set to **Yes** when defining a new queue

Events, Stats and Advanced

Event When Called: Yes ▾

Member Status Event: Yes ▾

Add Queue

The **Add Queue** section has several options and an **Extension Quick Pick** drop-down menu as shown in the following screenshot:

Add Queue

Add Queue

Queue Number: ❓

Queue Name: ❓

Queue Password: ❓

Generate Device Hints: ❓ ☐

Call Confirm: ❓ ☐

Call Confirm Announce: ❓ [Default ▼]

CID Name Prefix: ❓

Wait Time Prefix: ❓ [No ▼]

Alert Info❓ :

Static Agents: ❓

Extension Quick Pick ❓ [(pick extension) ▼]

Dynamic Members: ❓

Extension Quick Pick ❓ [(pick extension) ▼]

Restrict Dynamic Agents ❓ [Yes | No]

Agent Restrictions ❓ [Call as Dialed ▼]

Queue Number can be thought of as the "extension" assigned to the queue. If a caller is calling into the queue from an endpoint on this system, or someone is transferring a call to the queue, then this is the number they will use. The **Queue Number** is also used to sign in and out of the queue. If an employee is not permanently a member of the queue, they can dial this **Queue Number** plus *** to log into the queue, and this **Queue Number** plus **** to log out (for example, a Queue Number of 5000 would have agents dial 5000*** to log in and 5000**** to log out).

Queue Name is used to identify the queue when selecting it as a call target. This name is not parsed by CompletePBX.

If **Queue Password** is not left blank then queue agents will be required to enter this password before they are able to join the queue. This option is used in conjunction with the **Queue Number** (agents will be prompted for the password after dialing the **Queue Number** plus ***). This setting is optional.

The **CID Name Prefix** is used to prefix the caller ID of callers in the queue with the value entered in this field. This is often used when agents are members of multiple queues so that they know which department a person has called (for example, a CID may be prefixed with "Sales" or "Tech Support").

Wait Time Prefix is used to prefix the caller ID of callers in the queue with the number of minutes the caller has been waiting in the queue. The time will be rounded off to the nearest minute, and will be prefixed with the letter "M". If the call is subsequently transferred to another queue with this feature set, the time count will reset.

Alert Info is used to send specific *ALERT_INFO* SIP headers to SIP endpoints that support them. Many SIP endpoints can be set up to ring differently, or automatically answer calls when specific

ALERT_INFO values are received. For *ALERT_INFO* headers to trigger an action on an endpoint, the *ALERT_INFO* header that the phone is configured to look for must match the *ALERT_INFO* header that CompletePBX is sending. Some endpoints will allow the information headers to be configured on the device, while others have these header values statically configured so they cannot be changed. The documentation that accompanies a SIP endpoint should list whether or not the device supports *ALERT_INFO* headers, as well as if the headers can be configured, or are predefined.

Static Agents are extensions that are always members of the queue. Static agents do not need to log into the queue, and cannot log out of the queue. As long as an agent listed in this field is available, their endpoint will be rung when a caller is in this queue. Extensions should be listed one per line. Remote telephone numbers or extensions can also be added to this field. Note that unlike ring groups, external numbers can be added as they would be dialed from a normal endpoint on the system, and do not require a pound symbol at the end.

Advanced users can add agents from their */etc/asterisk/agents.conf* file to the **Static Agents** field by prefixing their agent number with the letter "A" (for example, agent number 2223 would be listed as A2223). For the time being, this functionality is considered experimental and may cause various issues with terminating and transferring calls that are received by these agents.

Any agent listed in the Static Agents field can have a penalty value appended to their extension after a comma (for example, agent 2223 with a penalty value of 1 would be listed as "2223,1"). Listing an agent without a penalty value assigns the agent a penalty of zero. Penalties affect when certain agents are called. The higher the penalty an agent has, the less likely they are to receive a call from the queue. An agent with a penalty of one will only be rung if nobody with a penalty of zero is available. Likewise, an agent with a penalty of two will only be rung if nobody with a penalty of zero or one is available. Penalties are useful for adding agents to the queue who would not normally be taking these type of calls, but should be available to handle overflow of calls if the need arises.

The **Extension Quick Pick** list allows any configured extension to be selected from the drop-down menu. The selected extension will be copied into the **Static Agents** field with a penalty of zero.

Queue Options

The **Queue Options** section has just under twenty options available for configuring the queue as shown in the following screenshot:

The **Agent Announcement** drop-down menu is used to select a sound file to be played back to the agent who answers the call before the call is bridged to them. This file must be pre-recorded or

uploaded to the system through the **System Recordings** CompletePBX section (more information on creating custom recordings for the Agent Announcement field can be found in the *Custom Voice Prompts* section in Chapter 8, **Personalizing Your PBX**. This announcement is optional, but may serve the purpose of telling the agent which queue the call came from (if CID is not available to provide this information).

The **Join Announcement** drop-down menu is used to select a sound file to be played back to the caller before they are dropped into the queue. This file must be pre-recorded or uploaded to the system through the **System Recordings** section of CompletePBX. This announcement is optional, but will usually state that the caller is being placed into a queue, and that their call will be answered in priority sequence.

The **Music on Hold Class** drop-down menu is used to configure the music that is played to the caller while they wait in the queue. The **inherit** option can be selected in order to have the queue use the same music on hold class already assigned to the call (usually this is set through an inbound route). Music on hold classes can be configured in the **Music on Hold** section of CompletePBX (more information on creating custom recordings for the **Agent Announcement** field can be found in the *Custom Music on Hold* section in Chapter 9).

If the **Ringing Instead of MoH** box is checked then callers will hear ringing instead of Music on Hold while they wait in a queue. Use this option carefully; a long wait time can mean dozens of ring tones and often a caller who hangs up (thinking their call is never going to be answered). If this option is enabled, the option selected in the **Music on Hold Class** drop-down menu is ignored.

Max Wait Time is the amount of time that a caller is able to wait in the queue before the call is considered failed and routed to the failover destination. Set it to **Unlimited** in order to configure the queue to use an unlimited wait time. This field has a maximum value of one hour otherwise.

Max Callers is the maximum number of callers allowed to enter the queue at any time. Callers over this limit will immediately be sent to the failover destination. Enter **0** to configure the queue for an unlimited number of callers. This field has a maximum value of 50 otherwise.

If **Join Empty** is set to **Yes**, then callers are able to join the queue and wait even if no agents are signed into the queue. The **Strict** option will not allow callers to join the queue if either no agents are signed in or all of the signed in agents are unavailable. The **No** option will not let callers join the queue if no agents are signed in, but it *will* allow callers to join if all of the signed in agents are unavailable. Callers that are not allowed to join the queue because of Strict or No settings are immediately sent to the failover destination.

The **Leave When Empty** drop-down menu carries the same settings as the **Join Empty** drop-down menu. Leave When Empty is employed when there are suddenly no more agents signed into the queue *after* a caller has already joined the queue.

The **Ring Strategy** drop-down menu determines the way that agents will be called when a caller enters the queue. It has the following options:

- **ringall**: All of the agents are rung until one answers.
- **roundrobin**: Agents are rung one at a time in the order they are listed in the **Ring Strategy** textbox until one of them picks up.
- **leastrecent**: It is similar to **roundrobin**, but starts with the agent who was least recently called by this queue. Note that CompletePBX does not account for call length, so the timer that counts how long ago an agent was rung starts counting from when an agent picks up a call, not from when the call is completed. An agent who is on a call for an hour is still considered to have been rung an hour ago.
- **fewestcalls**: Rings the agent who has answered the fewest calls from this queue.
- **random**: Rings a random agent.

- **memory**: It is similar to roundrobin, but remembers the last agent who was rung in the last round robin and starts by calling them first.

Agent Timeout is the number of seconds an agent's phone can ring before the call is considered failed for that agent. Set this to **Unlimited** in order to ring the agents continuously until somebody answers the call. This field has a maximum value of 60 seconds otherwise. Note that this field will be overridden if either the general ring time setting or an extension's own follow-me ring time setting is lower than the specified timeout. For example, if the CompletePBX general ring time setting is set to 30 seconds, and **Agent Timeout** is set to 45 seconds, then the maximum amount of time that an agents phone will ring is 30 seconds. Also, setting this field to **Unlimited** will only work for the **ringall** ring strategy. Other strategies will ring extensions for 60 seconds (or the value of the general ring time setting or the extension's follow-me ring time, whichever is shorter) before carrying on to ring the next agent.

The **Retry** value is the number of seconds the system will wait before calling an agent after the last attempt timed out. Selecting **No Retry** will send the caller to the failover destination as soon as the first attempted agent times out (additional agents will not be rung). This field has a maximum value of 20 seconds.

Wrap-Up-Time is the number of seconds an agent is considered unavailable after they have completed a call from this queue. Setting this option to **0 seconds** means no delay. This field has a maximum value of 60 seconds.

If **Call Recording** is not set to **No** then calls to this queue will be recorded in the selected format and saved to */var/spool/asterisk/monitor*. Calls to a queue that are recorded will be named using the format*: q(QueueNumber)-(Date)-(Time)-(CallID).(Format).* The tokens should be replaced with the following values to locate the file:

Token	Replaced with
(QueueNumber)	The number or extension of the queue. The number of the queue is configured in the **Queue Number** field
(Date)	The date that the call was answered on. The date is in the format of YYYYMMDD
(Time)	The time that the call was answered. The time is in the format of HHMMSS, and is in 24 hour format
(CallID)	A unique ID assigned to the call by CompletePBX. The ID is not referenced anywhere in the CompletePBX interface. Hence, if two calls come into the same queue at the exact same time, they will have to be heard in order to determine which call is which.
(Format)	The appropriate file extension for the format of the call. WAV and WAV49 formats will have an extension of .wav; the GSM format has an extension of .gsm

For example, a call that came into queue number 11 on July 19, 2009 at 7:05:45 p.m. might have a filename of *q11-20090715-190545-1247699145.4355.wav*.

If **Event When Called** is set to **Yes**, the events generated during the progress of a call to this queue include AgentCalled, AgentDump, AgentConnect, and AgentComplete. These events can be used by

third-party software programs, such as **My Switchboard**, in order to track call progress and generate call metrics.

Similar to **Event When Called**, when **Member Status** is set to **Yes** the QueueMemberStatus event will be generated during call progress.

If **Skip Busy Agents** is set to **Yes**, then agents who are on a call will not be rung when there is a caller in the queue. This option applies to extensions with call waiting active, or multi-line endpoints. Even though these endpoints have the ability to receive multiple calls at the same time, if they are in use they will not be rung.

Queue Weight gives all of the callers in the queue a particular priority. This weighting is used by CompletePBX to determine which call to deliver to agents signed into multiple queues. If an agent is signed into multiple queues and both queues have callers waiting in them, the queue with the higher Queue Weight will ring through to the agent first.

The **Autofill** option is only applicable to CompletePBX version 1.4 and later versions. This option allows multiple callers to be sent to multiple agents at the same time. For example, if three callers are in the queue, and three agents are available, enabling this option will send each caller to one of the agents. If this option is disabled then only one call is processed at a time, and every caller must wait to be first in line before any agent is rung for their call. This option has no effect in CompletePBX version 1.2. This option is not affected by the selected ring strategy (agents are still rung in the order defined by the ring strategy).

The **Agent Regex Filter** can be used to restrict the extensions of agents who are allowed to sign into the queue. If this field is left blank then it is possible for any agent who knows the queue number and password to sign into the queue. Entering a regular expression in this field can restrict the agents who can log into the queue to specific extensions or ranges of extensions.

Regular Expressions

Regular expressions can be very complicated. It is a good idea to have a thorough understanding of how to craft regular expressions before attempting to use the Agent Regex Filter field. An invalid regular expression in this field could make the queue entirely inaccessible to all of the agents. More information on regular expressions can be found at http://en.wikipedia.org/wiki/Regular_expression

Caller Position Announcements

The **Caller Position Announcements** section has three configurable options as shown in the following screenshot:

This section is used to control how often a message is played to each caller in the queue detailing their position in the queue and how long it is estimated that they will be on hold.

The **Frequency** drop-down menu controls how often this type of announcement is made. Setting this to **0 seconds** disables the announcements entirely. This field has a maximum value of 20 minutes otherwise.

If **Announce Position** is set to **Yes** then the caller will be told their place in the queue each time the position announcement is played.

If **Announce Hold Time** is set to **Yes**, then the caller will be told their estimated hold time each time this message is played. If this field is set to **Only Once**, then the estimated hold time will be played to the caller when they first join the queue, but never again. When estimated hold times are less than one minute, hold times are not announced.

If both **Announce Position** and **Announce Hold Time** are set to **No**, neither announcement will ever be played no matter to what the **Frequency** field is set.

Periodic Announcements

Periodic announcements can be played to each caller in order to give them the option to perform another task. Usually this means that an IVR is presented to the caller, and the caller is given the option to leave a voicemail or to reach a different department. The **Periodic Announcements** section has two configurable options as shown in the following screenshot:

IVR Break Out Menu is the IVR that should be presented to the caller each time the announcement is played. The recording set for the selected IVR is what will be played to the caller. The selected IVR must only contain single-digit options.

Repeat Frequency is how often the IVR should be presented to the caller. Setting this to **0 seconds** disables the periodic announcements. This field has a maximum value of 20 minutes otherwise.

The user can only select one **Fail Over Destination** as shown in the following screenshot:

Select the call target where callers should be directed if their call to this queue fails for any reason. A call could be considered failed because no agents were signed into the queue, the caller waited in

the queue for too long, or no signed in agents were available to answer the call (depending on the configured options).

When all of the options have been configured for the desired queue behavior, click on the **Submit Changes** button in order to save the queue, and then click on the red-colored **Apply Config** bar in order to load the queue into the running CompletePBX configuration.
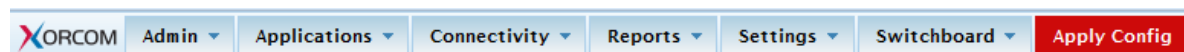
Time-based Rules

Time conditions are a set of rules for hours, dates, or days of the week. A condition has two call targets each time. Calls sent to a time condition will be sent to one target if the time of the call matches one of the conditions, or to the other target if none of the conditions match. Each time condition can have multiple time definitions (known as time groups). Time conditions are often used to control how a phone system responds to callers inside and outside of business hours, and during holidays.

Time Groups

Before we can set up a time condition call target, we need to define a set of time groups. Time groups are a list of rules against which incoming calls are checked. The rules specify a specific date or time, and a call can be routed differently if the time it comes in matches with one of the rules in a time group. Each time group can have an unlimited number of rules defined. It is useful to group similar sets of time rules together. For example, there may be one time group for business hours in which the time that the business will be open will be defined. Another popular time group is for holidays, in which each holiday that falls on a business day is defined.

In order to create a new time group, navigate to the **Applications** menu and select the **Time Groups** dialog. The **Add Time Group** screen will be displayed as shown in the following screenshot:

The **Description** field is used to identify this time group, when selecting it during the setup of a time

---

condition. This value is not parsed by CompletePBX.

The **New Time** section is used to define a time rule that a call will be matched against. Leaving a field as the default dash (-) will match all of the values (for example, **Week Day Start** to **Week Day Finish** will match on every day of the week). Keep in mind that all of the conditions must be true for the rule to match, so it is possible to create rules that never match (setting **Month Day start** and **Month Day finish** to 31 and **Month start** and **Month finish** to September will never match as there is no 31st of September).

Once the appropriate rule has been defined, click on the **Submit** button in order to save the rule. Additional time rules can be added to a time group by clicking on the group's name in the time group list on the right and selecting additional rule parameters in the **New Time** section at the bottom of the page. An example of a time group with multiple rules might be a group defining business hours as shown in the following screenshot:

In this example, the condition will match Monday through Friday, 8:30 a.m. through 5:30 p.m. and on Saturday from 9:30 a.m. through 3:30 p.m.

Time Conditions

Once a time group has been defined, a time condition can be set up as a call target. In order to create a new time condition, navigate to the **Applications** menu and select the **Time Conditions** dialog. The **Add Time Condition** screen will be displayed along with four configuration sections. The first section is the **Add Time Condition** section as shown in the following screenshot:

Add Time Condition

Add Time Condition

Time Condition name: ❓ [_____]
Generate BLF Hint ❓   ☐
Enable Override Code ❓   ☐
Time Group: ❓   --Select a Group-- ▾

Destination if time matches:

[== choose one == ▾]

Destination if time does not match:

[== choose one == ▾]

[Submit]

The **Time Condition name** field is used for identifying the time condition when selecting it as a call target. This value is not parsed by CompletePBX.

The **Time Group** drop-down menu is used to select the time group that this time condition should be referencing for time rules. When a call reaches this time condition, the rules in the selected Time Group will be parsed in order to determine where to send the call next.

Click on the **Submit Changes** button in order to save the queue, and then click on the red-colored **Apply Config** bar to load the queue into the running CompletePBX configuration.

| **XORCOM** | Admin ▾ | Applications ▾ | Connectivity ▾ | Reports ▾ | Settings ▾ | Switchboard ▾ | **Apply Config** |
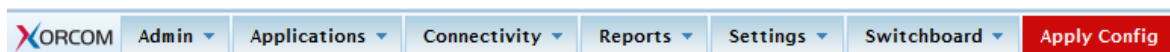
The last two sections, **Destination if time matches** and **Destination if time does not match** are for setting up the call targets for calls that match the rules set up in the selected time group and for calls that do not.

Select the desired call destinations, click on the **Submit Changes** button in order to save the destination, and then click on the red-colored **Apply Config** bar in order to load the destination into the running CompletePBX configuration.

| **XORCOM** | Admin ▾ | Applications ▾ | Connectivity ▾ | Reports ▾ | Settings ▾ | Switchboard ▾ | **Apply Config** |
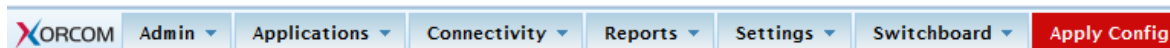
IVR (Digital Receptionist)

An **Interactive Voice Response** (**IVR**) system is often referred to as a digital receptionist. An IVR plays back a pre-recorded message to the caller that asks them to press various buttons on their telephone depending on which department or person they would like to speak with. The IVR system will then route the call accordingly.

CompletePBX IVRs allow any digits to be defined as destinations (for example, pressing one might route to the sales ring group). A destination of **t** can also be defined to route the call if the IVR times out without receiving any input. A destination of **i** can be defined to route the call if the IVR receives

invalid input.

In order to create an IVR, navigate to the **Applications** menu and select the **IVR** dialog. The **Digital Receptionist** screen will appear with two configuration sections: **IVR options** and **IVR destinations**. The **IVR options** section has configurable options as shown in the following screenshot:

Add IVR

- IVR General Options

IVR Name ❓
IVR Description ❓

- IVR Options (DTMF)

Announcement ❓	None
Direct Dial ❓	Disabled
Timeout ❓	10
Invalid Retries ❓	3
Invalid Retry Recording ❓	Default
Append Original Annoucement ❓	☐
Invalid Recording ❓	Default
Invalid Destination ❓	== choose one ==
Timeout Retries ❓	3
Timeout Retry Recording ❓	Default
Append Original Annoucement ❓	☐
Timeout Recording ❓	Default
Timeout Destination ❓	== choose one ==
Return to IVR after VM ❓	☐

- IVR Entries

Ext	Destination	Return ❓	Delete
digits pressed	== choose one ==	☐ ❓	🗑

Submit

Change Name is used as a name for identifying the IVR. This field is not parsed by CompletePBX.

The **Announcement** drop-down menu is used to select a pre-recorded message that will be played to the caller as they enter the IVR. Recordings must be pre-recorded or updated in the **System Recordings** CompletePBX section.

Timeout is the number of seconds that the system will wait for input from the caller. If this number of seconds passes without input, the call will fail over to the **t** destination (if defined), or the call will be disconnected.

If **Enable Directory** is checked then callers can press the pound (#) key to enter a directory system that will allow them to search for a person by first name or last name. Otherwise, pressing the

pound key will play back a message stating that the caller has provided invalid input.

If **VM Return to IVR** is checked then a caller who transfers from the IVR to a voicemail box will be transferred back to the IVR when they are done leaving a message. If this option is not selected then the caller will be disconnected after leaving a voicemail.

If multiple directory contexts are defined, the directory that should be accessible to the caller when the pound key is pressed can be selected from the **Directory Context** drop-down menu.

The **Enable Direct Dial** option enables the callers to dial an extension directly from the IVR. If this option is disabled then the callers will receive a message stating that they have provided invalid input when they enter an extension, *even if the extension is valid*.

Timeout Message is the message that will be played to the caller if they fail to enter any input before the number of seconds specified in the **Timeout** box. This message will only be played if the **t** destination is not defined.

Invalid Message is the message that will be played to the caller if they enter invalid options while in the IVR. This message will only be played if the **i** destination is not defined.

Repeat Loops is used to select the number of times the IVR will repeat itself when no valid input is received. After the specified number of loops, the caller will be disconnected. The maximum number of loops allowed is nine.

The **IVR destinations** section allows multiple key presses to be mapped to specific destinations:

When all of the IVR options are configured for the desired behavior, click on the **Save** button to save the IVR, and then click on the red-colored **Apply Config** bar to load the IVR into the running CompletePBX configuration.
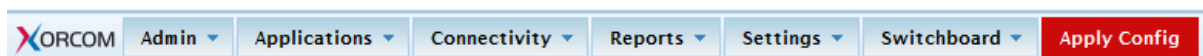
Conferences

Conferences enable multi-party conferencing that is available as a call destination.

Add Conference section

Navigate to the **Applications** menu and select the **Conferences** dialog.

Conference Number is the number that local users can dial to join the conference.

Conference Name is used as an Identifier (along with the number) when picking a conference as a destination.

If **User PIN** password is set, anyone calling into the conference will be prompted to enter a PIN. If **User PIN** is left blank, they can just press **#** to enter the conference.

If **Admin PIN** password is set, the conference bridge cannot be opened until a user enters the **Admin PIN**. If **Music On Hold** is enabled, users will be placed on hold with the default Music On Hold class until someone enters the **Admin PIN.**

Conference Options section

The **Join Message** is a sound that is played to all users upon entering.

If **Leader Wait** is checked on and the **Admin PIN** is set, the conference will not start until someone enters the **Admin PIN**.

Usually a sound is played when a user enter or leaves the conference, alerting other members to the fact that someone has joined or left. You can disable that by setting the **Quiet Mode** to **Yes.**

Setting the **User Count** to **Yes** will cause the conference to play a message saying that "'There are (number) people in this conference" when someone joins the call.

When someone connects to the conference, they will be asked to record their name if **User Join/Leave** is set to **Yes**. The conference will then announce their name when they join and leave the conference.

Music On Hold can be used to enable or disable **Music on Hold** in this conference.

The **Allow Menu** enables the user or admin to enter the management mode by pressing on *****. Commands available in management mode are:

Code	Action
1	Mute yourself.
2	Lock or unlock the conference. (Only available to the person, if any, who entered the conference using the Admin PIN.)
3	Eject the last person that joined the conference. (Only available to the person, if any, who entered the conference using the Admin PIN.)
4	Decrease the Conference Volume (e.g. decrease the level of sound you hear).
6	Increase the Conference Volume (e.g. increase the level of sound you hear).
7	Decrease your Volume (e.g. the level of sound other people hear).
9	Increase your Volume (e.g. the level of sound other people hear).

You can set **Record Conference** to **Yes** if you would like to record the conference.

Edit Conference

Conference Number: ❓ `9001`

Conference Name: ❓ `Technical Support`

User PIN: ❓ ` `

Admin PIN: ❓ `120546`

Conference Options

Join Message: ❓	None ▼
Leader Wait: ❓	Yes ▼
Talker Optimization: ❓	No ▼
Talker Detection: ❓	No ▼
Quiet Mode: ❓	No ▼
User Count: ❓	Yes ▼
User join/leave: ❓	Yes ▼
Music on Hold: ❓	No ▼
Music on Hold Class: ❓	inherit ▼
Allow Menu: ❓	Yes ▼
Record Conference: ❓	Yes ▼
Maximum Participants: ❓	5 ▼
Mute on Join: ❓	No ▼

Submit Changes

When all of the conference options are configured for the desired behavior, click on the **Save** button to save the conference, and then click on the red-colored **Apply Config** bar to load the conference into the running CompletePBX configuration.

| XORCOM | Admin ▼ | Applications ▼ | Connectivity ▼ | Reports ▼ | Settings ▼ | Switchboard ▼ | **Apply Config** |
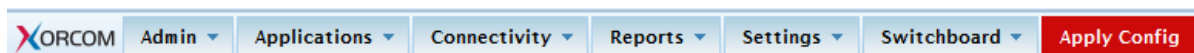
Paging & Intercom

Paging allows you, with phones that support it, to page a number of extensions. You dial a number, and all the phones in the paging group pick up automatically, go into hands-free mode, and play what the caller is saying through the speaker. This is very useful in a small office environment to announce that Lunch is ready.

To add a paging group, simply put in the Paging group number - this is the number that people will dial to page the group, and the list of devices, one per line, that are to be paged. This module is for specific phones that are capable of Paging or Intercom.

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 should work with Xorcom IP phones as well as most other SIP phones (not ATAs). Any phone that is always set to auto-answer should also work.

Navigate to the **Applications** menu and selecting the **Paging & Intercom** dialog.

Paging Extension is the number you will dial to talk to a paging group.

Group Description is a descriptive name to help you identify a paging group.

Busy Extensions enables you to determine how to handle paged extensions that are busy, with options to **Skip**, **Force**, or **Whisper** to the extension. Each option affects busy extensions only – other extensions will be paged in the normal manner.

- **Skip** will not page any busy extension. All other extensions will be paged as normal
- **Force**" will not check if the device is in use before paging it. This means conversations can be interrupted by a paging call, depending on how the device handles it. This is useful for emergency paging groups.
- Whisper" will attempt to use the ChanSpy capability on SIP channels, resulting in the paging call being sent to the device's earpiece, and being "whispered" to the user, but not heard by the remote party. If ChanSpy is not supported on the device or otherwise fails, no page will get through. It probably does not make much sense to choose duplex if using Whisper mode.

Paging is typically one way for announcements only. Checking on **Duplex** will make the paging call duplex, allowing all phones in the paging group to be able to talk and to be heard by all. This makes it like an instant conference.

Modify Paging Group

| Paging Extension | 9501 |

| Group Description | Management Team |

Selected	Not Selected
2001 - VP Marketing	1001 - DAHDI channel 1
2002 - VP Sales	1002 - DAHDI channel 2
2003 - VP Support	1003 - DAHDI channel 3
2020 - CEO	1004 - DAHDI channel 4
	1005 - DAHDI channel 5

Busy Extensions ❓

Skip | **Force** | **Whisper**

Duplex ❓ ☑

Default Page Group ☐

Submit   **Delete**

When all of the paging options are configured for the desired behavior, click on the **Save** button to save the paging group, and then click on the red-colored **Apply Config** bar to load it into the running CompletePBX configuration.

XORCOM | Admin ▾ | Applications ▾ | Connectivity ▾ | Reports ▾ | Settings ▾ | Switchboard ▾ | **Apply Config**

Parking

This module allows you to configure the settings for the parking lot functionality of CompletePBX. Navigate to the **Settings** menu and select the **Parking Lot** dialog.

Parking Lot Configuration

Parking Lot Options

Enable Parking Lot Feature ❓   ☐

Parking Lot Extension: ❓   70

Number of Slots: ❓   8 ▾

Parking Timeout: ❓   45 seconds   ▾

Parked Music Class: ❓   default ▾

Parking Lot Context: ❓   parkedcalls

Enable BLF Capabilities: ❓   ☑

Use Next Slot: ❓   ☐

Enable ADSI Announcements: ❓   ☐

Returned Call Behavior:

Pickup Courtesy Tone: ❓   ○ Caller ○ Parked ⦿ Both

Transfer Capability: ❓   ⦿ Caller ○ Parked ○ Both ○ Neither

Re-Parking Capability: ❓   ⦿ Caller ○ Parked ○ Both ○ Neither

One Touch Recording Capability: ❓   ○ Caller ○ Parked ○ Both ⦿ Neither

DTMF Based Hangup Capability: ❓   ○ Caller ○ Parked ○ Both ⦿ Neither

Parking Alert-Info: ❓

CallerID Prepend: ❓

Announcement: ❓   None   ▾

Alternate Destination:

Return Destination Behavior: ❓   ⦿ Originator ○ Alternate Destination

== choose one == ▾

Submit Changes

An especially useful part of this module is the ability 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, the call is diverted to the chosen destination which is any of the standard destinations. Prior to sending the call to that destination, you can configure options to further identify the

orphaned call:

- Parking Alert-Info, to provide a unique ring for the returned call
- CallerID Prepend (to identify the call with additional CID information)
- Announcement (to be played to the orphaned caller to reassure them that call is still active)

Parking Lot Options section

Enable Parking Lot Feature should be checked on if you want to enable the parking feature.

Parking Lot Extension defines the extension number where you will transfer a call to, in order to park it.

Number of Slots defines the total number of parking lot spaces to configure. For example, if 70 is the extension and 8 slots are configured, the parking slots will be 71-78.

Parking Timeout is the period that a parked call will attempt to ring back to the original parker if the call is not answered.

Parked Music Class is the music class that will be played to a parked call while in the parking lot UNLESS the call flow prior to parking the call explicitly set a different music class, such as if the call came in through a queue or ring group.

Parking Lot Context is the call context. You should not change it from the default unless you know what you are doing.

Enable BLF Capabilities should be checked on to have 'hints' generated to use with BLF buttons.

Use Next Slot should be checked on if you want the parking lot to seek the next sequential parking slot relative to the last parked call instead of seeking the first available slot. Normally this box is left un-checked.

Enable ADSI Announcements should be checked on if you are using ADSI enabled analog phones and want to have ADSI parking announcements sent to the phones. Other adsi.conf configurations may be necessary for this to work.

Returned Call Behavior section

Pickup Courtesy Tone defines to whom to play the courtesy tone to when a parked call is retrieved. This can be the caller, the parker, or both.

Transfer Capability toggles DTMF based transfers when picking up a parked call.

Re-Parking Capability enables or disables DTMF based parking when picking up a parked call.

One Touch Recording Capability enables or disables DTMF based one-touch recording when picking up a parked call.

DTMF Based Hangup Capability enables or disables DTMF based hangups when picking up a parked call.

Parking Alert-Info to be added to the call prior to sending back to the Originator or to the Alternate Destination.

Caller ID Prepend is the string to prepend to the current Caller ID associated with the parked call prior to sending back to the Originator or the Alternate Destination.

Announcement is an optional message to be played to the call prior to sending back to the Originator or the Alternate Destination.

Alternate Destination section

Return Destination Behavior determines where to send a parked call that has timed out. The call can be sent back to the original device that parked the call, or it can be sent back to the alternate destination. In both cases, any configured Alert-Info, Caller ID prepend, or announcement will be applied to the call. If configured to send back to the Originator and they are not available (phone is offline), the alternate destination will be used.

When all of the parking options are configured for the desired behavior, click on the **Save** button to save the configuration, and then click on the red-colored **Apply Config** bar to load it into the running CompletePBX configuration.
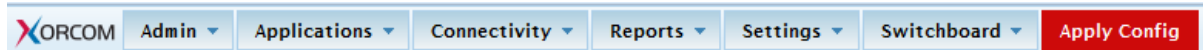
Summary

By this point, we should have a good understanding of advanced call targets. We should be able to route calls using complicated time conditions into an IVR system that eventually routes the caller into a call queue, if we desire to do so.

The next chapter will put our call targets into action with inbound call routes. We will also discuss setting up outbound routes (in order to utilize the trunks we set up a few chapters ago), in addition to setting up least cost routing in order to properly route calls over the cheapest available trunk.

6. Call Routing

Up until this point, we have discussed various call targets and what will happen to calls when they reach those targets. In this chapter, we will discuss the following:

- Inbound routing
- Follow Me and the VmX Locater
- Outbound routing
- Least Cost Routing (the ability to pick the cheapest route for a call, based on the dialed destination)

Inbound routing

Inbound routing is one of the key pieces to a functional CompletePBX phone system. Inbound routes in CompletePBX are where we finally get to put to use the call targets discussed in Chapters 4, **Basic Call Targets** and Chapter 5, **Advanced Call Targets**. CompletePBX allows two specific types of inbound routing:

1. DID-based routing: DID-based routing routes calls based on the trunk on which the call is coming in. CID-based routing routes calls based on the caller ID number of the person who is calling. Within those two routing methods, CompletePBX allows the detection of inbound faxes. **DID** (**Direct Inward Dialing**), in VoIP telephony, refers to a trunk and the telephone number associated with that trunk. As it is possible for a phone system to have several trunks, CompletePBX allows different routing rules to be set up for each trunk. This is commonly used when a company has a dedicated technical support phone number (which routes directly to their support department), while other calls to the company come in to a different phone number and are routed to an IVR or a receptionist.

2. CID-based routing: **CID** (**caller ID**) refers to the name and number of the person calling. CompletePBX allows inbound calls to be routed based on the number someone is calling from. This is commonly used to immediately disconnect calls from known telemarketers, but can also be used to route calls from specific people in a different way than routing calls from the general public.

These two routing methods can be used on their own or in conjunction with one another.

To set up inbound routing, navigate to the **Connectivity** menu and select the **Inbound Routes** dialog. A number of sections are present on the **Add Incoming Route** page:

- Add Incoming Route
- Options
- Privacy
- Call Recording
- Language
- Fax Detect
- CID Lookup Source
- Set Destination

The **Add Incoming Route** section has the following four options:

1. Description
2. DID Number
3. Caller ID Number

4. CID Priority Route

Add Incoming Route

Description🔵 :
DID Number🔵 :
CallerID Number🔵 :
CID Priority Route🔵 :

The **Description** field is used to hold a description to help you remember what this particular inbound route is for. This field is not parsed by CompletePBX.

The **DID Number** field is used when DID-based routing is desired. The phone number of the DID to be matched should be entered in this field. The DID number *must* match the format in which the provider is sending the DID. Many providers will send the DID information with the call as +15555555555, while others will leave out the country code information and simply send 5555555555. If the DID entered in this field does not exactly match with the number sent by the provider, then the inbound route will not be used. This field can be left blank to match calls from all DIDs (this will also match calls that have no DID information).

This field also allows patterns to match a range of numbers. Patterns must begin with an underscore (_) to signify that they are patterns. Within patterns, *X* will match the numbers 0 through 9, and specific numbers can be matched if they are placed between square parentheses. For example, to match both 555-555-1234 and 555-555-1235, the pattern would be _*555555123[45]*.

The **Caller ID Number** field is used when CID-based routing is desired. As with the DID Number field, the CID entered in this field must exactly match the format in which the provider is sending the CID. Providers may send 7, 10, or 11 digits; they may include a country code and the plus symbol. Check with your provider to see the format in which the CID is sent, in order to ensure that the field is entered correctly.

The Caller ID Number field can be left blank to match with all CIDs (this will also match calls that have no CID information sent with them). The field allows **Private**, **Blocked**, **Unknown**, **Restricted**, **Anonymous**, and **Unavailable** values to be entered, as many providers will send these in the CID number data.

Leaving both the DID Number and Caller ID Number fields blank will create a route that matches all calls.

Inbound routing priorities

It is possible to run into a situation in which a call matches several of the defined inbound routes. In this scenario, CompletePBX will give a route priority in the following sequence:
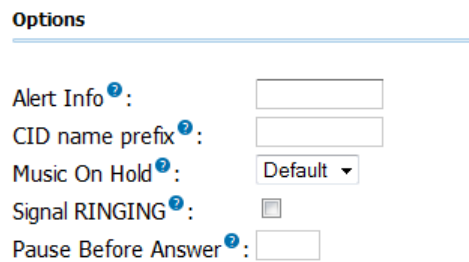
*   Routes with a specific DID and CID will always be first in priority.
*   Routes with a specific DID but no CID will be second in priority.
*   Routes with no DID, but with a specific CID will be third in priority.
*   Routes with no specific DID or CID will be last in priority.

It is important to note that by default, setting up a route for a specific DID will take preference over setting up a route for a specific CID.

The **CID Priority Route** toggle affects the inbound routing priority behavior. If this checkbox is selected, all calls from the number specified in the **Caller ID Number** field will be routed using the route even if there is a route for the DID on which the call came in. If there is a specific route that

specifies both DID and CID, then that route will still take precedence. Checking the CID Priority Route toggle simply swaps the second and third routing priorities for this particular route.

The **Options** section has five fields as shown in the following screenshot:

Alert Info is used to send a string of text in the SIP *ALERT_INFO* headers. This is often used for SIP endpoints that ring differently, or auto-answer calls based on the *ALERT_INFO* text that is received. Any inbound call that matches this route will send the text in this field to any SIP device that receives the call.
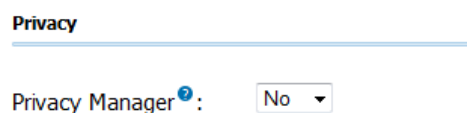
The **CID name prefix** field allows text to be prepended to the caller ID name information from the call. This is often used to identify where a call came from (calls to a number dedicated for technical support might be prefixed with "Tech").

The **Music On Hold** drop-down menu allows the music-on-hold class for this call to be selected. Whenever a caller who passes through this route is placed on hold, they will hear the music on hold defined in the class selected here. This is often used for companies that advertise in their music on hold and accept calls in multiple different languages. Calls to a French DID might play a music-on-hold class with French advertisements, while an English DID would play a class with English advertisements.

The **Signal RINGING** toggle will send "ringing" in the call progress data, before CompletePBX lets the other side know that the call has been answered. Some providers require this, while this can break functionality with others. Check with your provider to see if they require "ringing" to be sent as a call progress before the call is answered.

The **Pause Before Answer** field contains the number of seconds that CompletePBX should pause before answering the call. This is not useful for digital channels, but analog DAHDI channels may have security systems or fax machines installed in parallel with the VoIP system. In such cases, a tone is usually played within a few seconds of the call being picked up to identify that the call is not a voice call. Setting a delay of a few seconds allows this tone to be played and the other piece of equipment to seize the line for communications. If the line has not been picked up by the end of this delay, CompletePBX will answer the call.

The **Privacy** section only has one configurable field:

The **Privacy Manager** drop-down menu is used to enable or disable the CompletePBX privacy manager functionality. When enabled, calls that come in without an associated caller ID number will be prompted to enter their 10-digit telephone number. Callers will be given three attempts to enter this information before their call is disconnected.

The **Fax Detect** section has one configurable field:

Fax Detect

Detect Faxes ❓ :   No Yes

The **Fax Detect** toggle determines whether faxes should be detected on this route. If fax detection is activated, additional parameters can be configured and a dropdown will appear which is used to select the extension that the inbound faxes will be directed to. Typically, this extension is a DAHDI extension that has a physical fax machine plugged into it. However, it may also be a virtual extension answered by a software program on the PBX. The program will accept faxes and turn them into digital documents for review.

Fax Detect

Detect Faxes ❓ :   No Yes
Fax Detection type ❓ :   Dahdi ▼
Fax Detection Time ❓ :   4 ▼
Fax Destination ❓ :   == choose one == ▼

If **disabled** is selected, fax detection will not be used for calls that match this route. Any fax calls will be routed just like voice calls.

The **Fax Detection Type** drop-down menu determines which type of detection mechanism CompletePBX will use to detect the fax:

- **DAHDI** should be used when the fax is coming from a DAHDI trunk
- **SIP** should be used when the fax is coming from a SIP trunk

Enabling fax detection will cause CompletePBX to immediately answer the call and then listen for fax tones. CompletePBX will listen for fax tones for the number of seconds defined in the **Fax Detection Time** field. If no tones are heard within this time frame, the call is considered to be a voice call.

The **CID Lookup Source** section only has one configurable field as shown in the following screenshot:

CID Lookup Source

Source ❓ :   None ▼

The **Source** drop-down menu allows any predefined lookup source to be selected. Any calls that match this route will be checked against the source specified here, and will have their caller ID name changed if they match an entry in the source. Caller ID lookup sources are specified on the CompletePBX **CallerID Lookup Sources** page (more information on CallerID Lookup Sources can be found in the *CallerID Lookup Sources* section of Chapter 8,

Personalizing Your PBX). Lastly, the **Set Destination** section has one option to configure:
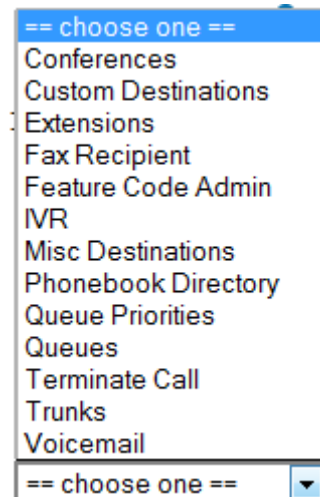
This is the place where the desired call target is selected. Any previously set up call target can be selected including extensions, IVRs, time conditions, conference rooms, and queues.

When all options have been selected, click on the **Submit** button to save the route. Clicking on the **Clear Destination & Submit** button will save the route without any associated destination. This is generally used for fax-only routes, where a fax must be detected or the call will be terminated. Once the route has been saved, be sure to click on the red-colored **Apply Config** bar at the top of the screen to make the route active.
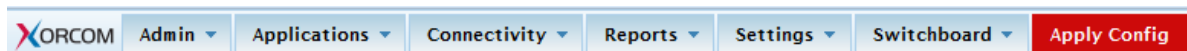
Follow Me and the VmX Locater

Follow Me will force CompletePBX to try and track a user down when their extension is called instead of simply ringing their assigned endpoint. This is often used to call a cell phone before dropping the caller to voicemail. With this enabled, the user never needs to give out their cell number because the system will always try to reach them there when their main endpoint is not answered. Likewise, if the cell number changes it simply has to be updated in one place and no calls are missed.

CompletePBX has two implementations of the "follow-me" system—the default Follow Me and the VmX Locater.

The default follow-me system is controlled by the CompletePBX administrator, and allows for varying ring strategies, ring times, music-on-hold selections, and various other complex settings. The VmX Locater is controlled by the end user, but is a much simpler implementation that consists of prompting the caller to press 0, 1, or 2 and directing calls to different numbers based on what was selected.

Default Follow Me

The default follow-me routing will ring a set number of extensions in a pre-selected ring pattern (similar to a ring group) before failing over to another call target. This could mean that several VoIP phones and a cell phone are rung at the same time, or they might be rung in order, until one of them picks up. If there is no answer, the call could fail over to voicemail, or be redirected to another person at the company.

To set up default follow-me routing in CompletePBX, navigate to the **Connectivity** menu and select the **Follow Me** dialog. Click on the user that requires Follow Me settings in the list on the right and

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the **Follow Me** screen is displayed as shown in the following screenshot:

The **Disable** checkbox will temporarily disable the follow-me setup while retaining the follow-me configuration. Any calls that are direct dial calls or calls that come from an IVR or a directory search will ring the extension directly instead of using the follow-me routing. Note that if follow me is specifically selected as a call target, the follow-me route will be respected for that call, regardless of whether this checkbox is selected.

The **Initial Ring Time** field is used to configure the number of seconds that the primary extension is rung before any extensions in the follow-me list are rung. This can be a value between 0 and 60. Setting this to 0 will cause CompletePBX to immediately start ringing the members of the follow-me list instead of trying to ring the primary extension first. As the primary extension can be included as a member of the follow-me list, this is often the desired behavior.

The **Ring Strategy** drop-down menu allows the order in which the follow-me list members will be rung to be configured. There are 10 available ring strategies. The ring strategies in a follow-me route

are identical to those used in a ring group. Please refer to the *Ring Groups* section of Chapter 4, **Basic Call Targets** for ring strategy definitions.

Ring Time (max 60 sec) configures the maximum number of seconds for which a phone will be rung before the call is considered failed. For hunt ring strategies, this is the number of seconds that each round of the hunt sequence will ring the appropriate phone.

Follow-Me List is the list of extensions that should be rung for this follow-me route. Extensions should be listed one per line. As with ring groups and queues, external numbers and extensions can be listed as long as they are suffixed with a pound (*#*) symbol. For example, 5551234567#.

The **Extension Quick Pick** drop-down menu lists all previously configured extensions on the system. Selecting an extension from this menu will place that extension into the **Follow-Me List**.

The **Announcement** option is the pre-recorded message that will be played to the caller before any of the Follow-Me List members are rung. This message can serve to inform the caller that several phones are being rung, or to ask them to be patient during the process (for example, "Please wait while the person you are trying to reach is located."). Recordings can be added in the CompletePBX **System Recordings** screen (more information on System Recordings can be found in the *Custom voice prompts* section of Chapter 8).

The **Play Music On Hold?** drop-down menu allows a selection of the music-on-hold class to be played for the caller while they are waiting for all follow-me list members to be rung. Selecting **None** will play nothing while extensions are tried. Selecting **Ring** will play ringing instead of music on hold. Music classes can be defined in the CompletePBX **Music on Hold** screen (more information on Music on Hold can be found in the *Custom Music on Hold* section of Chapter 08).

The **CID Name Prefix** field allows text to be prepended to the caller ID name of all of the callers who pass through this follow-me route.

The **Alert Info** allows custom text to be sent in the SIP *ALERT_INFO* headers for all calls that pass through this follow-me route. *ALERT_INFO* headers are often used to force a distinctive ring, or auto-answer on compatible SIP endpoints.

When **Confirm Calls** is enabled, then the person who answers a call from this follow-me route will be prompted to press *1* before the call will be bridged to their endpoint. Requiring the user to confirm the call avoids scenarios in which mobile phones send the call to voicemail, CompletePBX counts the call as "answered", and bridges the caller to the mobile voicemail (when the call should be sent to the failover destination instead).

Remote Announce is used to select the message that will be played to the person who picks up a call from this follow-me route if **Confirm Calls** is enabled. This message should inform the person receiving the call that they have an inbound follow-me call and need to press *1* to accept it.

Too-Late Announce is the message played to the person receiving the call if they try to confirm a call (by pressing *1*), but someone else has already answered the call. This message should inform them that the call they were trying to accept has been answered by someone else.

Once the follow-me settings have been configured, select a call target for the call to failover to in the event that none of the listed extensions pick up:

In many cases, the failover destination will be the voicemail for the user account that this follow-me belongs to (but the destination can be any valid call target on the system).

Once all options are configured, click on the **Submit Changes** button and then click on the orange-colored **Apply Config** bar at the top of the screen.
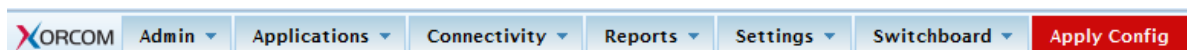
VmX Locater

The VmX Locater is a very simplified version of the default CompletePBX follow-me setup. These settings can be set and edited by the CompletePBX administrator, but if the locater is enabled for a user, then they will be able to change these settings themselves using **My Extensions**.

Note that the VmX Locater can work in conjunction with the CompletePBX follow-me routing, instead of replacing it entirely. The VmX Locator prompts are played when a caller would normally reach voicemail. At that point, the caller can be directed to press *0* for the operator, *1* to run through follow-me routing, and *2* for an optional additional extension.

To set up the VmX Locater in CompletePBX, navigate to the **Connectivity** menu and select either the **Users** or **Extensions** dialog. The dialog that is present will depend on the operational mode that CompletePBX is running in (DeviceAndUser or Extensions):

Click on the name of the user that requires VmX Locater configuration and scroll down to the bottom of the page to find the VmX Locater settings:

To enable the **VmX Locater**, select **Enabled** in the first drop-down menu.

The **Use When** checkboxes allow the locater to be engaged when the called user is **unavailable**, **busy**, or both. Note that if neither checkbox is selected, the locater will never be engaged.

The **Voicemail Instructions** checkbox configures whether or not to play the standard CompletePBX voicemail prompts after the user's pre-recorded outgoing voicemail message. If this box is not selected, the user's message will be played and will be immediately followed by a beep. Otherwise, the standard prompts asking the caller to "please leave their message after the tone" will be played.

The **Press 0**, **Press 1**, and **Press 2** fields can be used to configure specific extensions, ring groups, queues, or external phones to be rung when the appropriate digit is pressed during playback of the outgoing voicemail message. When the VmX Locater is enabled, the associated user should re-record their voicemail prompts to ask the user to press one of these three digits, in order to route the call appropriately. External phone numbers used in these fields *do not* require pound symbols at the end of them.

The **Press 0** field can optionally be configured to redirect the caller to the PBX operator (defined in the **Voicemail** section of the **Settings/General** dialog) instead of a specific extension or phone number.

The **Press 1** field can optionally be configured to redirect the caller to the default CompletePBX follow-me settings for the user (if they have also been configured).

When all settings have been configured, click on the **Submit** button followed by the red-colored **Apply Config** bar to save the changes and make them live.
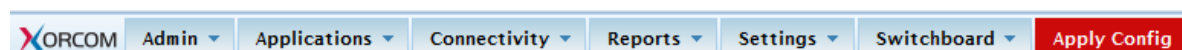
When the user is logged into **My Extension**, they can view and edit these settings by clicking on the **VmX Locator Settings** link in the navigation menu:

Outbound Routing and Least Cost Routing

Outbound routing is a set of rules that CompletePBX uses to decide which trunk to use for an outbound call. Many VoIP systems have multiple trunks, and it can often be unnecessarily expensive to route all calls over a single trunk. Outbound routing also allows dialed numbers to be rewritten on the fly (to remove or prepend dialed numbers with specific outside access codes or area codes). Routes are defined using patterns, against which the dialed numbers are matched.

Outbound routes have a priority. If a dialed number matches the pattern in two outbound routes, the route with the lower priority will be used to place the call.

To start setting up an outbound route, navigate to the **Connectivity** menu and select the **Outbound Routes** dialog. The **Add Route** page consists of configurable options and one "quick pick" drop-down menu used to populate certain fields as shown in the following screenshot:

Route Settings

Route Name❓: [_____]
Route CID:❓ [_____]   ☐ Override Extension❓
Route Password:❓ [_____]
Route Type:❓   ☐ Emergency ☐ Intra-Company
Music On Hold?❓   [default ▾]
Time Group:❓   [---Permanent Route--- ▾]
Route Position❓   [Last after Emergency   ▾]

Additional Settings

Call Limit❓:   [None   ▾]
PIN Set❓:   [None ▾]
+ Extension Routing❓:

Dial Patterns that will use this Route❓

[(prepend   ) + [prefix   ] | [[match pattern   ] / [CallerID   ]] 🗑]
[+ Add More Dial Pattern Fields]
Dial patterns wizards❓:   [(pick one)   ▾]

Trunk Sequence for Matched Routes❓

0 [▾]
1 [▾]
2 [▾]

[Submit Changes]   [Duplicate Route]

Route Name is just used to identify this route. This value cannot contain any spaces. The name is usually descriptive of the purpose of the route (for example, "local" or "international").

The **Route Password** field can be used to secure a route against unauthorized dialing. For example, a company may wish to restrict all long distance calls, or calls to 1-900 numbers. A numerical password can be placed in this field, or a direct path to a file containing a list of valid passwords (one password per line). When this field is not blank, all calls that pass through this outbound route will not be placed through any provider until the caller successfully enters a valid password.

The **PIN Set** field performs the same function as the **Route Password** field but uses previously created password sets. PIN sets can be configured in the CompletePBX **PIN Sets** screen (more information on PIN Sets can be found in the **PIN** Sets section of Chapter 8). If PIN sets will be used, the **Route Password** field must be left blank. If both the **Route Password** field is left blank and the **PIN Set** field is set to **None**, callers will be able to use this route without authentication.

The **Emergency Dialing** checkbox is used to specify that the route will *only* be used for emergency calls. When this option is set, any calls that pass through this route will set their caller ID to the emergency CID setting of the device the call originated from (if set). Do not select this option if the route may sometimes route emergency calls. Only select this option if the route will *always* be used to route emergency calls.

If the **Intra Company** option is checked, then the caller ID of the device that initiated the call will be

VoIPon   www.voipon.co.uk   sales@voipon.co.uk   Tel: +44 (0)1245 808195   Fax: +44 (0)1245 808299

preserved instead of having the caller ID rewritten to whatever is specified by the trunk. This option should be used when an outbound route is directing calls to another VoIP system from the same company, and the call is not passing over public telephony routes.

The **Music On Hold?** dropdown menu allows a specific music-on-hold class to be selected for calls that pass through this outbound route. Once a call is placed through this route, the music that is heard anytime the call is placed on hold will be defined by the class selected in this field.

If you have set up a call limit, it needs to be linked to an outbound route. Navigate to the **Connectivity** menu and select the **Outbound Routes** dialog, and navigate down to the **Additional Settings** section:

The **Call Limit** dropdown box will show all the call limits that were previously defined in the Outbound Call Limit module. Select the call limit that you want to associate with this outbound route.

The **Dial Patterns** field is the heart of every outbound route. Specific numbers can be matched here (for example, 911 or 999). Otherwise, patterns can be specified using the characters in the following table:

Pattern	Description
X	Any whole number from 0-9
Z	Any whole number from 1-9
N	Any whole number from 2-9
[###]	Any whole number in the brackets. Note that multiple numbers can be separate by commas, and ranges of numbers can be specified with a dash ([1, 3, 6-8] would match the numbers 1, 3, 6, 7, and 8.
.	Matches one or more characters (acts as a wildcard)

The **Trunk Sequence** drop-down menus are for configuring the order in which the trunks will be used when calls match the pattern specified in the **Dial Patterns** field. If the first trunk in the list is unavailable or congested, the call will fail over to the second trunk. This behavior will repeat until all listed trunks have been exhausted.

The **Dial patterns wizards** and **Trunk Sequence** fields are where the concept of **Least Cost Routing (LCR)** comes into play. LCR involves configuring each outbound route to send calls to the cheapest trunk first. This often involves setting up multiple outbound routes for similar styles of calls. For example, say that a CompletePBX system is configured with two SIP trunks—one from a provider that allows unlimited flat-rate calling to the state of New York and one from a provider that allows

unlimited flat-rate calls to the state of California. It seems logical that calls to the New York area codes should be routed via the New York trunk and calls to the California area codes should be routed via the California trunk. However, with a single route to match US-format calls, all calls will prefer a single trunk first and charges will apply when calls could be free. In this case, it would make sense to create two outbound routes.

The first route would have a pattern that would match all calls for New York state area codes (dialed in both local and long-distance formats). The **Trunk Sequence** set of fields would list the New York trunk first and the California trunk second.

The second route would have a pattern that matches all calls for California state area codes (again, in both local and long distance formats). The **Trunk Sequence** set of fields would be reversed to have the California provider first and the New York provider second:

In this manner, when CompletePBX routes a call to either New York or California, it will attempt to use the cheapest route first. The end goal is that all users on the system should be able to dial a number just as they normally would, without having to worry about manually picking the cheapest line or dialing the call in a special way. Whenever they dial a number, the PBX is smart enough to dial it properly, using the most economical trunk.

Once trunk options have been configured, click on the **Submit Changes** button followed by the red-colored **Apply Config** bar at the top of the screen to make the new outbound route live.
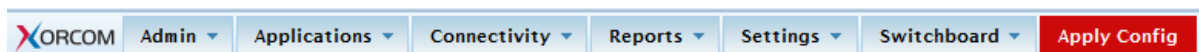
A note about routes

It is extremely easy to forget about a particular type of call and leave out the associated outbound route.

Doing so can leave users stranded when they need to make an important call.

Be sure to include methods to dial the following types of calls:

- **Emergency**: Dedicate a route just for this purpose. Calls for emergency services should never be mangled by another dial pattern.
- **Local**: Calls to local numbers (usually NXXXXXXXX).
- **Toll-free**: Calls to toll-free numbers (such as 1-888 or 1-800 numbers)
- **International**: Calls outside of the country, if permitted (usually 011)
- **Special: Calls** that do not fit any other category. This includes calls such as calls to the operator (0) and directory assistance (411)
- **Long distance**: Calls outside of the local calling area, if permitted (usually 1NXXXXXXXX)

Emergency Notifications

The Emergency Notifications module in CompletePBX is used to detect when someone is dialing an emergency call center, such as 911 in the United States. While the call is being processed, another call is made to an alternate destination to notify staff that someone just dialed an emergency number.

This feature is very useful in hotels, assisted living facilities, hospitals, and virtually any large office where someone dials 911 and the Xorcom PBX can immediately advise other staff members from which extension the call originated.

Note that the outgoing emergency call is not interrupted. The call will select the appropriate outside

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line based on what is programmed in outbound routes. CompletePBX detects when the emergency call is dialed, then initiates another call for notification.

Multiple emergency numbers may be created. For example, different numbers may exist for fire or medical emergency. This also comes in handy for international applications as this feature is not limited to a 911 code. Virtually any phone number can be used as an emergency code.

There is a hijack emergency call feature included in this module as well. While Xorcom does not condone interrupting a 911 emergency call for any reason, this feature could apply to large resort (typically found in the Caribbean or other parts of the world) that has their own emergency response facilities. Rather than the call going out to the regular police or hospital that may be a great distance away, the resort has the option of interrupting the call and rerouting it to their own internal emergency facilities, in order to provide faster response to their guests.

Important

When using analog extensions that use FXS Ports, select "generic DAHDi device" when creating the extension in order for it to work properly with this application. If the extensions already exist, make sure they all indicate "This device uses dahdi technology." under "Device Options".

Outbound Call Limit

Outbound Call Limit dialog allows you to place restrictions on how many times a number is called during a certain time period. For example, if you have an outbound route dedicated to international numbers and had Outbound Call Limit activated on that route, the Outbound Call Limit module would allow you to only make the allowed number of calls to any specific international number. If you exceed the call limit during the specified time period, any additional calls are blocked, and a recording would is played advising you that the call is not allowed.

Creating A Call Limit

Navigate to the **Connectivity** menu and select the **Outbound Call Limit** dialog.

Add Outbound Call Limit

An Outbound Call limit allows you to restrict the number of calls people make that match particular outbound routes.

Add Call Limit

Name:
Call Limit: 0
Date Range: 0
Recording: Default

Submit

For each call limit you can set the following options:

- **Name** is a text used to identify a call limit.
- **Call Limit** is set to a value between 0 and 20 to control the total number of times you want to allow users call a number on an outbound route that is being monitored by the Outbound Call Limit.

- **Date Range** can have a value between 0 and 20 to determine how many days back to count the calls. A value of 0 means that only calls made today will be counted, while a value of 1 would mean that calls made today and yesterday should be counted.
- **Recording** should be any system recording on CompletePBX that will be played to the user when they triy to call a phone number that has already exceeded the call limit.

Once the options have been configured, click on the **Submit Changes** button followed by the red-colored **Apply Config** bar at the top of the screen to make the changes take effect.

Linking Outbound Routes to Call Limits

If you have set up a call limit, it needs to be linked to an outbound route. Navigate to the **Connectivity** menu and select the **Outbound Routes** dialog, and navigate down to the **Additional Settings** section:

The **Call Limit** dropdown box will show all the call limits that were previously defined in the Outbound Call Limit module. Select the call limit that you want to associate with this outbound route.

Once the option has been configured, click on the **Submit Changes** button followed by the red-colored **Apply Config** bar at the top of the screen to make the change take effect.
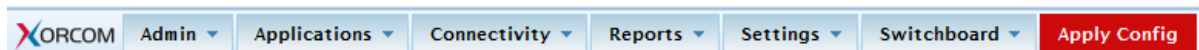
Summary

By this point, our CompletePBX system should be routing calls! This chapter discussed setting up inbound routes to match against the number that was dialed and the number that the caller is dialing from. Inbound calls can also be run against a follow-me route or a VmX Locater route in order to attempt to find a user who is not at their desk phone. Our CompletePBX system should also be able to detect and route inbound faxes, should any fax come through our trunks. In addition, outbound calls will now be routed properly using the most economical routes.

The next chapter will discuss taking all those calls that are now running through our system and recording them for playback later.

7. Recording Calls

CompletePBX has a wonderful, built-in ability to record calls. No additional software is required to make this happen. When CompletePBX records a call, both sides of the call are recorded and written out to a file for playback on a computer. Call recording is often performed in call centers to ensure call quality, or to keep calls for later review, should the need arise. CompletePBX provides the ability to record all of the calls, or to selectively record calls.

In this chapter, we will discuss the following:

- General recording options
- Recording calls to extensions
- Recording calls to queues
- Recording calls to conferences
- Maintaining call recordings

Know the law!

Before enabling call recording, make sure that you are aware of the legalities surrounding call recordings and privacy laws.

Call recordings are prohibited in certain places, unless the caller is told that the call will be recorded.

For example, in the state of California all of the parties on the call must consent to the call being recorded before it begins.

Recording formats

CompletePBX allows calls to be recorded in the following formats:

- WAV
- WAV49
- ULAW
- ALAW
- SLN
- GSM

Each format has its own ratio of file size to recording quality, and certain formats will not play on all of the computers. A comparison between all of the available formats is as follows:

Format	Description
WAV	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

Format	Description
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
ULAW or 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
SLN	CompletePBX native SLINEAR format.
	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
GSM	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

Transcoding during recording

One very important aspect of call recording to keep in mind is that if the recording uses a different codec than the original call, transcoding must occur. For example, a call that uses the G.711 ULAW codec that is being recorded using the WAV49 format will need to be transcoded into the GSM codec before being saved. Transcoding recordings place additional load on a server's disk I/O and processor resources. On high traffic systems, it is possible to max out all of the available resources if the transcoding recordings are not accounted for.

In general, the rule of thumb for recording will be the same as it is for selecting the codec that a trunk or extension will use: try to make everything match. If all of the calls are using the GSM codec, then it would be safe to record those calls in the GSM format. A system that has all of the calls in G.711 ULAW format would be put under an unnecessary amount of stress to record in GSM format. If transcoding is required (for example, limited disk space dictates the use of GSM recordings), be
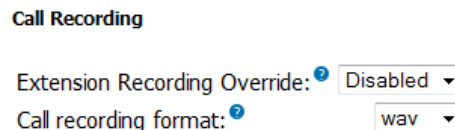
sure to size the resources of the system accordingly in order to allow the additional load.

General call recording options

CompletePBX has several settings that govern the global operation of call recording. These settings determine if call recording should be globally enabled or disabled, which format to record calls in, and a command to run after a call has been recorded (if desired).

In order to access the global options, navigate to the **Settings** menu and select the General dialog.

The **Call Recording** section has the settings that we are interested in:

Call Recording

Extension Recording Override: ❓ Disabled ▾
Call recording format: ❓   wav ▾

The **Extension Recording Override** option allows all user-level automatic call recording to be disabled. A user can be set up to record all of the calls (which will be discussed shortly). If this option is set to **Enabled**, all of the user settings will be ignored and calls will not be recorded automatically. Note that this option neither affects recording settings for queues or conferences, which can be set up in order to record all of the calls, nor does it affect on-demand recording. This setting only affects users who have been set up to automatically record all of the calls. If this field is set to **Disabled** then the user-level recording options will be respected.

The **Call recording format** field allows us to change the format of the call in which it will be stored. The default value is **wav**, but can be changed to **WAV** (WAV49—a GSM file stored as a **wav**), **ulaw**, **alaw**, **sln**, **gsm**, and **g729**. More information on recording formats can be found earlier in this chapter in the **Recording formats** section.

Recording calls

CompletePBX can be configured in order to record calls to a particular extension, queue, or conference. If a call encounters a request to initiate recording twice (for example, a caller enters a queue that is being recorded and the agent who picks up has set up their extension to record all of the calls), the call will only be recorded once. The call will be recorded by the target that first answered the call. In the previous example, the queue would record the call as the queue answered the call before the extension did.

If a call encounters conflicting recording instructions (for example, a caller enters a queue that is being recorded, but the agent who picks up has set up their extension to never record calls), the call will still be recorded. The target that has recording enabled will record the call. It is important to note that there is no way to stop a call from being recorded if it utilizes a call target that is set up to always record calls.

Recording calls to extensions

CompletePBX allows all of the calls to a particular user to be recorded, or for calls to be selectively recorded. CompletePBX allows the choice between recording only incoming calls, only outgoing calls, or both.

In order to set up call recording for a particular user, navigate to the **Connectivity** menu and select **Users** or **Extensions** dialog (this depends on the operational mode in which CompletePBX is running. More information on setting up users/extensions can be found in Chapter 2, **Devices and**

Extensions).

Click on the name of the user you wish to edit and scroll down to the **Recording Options** section:

The Record Incoming and Record Outgoing settings can be configured as Always, Don't Care, or Never.

When the **Don't Care** option is selected, the user can dial *1 during a call to start recording the call. *1 can be dialed again to stop recording the call. The on demand toggle only lasts for the current call. In order to record the next call, the user would have to dial *1 again.

If these options are set to **Always**, then all of the calls in the selected direction will be recorded. When **Always** is selected, pressing *1 will *not* stop call recording.

If these options are set to **Never**, then no calls in the selected direction will be recorded. When **Never** is selected, pressing *1 will *not* start recording.

Be sure to click on the **Submit** button, followed by the red-colored **Apply Config** bar at the top of the screen in order to save any changes made to the recording settings.
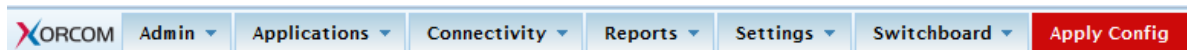
Recording calls to queues

All the calls to a particular queue can be recorded. This is often used on larger call queues for later call review by management, or to resolve disputes about what was said on a particular call. In order to record all of the calls that are picked up out of a particular queue, navigate to the **Applications** menu and select the **Queues** dialog.

Click on the name of the queue in the menu on the right, which needs call recording enabled:

Under the **General Queue Options** section, the **Call Recording** drop-down menu can be changed from **No** to **wav49**, **wav**, or **gsm**.

General Queue Options

Ring Strategy: ⍰ `ringall ▼`

Autofill: ⍰ ☐

Skip Busy Agents: ⍰ `No ▼`

Queue Weight: ⍰ `0 ▼`

Music on Hold Class: ⍰ `inherit ▼`

Ringing Instead of MoH: ⍰ ☐

Join Announcement: ⍰ `None ▼`

Call Recording: ⍰ `No ▼`

```
wav49
wav       old Time ▼
gsm       ment ▼
No        No Adjustment ▼
```

Recording Mode: ⍰

Caller Volume Adjustment: ⍰

Agent Volume Adjustment: ⍰

Mark calls answered elsewhere: ⍰ ☐

Select the desired recording format, click on the **Submit Changes** button, and click on the orange-colored **Apply Config** bar in order to enable recording on the queue.

Note that queues tend to have a high volume of calls. It is important to select an appropriate call recording format, and to ensure that recordings are maintained and cleaned up periodically. More information on recording formats can be found earlier in the **Recording** formats section of this chapter. Recording maintenance is discussed later in this chapter in the *Maintaining call recordings* section.

Recording calls to conferences

All of the calls to a particular conference room can be recorded. All of the members of the conference will be recorded and merged into a single file. In order to enable call recordings for a particular conference, navigate to the **Applications** menu and select the **Conferences** dialog.

Select the conference room that should have recording enabled from the menu on the right. Under **Conference Options**, the **Record Conference** drop-down menu will turn recording on or off:

Selecting **Yes** will record the entire conference (from the time the first member joins) in WAV format (more information on the WAV format can be found earlier in this chapter in the Recording formats section). Selecting a different format to record the conference in is not currently supported. Once **Yes** has been selected from the **Record Conference** drop-down menu, click on the **Submit Changes** button followed by the red-colored **Apply Config** button in order to enable call recordings for the conference.

`XORCOM　Admin ▼　Applications ▼　Connectivity ▼　Reports ▼　Settings ▼　Switchboard ▼　Apply Config`
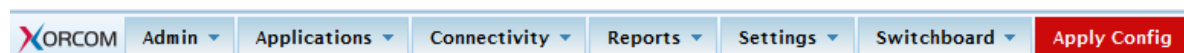
Maintaining call recordings

It is worth noting that call recordings can be quite large. If left unattended, a PBX that automatically records all of the calls will eventually fill up the entire available hard disk space and stop processing calls. It is important to have a maintenance strategy for dealing with call recordings in order to avoid this. While third-party tools exist to maintain recordings, adequate maintenance can often be performed with simple shell scripts.

The most common way of maintaining call recordings is to automate the deletion of recordings that are older than a certain time frame. The following script called *OldRecordingDeletion.sh* will remove all of the recordings older than 14 days.

```
#!/bin/bash

# Change this path to reflect your recording storage location
RECORDINGS=/var/spool/asterisk/monitor

# Change this number to reflect the maximum age of call
# recordings
RECORDINGEXPIRY=14

# Change this number to reflect the maximum age of the
# deletion logs
LOGEXPIRY=365

# Current date
DATE=`date`

# Delete recordings older than $EXPIRY days
find $RECORDINGS -mtime +$EXPIRY -exec rm -rfv > removal-$DATE.log\

# Delete log files older than $LOGEXPRY
find . -mtime +$LOGEXPIRY -exec rm -rf\
```

It is best to run a script like this once a day. Adding the following line to cron will execute the script from the */etc/recordingdeletion* file once daily at 5:00 a.m.:

```
0 5 * * * /etc/recordingdeletion/OldRecordingDeletion.sh
```

More information on using the cron system can be found later in this chapter in the *Using cron* section.

Another possible method of maintaining recordings is to find the longest calls, and only keep them (since the longer calls are often the calls that prove problematic). The script below (*SmallRecordingDeletion.sh*) will delete all of the recordings under 15 MB.

Script Example

The following script is written to maintain recordings of a single format, and defaults to removing recordings that are less than 15 megabytes in size. This is equivalent to about 15 minutes of calls that are recorded in WAV, ULAW, ALAW, or SLN format. For WAV49 or GSM formats, recordings in a 15 megabyte file would contain over two hours of call time.

For these formats, the equivalent value for 15 minutes of talk time is about 1.5 megabytes. If recordings are in multiple formats (for example, queues are recorded in GSM format while conferences are in WAV format), the script would need to be modified to only remove a specific format.

```
#!/bin/bash

# Change this path to reflect your recording storage
# location
RECORDINGS=/var/spool/asterisk/monitor

# Change this number to reflect the minimum size of
# recordings
RECORDINGSIZE=15

# Change this number to reflect the maximum age of the
# deletion logs
LOGEXPIRY=365
```

```
# Current date
DATE=`date`

# Delete recordings older than $EXPIRY days
find $RECORDINGS  -size -$RECORDINGSIZE M -exec rm -rfv > removal-$DATE.log\

# Delete log files older than $LOGEXPRY
find . -mtime +$LOGEXPIRY -exec rm -rf\
```

This script should also be run once per day. Adding the following line to cron will execute the script from */etc/recordingdeletion* once daily at 5:00 a.m.:

    0 5 * * * /etc/recordingdeletion/SmallRecordingDeletion.sh

Using cron

The cron system allows scheduled executions of any command that can be run from the server's command line. The easiest way to add an entry to cron is to edit the *crontab* file using the following command:

    crontab -e

If *crontab* has not previously been used, then a blank text file will be opened for editing. If the file has previously been edited, then all of the existing entries will be shown in the text editor.

In order to add a new entry to the file, simply place your cursor at the end of the file (below all other text) and start typing. A crontab entry should use the following syntax: *(min) (hour) (day) (month) (dayofweek) (command)*. Each token can be replaced as follows:

- *(min)* can be replaced with the minute when the command should be run. For example, if the command is to be run at 2:30 a.m., enter *30* for the minute. Valid values are any whole numbers between zero and 59.
- *(hour)* should be replaced with the hour when the command should be run. For the previous example, running a command at 2:30 a.m. would replace *(hour)* with *2*. Valid values are any whole numbers between zero and 23.
- *(day)* should be replaced with the day of the month that the command should be run on. For example, to run a command on July 20th, enter *20*. Valid values are whole numbers between one and 31. Note that the system will not generate an error if the entered date does not exist (for example, entering a cron job to run on September 31st will not generate an error). If an invalid date is entered, the command will never run.
- *(month)* should be replaced with the number corresponding to the month that the command is to be run in. For example, to run the command on July 20 enter *7* (because July is the seventh month). Valid values are whole numbers between one and 12.
- *(dayofweek)* should be replaced with the number corresponding to the day of the week that the command should be run on. For example, to run a command on a Thursday, enter *4* (because counting from Sunday as day zero, Thursday is the fourth day of the week). Valid values are whole numbers between zero and seven. Sunday can either be zero or seven, cron will recognize both.
- *(command)* should be replaced with the shell command that is being run. As cron does not run under a particular shell, it has no environment variables and does not know the path to various system executables. The full path to a command must be entered. For example, in order to reboot the system each time the cron job is executed, the command would be */sbin/reboot now*.

For any of the timing values (minute, hour, day, month, or day of week), an asterisk (*) can be used to represent all of the possible values. For example, the following cron entry would reload CompletePBX's configuration on the fifteenth minute of every hour:

15 * * * * /usr/sbin/asterisk -rx "reload"

A range of values can also be specified in order to execute a command within a specific time frame. For example, the cron entry below would reload CompletePBX's configuration once on the fifteenth minute of every hour, but only on business days:

15 * * * 1-5 /usr/sbin/asterisk -rx "reload"

It is also possible to specify a set of values using a comma (,). For example, the cron entry below would reload CompletePBX's configuration once per hour on the fifteenth minute of the hour, only during business hours (excluding lunch hour), and only on business days:

15 9,10,11,13,14,15,16,17 * * 1-5 /usr/sbin/asterisk -rx "reload"

Finally, the forward slash (/) can be used to specify what is known as "step values". Step values allow a command to be run a fraction of the amount of times it would be normally within a given time frame. For example, specifying *9-17/2* for the hour field would execute the command every two hours between the hours of 9:00 a.m. and 5:00 p.m. instead of once per hour during that time frame. The following cron entry would reboot the server on the first day of every third month at 2:00 a.m.:

0 2 * */3 * /sbin/reboot now

Once the entry has been completed, save the file and close the text editor (the method to accomplish this is different based on which editor is being used). A message will be displayed indicating that the *cron* file has been updated with a new version:

crontab: installing new crontab

The schedule for cron entries can be changed any time by reopening the *crontab* file for editing.

Summary

By now, we have learned how to record any call that enters our PBX, if we choose to do so. We now know of the importance of maintaining our recordings so that they do not fill up the hard disk on our PBX, and we are able to create some basic scripts to accomplish recording maintenance (with a little help from cron).

The next chapter will focus on personalizing our PBX by changing the default music on hold, recording custom voice prompts, and customizing feature codes/star codes.

8. Personalizing Your PBX

After all that hard work, we finally have a working PBX. It's time to make the PBX our own. CompletePBX allows for several options to customize the way our call structures sound. In this chapter, we will discuss how to:

- Upload our own custom music that will be played when callers are placed on hold
- Record custom voice prompts to answer incoming calls automatically
- Customize star codes that users will dial to activate various features
- Set up automated callback systems
- Configure Direct Inward System Access (DISA) to give remote agents a dial tone
- Configure customer caller ID lookup sources to properly identify incoming callers
- Configure PIN sets for increased security
- Configure custom applications
- Configure custom destinations

Custom Music on Hold

CompletePBX allows two styles of music-on-hold customization—static files and streaming. Static files are audio files (such as WAV or MP3 files) that are uploaded to the CompletePBX server and played back when a caller is placed on hold. Streaming audio is used to connect to a live audio feed from a particular source. Typically, this would be an Internet stream (many radio stations broadcast over the Internet), but it could also be a stream from a sound card or some other audio device.

CompletePBX separates different groups of music on hold into categories. Different static file categories can contain different sets of files. Each streaming category can only contain one audio stream. Music-on-hold categories can be applied to inbound routes (so that all calls that match the route will hear that category) as well as to queues and ring groups. Categories assigned at the call target level will override the music-on-hold category for that particular target only. Once a call leaves that target it will fall back into the music-on-hold category specified in the inbound route, which it was matched against.

Using audio files for Music on Hold

In order to create a music-on-hold category using static audio files, navigate to the **Applications** menu and select the **Music On Hold** dialog. Click on the **Add Music Category** link in the menu on the right:

Add Music Category
Add Streaming Category
default
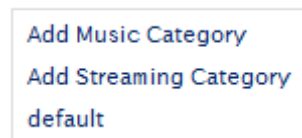
The initial setup screen for a static file category has one only field as shown in the following screenshot:

On Hold Music

Add Music Category

Category Name: ❓ [   ]

Submit Changes

Type a name for the category in the textbox and click on the **Submit Changes** button. On the resulting screen, click on the **Browse** button to find an MP3 or WAV file for upload, and then click on the **Upload** button. The quality of the files is not terribly important, as CompletePBX will downsample everything it gets to mono 8 kHz, 8 bits-per-sample, ULAW format.

On Hold Music

Category: default

Upload a .wav or .mp3 file:
[   ] [Browse…] [**Upload**]
[Volume 100% ▼] Volume Adjustment ❓

[**Enable Random Play**]

After clicking on the **Upload** button, CompletePBX will display a warning as shown in the following screenshot:

Please wait until the page loads. Your file is being processed.

[OK]

Warning!

After you click on the OK button, the page will still be loading. With larger files or slower connections, it can take several minutes for the page to load. Navigating to a new page before the page has finished loading will result in an incomplete upload and a corrupted music-on-hold file that will not function correctly.

Once the page has finished loading, the music file that was uploaded will be present as shown in the following screenshot:

On Hold Music

Category: default

Upload a .wav or .mp3 file:

[   ] [Browse…] [**Upload**]

Volume 100% ▾ Volume Adjustment ❓

[**Enable Random Play**]

Completed processing Kalimba.mp3!

| **wav_Kalimba.wav** | ⊖ |

Click on the **Submit Changes** button followed by the red-colored **Apply Config** button in order to save your configuration:

XORCOM   Admin ▾   Applications ▾   Connectivity ▾   Reports ▾   Settings ▾   Switchboard ▾   **Apply Config**
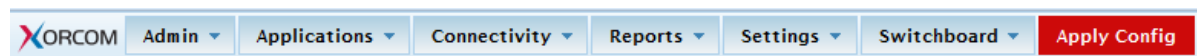
The music category is now ready for use (although it will only have one music file in it that will be played over and over again). Repeat the upload process for as many additional files as you would want to add. Music files will be played in the order in which they are uploaded unless random play is enabled. By clicking on the **Enable Random Play** button, the songs are played to the caller in a random order.

Using audio streams for Music on Hold

Using an audio stream for music on hold is a bit more complicated than using static audio files, and the results can be somewhat unpredictable depending on the source and type of the stream. Nonetheless, it is definitely possible to set up a live stream as a music-on-hold category. The only restriction CompletePBX imposes on the music stream is that it must be a mono 8 kHz, 8 bits-per-sample, ULAW audio stream. Most streaming audio programs support converting their streams to a new output format on the fly.

A streaming music-on-hold class can negatively affect the performance of a PBX due to bandwidth usage or transcoding issues. Each streaming class should be streamed only once at any given time (if multiple callers are placed on hold, all of them will be connected to the same stream). However, if multiple streaming classes are created, it is possible for multiple streams to be active at the same time. Each active stream can take up a significant amount of bandwidth (standard online broadcasts usually run at the speed of 96 kbps or 128 kbps). It is important to take bandwidth usage into account when planning a system. Five different streaming classes active at the same time could take up one-third of the available downstream bandwidth of a T1 line.

In addition to bandwidth issues, a streaming music-on-hold class may require transcoding to be played to a caller. CompletePBX requires that the a stream be in ULAW format, so calls already using the G.711 ULAW codec will be able to listen to the stream without any transcoding. A call that is using a compressed codec such as G.729 would force the stream to be transcoded for the caller to hear it. Not only does this place additional processing load on CompletePBX, but it also requires an additional G.729 license for the concurrent caller that the stream has to be transcoded for. If there are few processor resources available or the available codec licensing will be a problem, it may be beneficial to avoid streaming music on hold all together.

In order to create a streaming music-on-hold category, navigate to the **Applications** menu and select the **Music On Hold** dialog. Click on the **Add Streaming Category** link in the menu on the right:

Add Music Category
Add Streaming Category
default

The streaming category setup screen has three fields as shown in the following screenshot:

On Hold Music

Add Streaming Category

Category Name: ❓
Application: ❓
Optional Format: ❓

Submit Changes

The **Category Name** is a name for the streaming class, which will be used as a reference to the stream when selecting it wherever it is used in your call trees.

The **Application** field is likely to be the most complicated one that CompletePBX asks for. The value in this field is the program or script that CompletePBX will invoke in order to start "listening" to the stream. For RAW TCP streams, the built-in streamplayer application can be used in the format of */usr/sbin/streamplayer server port*. However, this will only work for RAW TCP streams, and will not play icecast or shoutcast streams.

In order to play icecast or shoutcast streams, a shell script is typically written, which is invoked by CompletePBX. The shell script will use a combination of various audio programs to start the stream, convert it to an acceptable format, and then feed it to CompletePBX. For example, the following script called *streamicecast.sh* uses *ogg123* to start a stream from *icecastserver.net* and uses *sox* to convert the stream to the required format:

> #!/bin/bash
>
> # /usr/bin/streamicecast.sh
>
> # Streams icecast feed from icecastserver.net for use with CompletePBX
>
> /usr/bin/ogg123 -q -b 128 -p 32 -d wav -f - http://icecastserver.net/| sox -r 16000 -t wav - -r 8000 -c 1 -t raw - vol 0.25

Then, we would just simply have to put */usr/bin/streamicecast.sh* in the **Application** field.

The **Optional Format** field is used to tell CompletePBX the format of the stream. This must be a format that CompletePBX understands (such as ULAW or GSM). For the most part, it is safest to leave this field blank and force the stream to the proper ULAW format.

Click on the **Submit Changes** button followed by the red-colored **Apply Config** button in order to save your configuration:

XORCOM   Admin ▼   Applications ▼   Connectivity ▼   Reports ▼   Settings ▼   Switchboard ▼   **Apply Config**
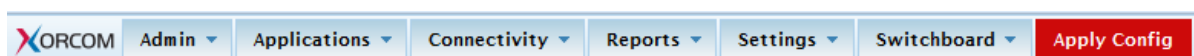
Custom voice prompts

Voice prompts are played to a caller when the caller enters an IVR, a follow-me, a ring group, or

anywhere else they might be prompted for input. The purpose of a prompt is to instruct the user what to do. This is usually something as simple as, "Thank you for calling Packt Publishing. Press *1* for sales, *2* for publishing, or an extension at any time."

Recording custom voice prompts

CompletePBX provides a simple method for recording new voice prompts right from a telephone handset.

To record a new script, navigate to the **Admin** menu and select the **System Recordings** dialog, and click on the **Add Recording** link in the menu on the right.

The **Add Recording** screen provides two methods to add system recordings to CompletePBX:

- Uploading existing audio files
- Recording new audio files via a VoIP endpoint

System Recordings

Add Recording

Add Recording
Built-in Recordings
followme_announcement
StandardBusinessHoursGreeting

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, (eg, recorded with Microsoft Recorder) the file **must** be PCM Encoded, 16 Bits, at 8000Hz:

[   ] [Browse…]   [Upload]

Step 2: Name

Name this Recording: [   ]

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

To record a new prompt, enter the extension of the endpoint from which you will be recording into the extension number box and click on the **Go** button. The screen will change to reflect that an extension has been entered:

Uploading an existing recording is done in the same way in which a static music file is uploaded to a music-on-hold category (click on the **Browse** button, select the desired file, and click on the **Upload** button).

In order to record a new prompt, dial *77* on the extension that was entered and begin speaking after the beep. Hang up when recording is complete. To review the recording, dial *99* from the extension that was entered. Repeat this process until you are satisfied with the recording.

Enter a name for the recording in the **Name this Recording** box. Note that the name can only contain letters and numbers, not spaces and special characters (for example, the characters that may not be used include ", -, +, =, *, &, ^, %, $, #, @, !).

Click on the **Save** button when you have finished recording. The recording will show up in the menu on the right.

Merging existing voice prompts

CompletePBX also allows existing voice prompts to be merged together. A merged prompt will be played as a single, continuous file. To merge prompts, navigate to the **Admin** menu and select the **System Recordings** dialog, and click on the **Built-in Recordings** link in the menu on the right.

The **Built-in Recordings** screen has a single drop-down menu. Select the recording you wish to start with as the base recording from this menu, and click on the **Go** button as shown in the following screenshot:

System Recordings

Built-in Recordings

Select System Recording:

.asterisk-core-sounds-en-gsm-1.4.21   ▼   Go

The **Edit Recording** screen will be shown as follows:

System Recordings

Edit Recording

◀️ Remove Recording *(Note, does not delete file from computer)*

Change Name ❓   .asterisk-core-sounds-e

Descriptive Name ❓   No long description available

Direct Access Feature Code Not Available ❓ :

Files:

.asterisk-core-sounds-en-gsm-1.4.21   ▼   🔊   ↴   🗑

                                       ▼   🔊 ↵   🗑

Save

The **Change Name** changes the name of the recording. The name should be changed to distinguish the merged recording with the original recording.

The value of the **Descriptive Name** field is displayed as a hint to what the recording contains when it is being selected in places such as queues and IVRs.

The **Files** drop-down menus are where individual recordings are selected to be merged. Select a recording from the second drop-down menu and click on the **Save** button to add it to the merged recording. This can be done many times to include several recordings as shown in the following screenshot:

System Recordings

Edit Recording

Remove Recording *(Note, does not delete file from computer)*

Change Name ❓   `.asterisk-core-sounds-e`

Descriptive Name ❓   `No long description available`

Direct Access Feature Code Not Available ❓ :

Files:
`.asterisk-core-sounds-en-gsm-1.4.21`

`.asterisk-extra-sounds-en-gsm-1.4.9`

Save

Clicking on the arrow buttons will move a recording up or down, changing the order in which it will be played; clicking on the garbage buttons will remove a recording from the merged recording.

You must make sure to click on the **Save** button after making any of the changes.

Customizing feature codes

Feature codes are "extensions" that users can dial to enable or disable certain features for themselves. Feature codes are typically prefixed with an CompletePBX (*) sign followed by two numbers. The feature codes on a CompletePBX system can be altered to use a different extension or disabled.

To customize feature codes, navigate to the **Admin** menu and select the **Feature Codes** dialog:

All the available features codes will be displayed, ordered by sections, as shown in the following screenshot:

Feature Code Admin

		Use Default?	Feature Status
Blacklist			
Blacklist a number	*30	☑	Enabled ▾
Blacklist the last caller	*32	☑	Enabled ▾
Remove a number from the blacklist	*31	☑	Enabled ▾
Call Forward			
Call Forward All Activate	*72	☑	Enabled ▾
Call Forward All Deactivate	*73	☑	Enabled ▾
Call Forward All Prompting Deactivate	*74	☑	Enabled ▾
Call Forward Busy Activate	*90	☑	Enabled ▾
Call Forward Busy Deactivate	*91	☑	Enabled ▾
Call Forward Busy Prompting Deactivate	*92	☑	Enabled ▾
Call Forward No Answer/Unavailable Activate	*52	☑	Enabled ▾
Call Forward No Answer/Unavailable Deactivate	*53	☑	Enabled ▾
Call Forward Toggle	*740	☑	Enabled ▾
Call Waiting			
Call Waiting - Activate	*70	☑	Enabled ▾
Call Waiting - Deactivate	*71	☑	Enabled ▾

To change the "extension" that a feature code uses, uncheck the **Use Default?** and type in the new "extension" that should be used as shown in the following screenshot:

Blacklist

Blacklist a number	*29	☐	Enabled ▾
Blacklist the last caller	*32	☑	Enabled ▾
Remove a number from the blacklist	*31	☑	Enabled ▾

To disable a feature code entirely, change the drop-down value from **Enabled** to **Disabled** as shown in the following screenshot:

Blacklist

Blacklist a number	*29	☐	Disabled ▾
Blacklist the last caller	*32	☑	Enabled ▾
Remove a number from the blacklist	*31	☑	Enabled ▾

Click on the **Submit Changes** button followed by the red-colored **Apply Config** button in order to save your configuration:

XORCOM   Admin ▾   Applications ▾   Connectivity ▾   Reports ▾   Settings ▾   Switchboard ▾   **Apply Config**
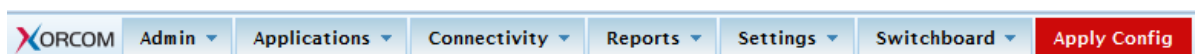
Callback

A **callback** is a call target that will immediately hang up on a caller, call them back, and then redirect the call to another call target. This is most often used to avoid long-distance charges for remote agents who do not have access to a VoIP endpoint. This is especially relevant in the case of mobile phones where incoming calls are usually significantly cheaper than outgoing calls. The callback target may connect the caller with any resource on CompletePBX (such as an extension, the voicemail messaging center, or a queue), or it may be used in conjunction with DISA to give the caller a dial tone on the system from which they can call any telephone number they wish (more information on DISA is available later in this chapter in the *Direct Inward System Access (DISA)* section).

In order to set up a callback target, navigate to the **Applications** menu and select the **Callback** dialog.

Callback targets have only four configuration options:

- Callback Description
- Callback Number
- Delay Before Callback
- Destination after Callback

The **Callback Description** field is used to set a name to refer to this callback configuration whenever call targets are being selected in our call trees.

The **Callback Number** field is the telephone number that CompletePBX will dial to reconnect with the caller after the call that initiated the callback is terminated. The number must be in a format that one of the outbound routes configured in the **Outbound Routes** section of CompletePBX can be matched with (for example, if there is no outbound route defined to match a 10-digit dialing pattern, entering *5551234567* for this field would render the callback configuration useless, as the outbound callback would never be completed). If the field is left blank, then CompletePBX will attempt to call back the caller ID number that initiated the callback.

Test thoroughly

Test the callback module thoroughly if the Callback Number field is left blank.

Many mobile phones have their caller ID show up as "unknown" or "unavailable" when calls are placed to landline telephone services. If a caller with an unknown caller ID number initiates a callback, the callback will silently fail without any

notification to the caller.

If there is no way for a caller to send caller ID information, then a callback target will only be useful to that person if it calls them back at a specific predefined telephone number.

Entering a number into the **Delay Before Callback** field will delay the start of the callback process for the number of seconds specified in the field once a callback is initiated. This field is optional and may be left blank if no delay is desired. Certain phone services can take several seconds to be available again after a call is terminated. It is often useful to configure a delay of a few seconds in order to mitigate the risk of the callback module receiving a busy signal while the original caller's service resets itself to accept incoming calls again.

The **Destination after Callback** field is used to configure the call target that the caller will be connected to, once the callback module reconnects the caller to the PBX. Any existing call target is available.

Once all options have been configured, click on the **Submit Changes** button followed by the red-colored **Apply Config** bar to save the callback target and make it active.
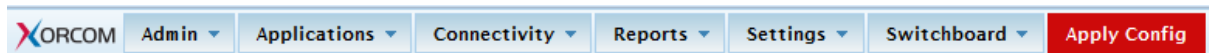
A few common examples of when a callback target might be used are as follows:

A company where employees need the ability to check their voicemail from anywhere. Calling the toll-free company phone number costs the company too much money. A callback target could be set up to call back the incoming caller ID, and be directed to the miscellaneous destination of *98 (more information on **Misc Destinations** can be found later in this chapter in the *Misc Destinations* section). Callers would receive a call on the number they called from, would be prompted for their extension and their password, and would then have access to their voicemail messages.

A company receives better per-minute rates on calls made through its VoIP trunks than calls made through employees' mobile phones. Employees' mobile phones have free incoming calls. A callback target could be set up for each employee with a mobile phone to call back the employee's mobile number. The callback would be directed to a DISA destination to give the employee a dial tone on the PBX (allowing them to dial out using the company's VoIP trunks without using any outgoing mobile minutes). More information on DISA is available later in this chapter in the *Direct Inward System Access (DISA)* section.

A company that receives collect calls from anywhere in the world (such as a credit card company that needs to receive calls if a customer's card is lost or stolen). The company reduces their costs if they use a VoIP trunk local to the country that the customer is in, rather than paying for the entire collect call at hefty international rates. A callback target could be set up to call back the incoming caller ID of the customer and be directed to a queue. The customer would receive a call to the number they called from and would be connected with a company representative as soon as one is available.

Direct Inward System Access (DISA)

A DISA call target will provide a caller with a dial tone on CompletePBX. Once the caller has a dial tone, they can utilize the same set of functions that are utilized by a user with VoIP endpoint attached to CompletePBX. This means that a person who is remotely located could be given access to dial any extension directly, check their voicemail messages, or even place calls to external telephone numbers through CompletePBX.

In order to set up a DISA call target, navigate to the **Applications** menu and select the **DISA** dialog.

DISA call targets have following eight configuration options:

- DISA name
- PIN
- Response Timeout
- Digit Timeout
- Require Confirmation
- Caller ID
- Context
- Allow Hangup

Add DISA   Add DISA

DISA is used to allow people from the outside world to call into your PBX and then be able to dial out of the PBX so it appears that their call is coming from the office which can be handy when traveling. You can set a destination in an IVR that points to the DISA or set a DID. Make sure you password protect this to keep people from dialing in and using your PBX to make calls out.

Add DISA

DISA name: ❓
PIN❓
Response Timeout❓   10
Digit Timeout❓   5
Require Confirmation❓   ☐
Caller ID❓
Context❓   from-internal
Allow Hangup❓   ☐

Submit Changes

The **DISA name** field is used to identify the DISA configuration when it is being selected as a call target in other parts of the CompletePBX interface.

PIN is used to authenticate a caller when they reach the DISA call target. If the PIN field is not left blank, then the caller will be prompted to enter their authentication code. The PIN that the user enters must match with the value of the PIN field, otherwise the call will be disconnected and the caller will not be able to access the DISA call target. Multiple PIN values can be entered by separating valid PIN values by commas (for example, entering 1234, 5678, 9012 would authenticate callers if they entered any one of the numbers—1234, 5678, 9012 as their PIN when prompted).

The **Response Timeout** field is used to specify how long CompletePBX will wait for valid input before disconnecting the call. This not only applies when a caller has not entered any digit yet, but also if a caller has partially entered a number to call without finishing the entry. The default value for this field is 10 seconds, but this is often too short and a caller may feel rushed with this timeout. A setting of 15 or 20 seconds is usually sufficient to allow callers time to enter their desired telephone number without wasting too much time on a dropped call.

The **Digit Timeout** field is used to specify how long CompletePBX will wait between digits before dialing the call. If a caller begins entering digits and then stops, CompletePBX will wait for however many ever seconds specified in this field, before sending the entered digits to CompletePBX for dialing. The default value for this field is five seconds. This is usually sufficient as most people do not take more than five seconds between button pushes on their phone once they have started dialing.

If the **Require Confirmation** option is checked, then CompletePBX will prompt the caller to press 1, in order to initiate the DISA process as soon as they are transferred to the DISA call target. If the user

fails to press 1, then the call is disconnected. This is useful when calls are transferred directly to a DISA call target through an inbound route that matches the caller ID. It is common for the call to be set up and answered so quickly that the caller will not hear any ring or confirmation before being placed into the DISA call target, and the caller will then time out and disconnect before they realize what happened. Requiring confirmation alerts the user that they are about to enter the DISA call target and allows them to press *1* when they are ready.

The **Caller ID** field is used to set the outbound caller ID of any of the calls that are placed from within the DISA call target. The desired caller ID should be specified in the format of *"Caller Name" <##########>*—*Caller Name* is replaced with the name that should be set on outbound calls and *##########* should be replaced with the phone number that should be set on outbound calls (for example, *"Packt Publishing" <5551234567>*). This is an optional field. If this field is left blank, then the caller ID of the person placing the call will be used.

> Setting a caller ID value in the Caller ID field does not guarantee that the specified value will be used. If the trunk that routes the call has the Never Override CallerID option enabled, then the caller ID will be whatever is specified in the trunk configuration. More information on trunk caller ID options can be found in the Setting up a New Trunk section of Chapter 3, **Trunks.**

The **Context** field is used to specify the context in which CompletePBX will place the caller when they enter the DISA call target. The context that a user is in defines the features and trunks they have access to. CompletePBX places all users into the *from-internal* context by default. As DISA is meant to mimic the functions of an internal user, CompletePBX sets the **Context** field to *from- internal* by default as well. Changing this field will require an understanding of the CompletePBX configuration files, and will likely mean that a custom context has been manually created in the */etc/asterisk/extensions_custom.conf* file. Setting a different context in this field could render the DISA call target useless, hence be sure that the context entered here is valid.

If the **Allow Hangup** checkbox is selected, then the caller can press the hangup feature code (by default this code is **) to end their current call and be presented with a dial tone again. This prevents the caller from actually hanging up their phone, calling back into CompletePBX, and re-authenticating to gain DISA again. If desired, the feature code for hanging up a call can be changed. More information on how to change the feature codes can be found in the *Customizing Feature Codes* section of this chapter.

Once all options have been configured, click on the **Submit Changes** button followed by the red-colored **Apply Config** bar to save the DISA call target and make it active.
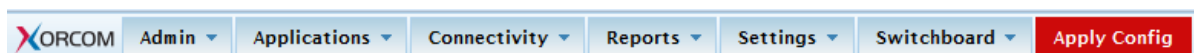
PIN Sets

A **Personal Identification Number** (**PIN**) is a numeric password that CompletePBX can use to authenticate users when they attempt to dial specific features on the PBX such as outbound routes or DISA. Without entering a valid PIN, these features are inaccessible. A PIN set provides the ability for a group of PINs to be used to authenticate the use of a feature instead of just a single PIN. With a PIN set, each individual user can be given a unique PIN, and it is possible to revoke one user's access to a PBX feature without changing anyone else's PIN.

In addition to authenticating users, PIN sets can also be helpful in tracking which users used which features. When a PIN set is used as an authentication mechanism, CompletePBX can log which

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unique PIN was used to access a feature in the *accountcode* column of the call detail records. Even if a user places a call from someone else's extension, they will need to enter their own account code. When generating a report of which users used certain features, the detail records will provide an accurate report based on which PINs were used. This feature is often used by companies which implement restrictions on long-distance calling. It is possible to track which users make heavy use of the long-distance outbound routes by checking the PINs used to dial out.

In order to set up a PIN set, navigate to the Applications menu and select the **PIN Sets** dialog.

PIN sets have only three configuration options:

- PIN Set Description
- Record In CDR?
- PIN List

Add PIN Set

Add Password Set

PIN Sets are used to manage lists of PINs that can be used to access restricted features such as Outbound Routes. The PIN can also be added to the CDR record's 'accountcode' field.

New PIN Set

PIN Set Description: |
Record In CDR?: ☐

PIN List:

PIN Set Description is used to identify this PIN set when it is being selected during the configuration of an outbound route.

If the **Record In CDR?** checkbox is selected, then anytime a user authenticates themselves using this PIN set, the PIN they used will be stored in the *accountcode* field of the call detail records alongside the details of the call they made. This is a reliable method of tracking the usage of outbound routes, because even if a user calls from an extension other than their own, they still must enter their own PIN to place a call through a protected route. If this option is not enabled then the call is logged normally without any details in the *accountcode* field.

The **PIN List** is a list of one or more PIN codes. Each PIN should be on its own line. PINs can be as long as desired. However, as users are supposed to remember their PINs, it is typical to make PINs only four or five digits long.

A completed PIN set might look like the following:

Add PIN Set

Add Password Set

PIN Sets are used to manage lists of PINs that can be used to access restricted features such as Outbound Routes. The PIN can also be added to the CDR record's 'accountcode' field.

New PIN Set

PIN Set Description: Long Distance

Record In CDR?: ☐

PIN List:
```
63926
83592
64363
28942
```

In order to put our new PIN set to use we must assign it to an outbound route. Navigate to the **Connectivity** menu and select the **Outbound Routes** dialog.

Click on the name of outbound route that the new PIN set will be associated with (in this example, **International**).

From the **PIN Set** drop-down menu, select the name of the new PIN set:

Route Settings

Route Name:

Route CID: ☐ Override Extension

Route Password:

Route Type: ☐ Emergency ☐ Intra-Company

Music On Hold?   default ▼

Time Group: ---Permanent Route--- ▼

Route Position: Last after Emergency ▼

Additional Settings

Call Limit :   None ▼

PIN Set :   LongDistance ▼

+ Extension Routing :

Click on the **Submit Changes** button followed by the red-colored **Apply Config** bar. All calls that pass through the selected outbound route will now be authenticated through one of the PINs in our new PIN set.

Misc applications

A misc application is a custom feature code. A misc application allows a custom extension or star code to be defined, which will direct the caller to any call target when dialed. For example, if we have a ring group that calls the cell phones of all staff members, we might create a misc application that calls that ring group when *CELL* (*2355*) is dialed.

In order to create a misc application, navigate to the **Applications** menu and select the **Misc Applications** dialog.

Misc applications have four configuration options:

---

1. Description
2. Feature Code
3. Feature Status
4. Destination

Add Misc Application

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

Add Misc Application
Dynamic Conference
Security Alert

Add Misc Application

Description: ❓ [   ]
Feature Code: ❓ [   ]
Feature Status: ❓ [Enabled ▾]

Destination:

[== choose one == ▾]

[Submit Changes]

Description is used to identify this application if it needs to be edited or removed at a later time.

Feature Code is the custom feature code that users will dial in order to access this application. This can be a star code (for example, *1234*) or simply a normal extension (for example, 1234). The value of this field must be unique (a misc application cannot share a feature code with any other user, application, or star code on the system).

Feature Status determines whether or not the application is active. If this field is set to **Disabled**, then users will be informed that the extension they dialed is not valid if they attempt to use the application. This field allows an application to be quickly disabled without having to remove the application entirely.

Destination is the call target that the application should route the caller to. Any call target that has been previously configured is a valid destination for a misc application.

Once all options have been configured, click on the **Submit Changes** button at the bottom of the page followed by the red-colored **Apply Config** bar at the top of the page.

XORCOM   Admin ▾   Applications ▾   Connectivity ▾   Reports ▾   Settings ▾   Switchboard ▾   **Apply Config**
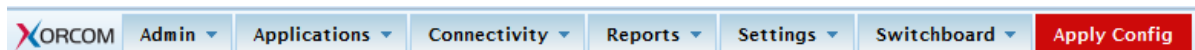
The application is now active and can be accessed by dialing the selected feature code from any extension on the PBX.

Misc Destinations

A misc destination is used to add a custom call target that can be used by CompletePBX modules. Anything that can be dialed from a user's extension can be turned into a misc destination. For example, by default, there is no way to send an inbound caller directly to the messaging center so that the caller could log in and check their voicemail messages. A misc destination could be set up to dial *98* and then an inbound route could point directly to that misc destination. A caller who was routed through that inbound route would immediately hear the prompts to log into their voicemail box, just as if they were a user on the PBX and had dialed *98.

To create a misc destination, navigate to the Applications menu and select the **Misc Destinations** dialog.

Misc destinations have two configuration options:

1. Description
2. Dial

Description is used to identify this destination when it is being selected as a call target in other modules.

Dial is the extension, telephone number, or feature code that the system should dial when a caller is routed to this destination. Anything that can be dialed from a user's extension can be entered into this field.

Once all options have been configured, click on the **Submit Changes** button at the bottom of the page followed by clicking on the red-colored **Apply Config** at the top of the page.
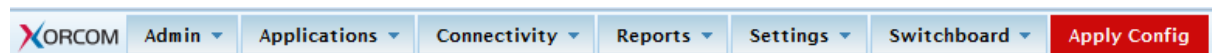
The destination we just created will now be available as a call target from any other CompletePBX module.

Faxing

Inbound Fax

Navigate to the **Connectivity** menu and select the **Inbound Routes** dialog. This will allow you to select the inbound route that will receive faxes

Navigate down to **Fax Detect** section, and set **Detect Faxes** to **True**. This will enable the selected inbound route to receive faxes

Define **Fax Detection Type** – typically DAHDI

Set the **Fax Detection Time** to 5 to enable CompletePBX to detect an incoming fax before the call is answered, and set **Fax Destination** to deliver the incoming fax to the required extension.

Now navigate to the **Connectivity** menu and select the **Extensions** dialog. You can now select extension that will receive faxes.

Navigate down to **Fax** section, and check on **Enabled**, and type the email address to which faxes should be delivered in the **Fax Email** field. Check on **Store Locally**, and set the remaining coversheet details as appropriate.

Outbound Fax

This module is standard in CompletePBX CXE and CTS models only.

Navigate to the **Settings** and select the **Fax Configuration** dialog. This dialog has three main sections

- Fax Presentation Options
- Fax Cover Page options
- Local Storage options

Presentation Options

Navigate down to the **Fax Presentation Options** section where you can configure information for all outbound faxes, such as:

- Default Fax header
- Local Station Identifier
- Outgoing email address

Cover Page

Navigate down to the **Fax Cover Page Options** where you can define cover page details such as:

- Company logo in jpg format
- Company details
- Name
- Address

- Phone
- Website
- Footer

When you have completed defining your cover page, you can click on the **Preview** button to see how the cover page will look.

Local Storage

Navigate down to **Local Storage Options** which allows you to define how many fax pages can be stored locally for each user, and to define what action to take when the storage is full.

Once all options have been configured, click on the **Submit Changes** button at the bottom of the

page followed by clicking on the red-colored **Apply Config** at the top of the page.
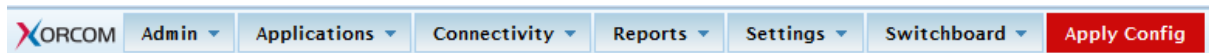
Summary

By now our CompletePBX should truly be our own. In this chapter, we've learnt how to:

- Upload custom on-hold music
- Create on-hold music from streaming Internet sources
- Set up automated callback systems
- Configure DISA
- Configure custom caller ID lookup sources
- Configure PIN sets
- Create custom applications
- Create custom destinations

Our CompletePBX is now entirely unique and customized to our specifications.

In the next chapter, we will discuss important topics such as system protection, backup and restoration, just in case disaster strikes.

9. Managing Call Flow

Configuring My Switchboard

Important note: if you want the Switchboard to track the status of queues and to create statistics for call center reporting, make sure that both **Event When Called** and **Member Status Event** (in the **Events, Stats and Advanced** section of the Queue dialog) are set to **Yes** when defining a new queue

Events, Stats and Advanced

Event When Called: ❓   Yes ▾

Member Status Event: ❓   Yes ▾

Note that the Switchboard on CXR series machines can display up to 15 buttons only.

Check on all the buttons that you want to hide, so that no more than 15 buttons remain unchecked

Permissions

This dialog allows you to create groups of permissions, which can be used either in templates or to define the permissions granted to a user

Navigate to the **Switchboard** menu and select the **Permissions** dialog.

This module will let you manage advanced FOP2 permissions.

Delete FOP2 Permission 'basic'

Edit Permission

Name: ❓　　　　　　　　　　　　　　　　basic

Permissions: ❓　　　　　　　　　　　　dial
　　　　　　　　　　　　　　　　　　　hangup

Permission Quick Pick ❓　　　　　　　(pick permission) ▼

Apply to Groups

Select the groups you want to apply this permission to. You will be able to perform the actions only to the extension/buttons that are part of the selected groups. If no groups are selected, the permission will be allowed to every button on the panel.

All Buttons　　　　　　　☐
All Extensions　　　　　　☐
All Queues　　　　　　　☐
All Conferences　　　　　☐
All Trunks　　　　　　　☐

Submit Changes

Name allows you to reference the permission group.

Permissions allow you to specify what actions can be performed by a user who is granted this permission group

Click on the **Permission Quick Pick** to see a list of all available permissions.

Apply to Groups enables you to select which groups should belong to this permission group. If no groups are selected the permission group will be granted to all buttons.

Once all options have been configured, click on the **Submit Changes** button at the bottom of the page followed by clicking on the red-colored **Apply Config** at the top of the page.

XORCOM　Admin ▼　Applications ▼　Connectivity ▼　Reports ▼　Settings ▼　Switchboard ▼　Apply Config
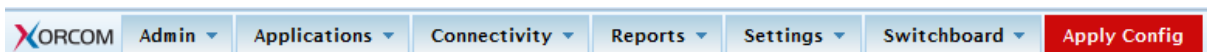
Templates

This dialog simplifies user management by allowing you create and manage templates, which you can then assign to users.

Navigate to the **Switchboard** menu and select the **Templates** dialog.

Add FOP2 Template

This module will let you manage FOP2 templates, that you can assign to users for easy management.

Add Template

Name: ❓

Permissions: ❓

Permission Quick Pick ❓   (pick permission) ▾
                                 (pick permission)
Make Default: ❓   all
                                 dial
                                 hangup
                                 hangupself
                                 meetme
Allowed Groups   pickup
                                 record
Select the groups you want to add to this template. If no gro   e user will be able to see all buttons.
                                 recordself
                                 spy
                                 transfer
                                 transferexternal
                                 whisper
All Buttons   queuemanager
All Extensions   queueagent
All Queues   phonebook
All Conferences   chat
All Trunks   preferences
                                 voicemailadmin

Submit Changes

Name allows you to reference the template.

Permissions allow you to specify what actions the user will be able to perform

Click on the **Permission Quick Pick** to see a list of all available permissions.

Check on **Make Default** if you want this template to become the default template, which will be applied to all users for which another template has not been specified.

Allowed Groups enables you to select which groups should be added to this template. If no groups are selected the user will be able to see all buttons. This gives you the ability to restrict the buttons that are displayed to users of this template.

Once all options have been configured, click on the **Submit Changes** button at the bottom of the page followed by clicking on the red-colored **Apply Config** at the top of the page.

XORCOM Admin ▾ Applications ▾ Connectivity ▾ Reports ▾ Settings ▾ Switchboard ▾ **Apply Config**
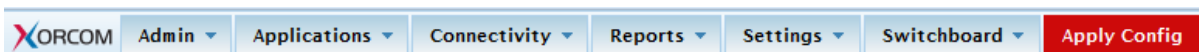
Users

This dialog enables you to manage Switchboard users and permissions, and also select which groups each user can view.

Navigate to the **Switchboard** menu and select the **Users** dialog

This module will let you manage FOP2 users and permissions, and also mark the groups the user will be able to view.

Delete FOP2 User '2002'

Edit User

Extension: ❓   2002

Secret: ❓

Template Quick Pick ❓   (pick template) ▾

Permissions

Permissions: ❓   all

Permission Quick Pick ❓   (pick permission) ▾

Allowed Groups

Select the groups you want the user to see in the panel. If no groups are selected the user will be able to see all buttons.

All Buttons   ☐
All Extensions   ☐
All Queues   ☐
All Conferences   ☐
All Trunks   ☐

Extension is the extension allocated to the user.

Secret is the password that will be used by the user to access the Switchboard. If no password is defined, the user's voicemail password will be used.

Template Quick Pick provides a dropdown list of previously defined templates that can be applied to the user.

If no template is used, you can grant the user a selected list of permission from the **Permission Quick Pick** dropdown list.

Allowed Groups enables you to select which groups should be available to the user. If no groups are selected, the user will be able to see all groups. This gives you the ability to restrict the groups of buttons that are displayed to the user.

Once all options have been configured, click on the **Submit Changes** button at the bottom of the page followed by clicking on the red-colored **Apply Config** at the top of the page.

XORCOM   Admin ▾   Applications ▾   Connectivity ▾   Reports ▾   Settings ▾   Switchboard ▾   **Apply Config**
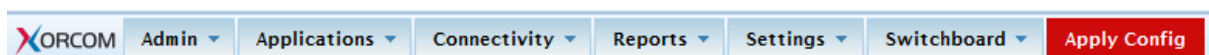
Buttons

The buttons dialog allows you to define the behavior of each button.

Navigate to the **Switchboard** menu and select the **Buttons** dialog

For example, you can change button labels (overriding the existing CompletePBX label) and privacy options (to prevent the button to be monitored and/or hide the CallerID) among other things. You can also sort the list by dragging the elements

> Note that the Switchboard on CXR series machines can display up to 15 buttons only.
>
> Check on all the buttons that you want to hide, so that no more than 15 buttons remain unchecked

- **Label** defines the name that will be displayed on the button in the Switchboard.
- **Privacy** allows you to disable monitoring or hide the CalledID for this button.
- **Email** allows you to send emails to the user from the Switchboard.
- **External Transfer** allows you to define the ***number@context*** that will be used for external transfers. The default context for the extension will be used if the context is omitted.
- **Channel** can used to add extra channels for the button.
- **Group** allows you to assign an extension to a group. The Switchboard will display all extensions belonging to the same group in a separate box.
- **Custom ASTDB** is the ASTDB key to search for custom states. By default, this set to Call Forward Unconditional state.
- **Originate Channel** allows you define which channel to use when originating a call from the Switchboard.
- **ChanSpy** Options are passed to the ChanSpy function.
- **Tags** can be defined for buttons, which can used to filter the buttons that will be displayed, as described in the **Filter box** section below.

Groups

Navigate to the **Switchboard** menu and select the **Groups** dialog:

This module will let you manage FOP2 groups.

Add Group

Name: ❓   [   ]

Included Buttons

Select the buttons you want to include in this group.

Extensions		Check All
1001	DAHDI channel 1	☐
1002	DAHDI channel 2	☐
1003	DAHDI channel 3	☐
1004	DAHDI channel 4	☐
1005	DAHDI channel 5	☐
1006	DAHDI channel 6	☐

Using My Switchboard

Queue Events

Important Note

If you want the Switchboard to track the status of queues and to create statistics for call center reporting, make sure that both **Event When Called** and **Member Status Event** (in the **Events, Stats and Advanced** section of the Queue dialog) are set to **Yes** when defining a new queue.

Important note: if you want the Switchboard to track the status of queues and to create statistics for call center reporting, make sure that both **Event When Called** and **Member Status Event** (in the **Events, Stats and Advanced** section of the Queue dialog) are set to **Yes** when defining a new queue

Events, Stats and Advanced
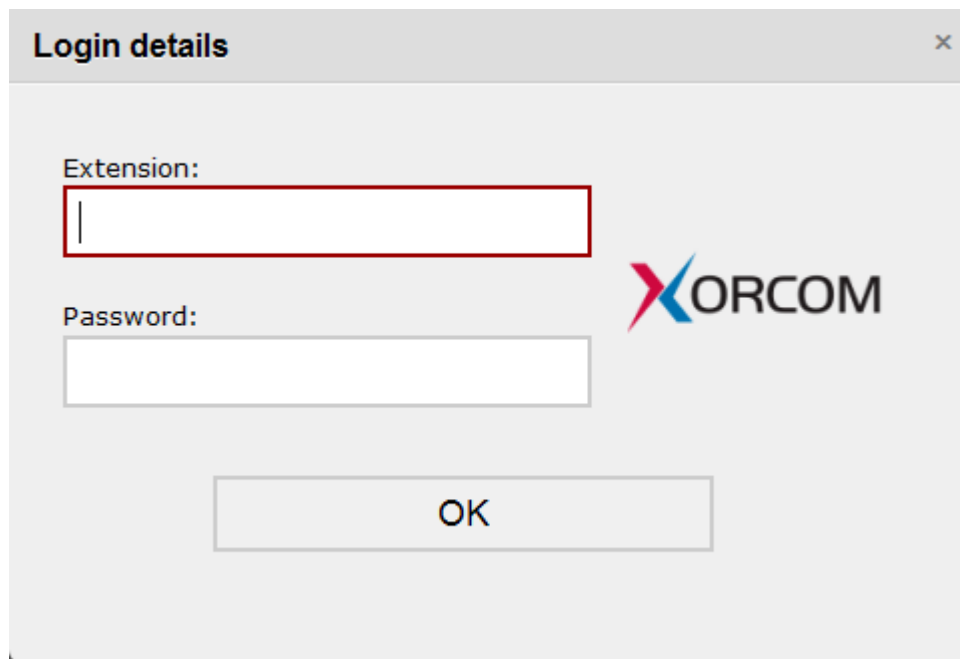
Event When Called: ❓   Yes ▾
Member Status Event: ❓   Yes ▾

Login

To reach the login dialog, navigate to the CompletePBX Launch Pad, click on the **My Switchboard** icon. Alternatively, you can open an Internet browser and enter the Web address for the panel in the address panel. The address depends on the IP address of your CompletePBX machine, but it could be something like **http://192.168.0.11/fop2**

If the Switchboard server is running, you will be presented with the login box.

Use your extension number, and voicemail password to login.

In order to perform actions with the panel, you must login with an extension that has been granted permissions, as that extension will be the originator of the actions you perform.

Passing credentials via URL

It is possible to pass the extension and password in a URL to avoid being prompted to login. You can use this feature to embed the panel into another application, or to create a bookmark. In order to do this, you must append extension and password parameters, as in this example:

> http://192.168.0.11/fop2/?exten=2002&pass=9999

Main Panel View

After a successful login, you will be presented with the panel window.

Note that the Switchboard on CXR series machines can display up to 15 buttons only.

Check on all the buttons that you want to hide, so that no more than 15 buttons remain unchecked

There is a toolbar that is always visible at the top, and four sections in the content area: one for each kind of button.

The button corresponding to your extension (the one you used to login), is always at the top left of the Extension block. Its label will be highlighted with a bold font.
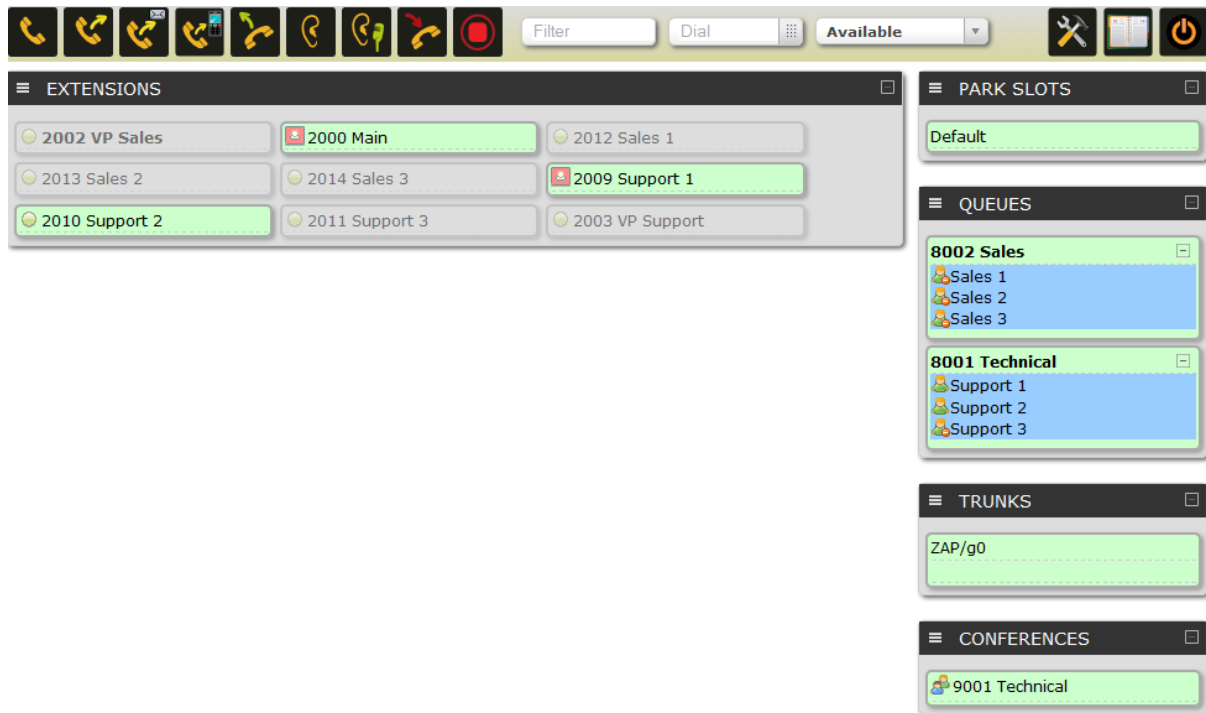
Each block has a title bar with a plus/minus button to the right. Clicking on the button will collapse or expand the view of the section. The collapse state is remembered across all future sessions. Suppose that you are not interested in trunks: you can collapse the section and it will remain minimized until you click the plus sign again.

Anatomy of the toolbar

At the top of the panel there is a toolbar. From here you will perform most of the actions that can be done with the Switchboard. There are some distinctive sections as follows:

Action bar

The action bar consists of a row of buttons that represent actions that can be taken for active calls. The buttons will be displayed only is the user has been given permission for the action. To activate an action, you need to first select the destination or target button with a click of the mouse. After you have done that, the borders of the target button will turn to red.

After selecting a target extension, you can perform your command by clicking on the appropriate action button. The following actions are available:

Button	Action
Dial	Required permission: **dial** Will originate a call to the button previously selected from the list of available extensions. Your phone will ring and after you pick it up, it will dial the requested destination extension

Button	Action
Transfer	Required permission: **transfer** Will initiate a call transfer to the button previously selected from the list of extensions
Transfer to voicemail	Required permission: **transfer** Will initiate a transfer to the voicemail of the selected extension
Pickup	Required permission: **pickup** Will pickup the ringing extension that was selected.
Listen	Required permission: **spy** Will start a call-spying session. It will first ring your phone, and after you pick it up, it will start spying on the previously selected extension.
Whisper	Required permission: **whisper** Will start a call-spying session with whisper enabled. It will first ring your phone and when you pick it up, it will start spying on the previously selected extension. You can then talk to the target extension without the other party hearing you.
Hangup	Required permission: **hangup** Will hangup the first active call on the selected extension.
Record	Required permission: **record** Will initiate or stop recording on the selected extension. A call that is being recorded will be marked with a cassette icon.

Filter box

Type text in this input box to filter out any button having a label that does not match your filter. If you have a panel with a hundred buttons and you only want to see the state for "Bob", just type "Bob" into the input box, and all buttons except those containing "Bob" will be hidden.

Dial box

The dial box is powerful option, with multiple actions:

Option	Action
Dial to a phone number	When your phone is idle, just type in a number and press ENTER: your phone will ring and will originate a call to that number

Option	Action
Transfer to any number	When you are on an active call you can type in a number (either an internal extension or an external number) and press ENTER. This will transfer your current call to that number.
Invite any number to a conference	If you select a conference button, type a number in the dial box, and hit enter, the Switchboard will originate a call to the number typed, and will send the call to the selected conference.
Direct VoIP dial	Type a sip address with the format: SIP/$(ext)@$(domain) and the Switchboard will originate a direct sip call to that extension @ server.
Phonebook search	Anything you type here will be searched in the visual phonebook as you type, and results will appear as a selection list at the bottom of the box. You can use the arrow keys to select the proper result, and then click on ENTER to originate a call to the selected number.
.tel domain lookup	If you type a .tel domain, the system will perform a DNS domain lookup and display any matching results that are found. You can find voice numbers, direct VoIP dial, and Web sites. If you select a result and press enter, a call will be originated to the phone number, direct VoIP call, or the target Web page will be opened in a new window.

Presence select box

You can use this to set your presence information. The information is stored in the CompletePBX database, and is immediately shown to other Switchboard users. Any presence state other than **Available** will set your phone into **do-not-disturb** (DND) mode. In a similar manner, if you use the appropriate feature code (*78 or *79) to toggle the DND status of your extension, the change will also be reflected in the Switchboard.

The presence state will not affect your dialing behavior: it will just tell other Switchboard users about your current availability.

A little presence icon on your button will reflect your status. If you mouse over the presence icon of any extension you will see a tooltip with the textual state, in addition to the color.

If the standard present states do not answer your requirements, the last option in the select box, named **Other,** allows you to specify any free text for your presence.

Extensions

Extension buttons show lots of information packed in a small rectangle. There are two colors for the button:

- green indicates that the extension is free and not engaged in a call
- red indicates that the extension is busy and currently engaged in a call

The icon will remain green when the extension is ringing, but the line icon will indicate the ringing

state. There is also a third (yellowish) color when DeviceAndUser mode enabled in CompletePBX to indicate that the device is in adhoc mode. This is an example button that is ringing:

The button has a number of elements:

Element	Explanation
1 - Presence icon	Indicates the presence state for the extension. A green circle means that the extension is not in a call. A red circle indicates that the extension is busy or in a call. A little card icon indicates a special presence that was set by the user. The color of the custom presence is configurable. If you mouse over the icon, a tooltip with the custom presence text will appear. Lastly, if someone initiated a call recording from within the Switchboard, a tape icon indicates that the call is being recorded. Clicking on this icon will display an action menu: if you have the appropriate permission, you will be able to add, remove, or pause the member in a queue, or send an email to that user (if the email setting is defined in the button configuration).
2 - Button Label	Displays the text label for the extension. It shows the extension number followed by the text label that is associated with the button.
3 - Information icon	Shows if the extension is paused, by showing a clock (for queue members). If the extension is engaged in a call that came from a queue, mousing over the icon will display an informational icon that will show the queue from where the call came from.
4 - Voicemail icon	This icon, in the shape on envelope, indicates if there is voicemail waiting or stored for the extension. The number of messages is displayed in the tooltip when you mouse over it.
5 - Line activity icon	Indicates whether the line is ringing, is engaged in an outgoing call (right arrow), engaged with an incoming call (left arrow), or on hold (hourglass).
6 - Line callerid	Shows the callerid name and number, if available
7 - Line timer	Shows the timer for the current call.

Performing Actions

Buttons can be clicked to select or deselect them. When a button is selected, the border of the button will become red. Once a button is selected it will become the target button for any action that you choose.

Every button in the toolbar requires that a destination button also be selected. Clicking an action button with no extension selected will not trigger any action. Some actions will work only on the active line for that extension. For example, if an extension has two calls, and one of them is on hold, if you click the record button it will start recording only the active line and not the line on hold. The same happens with the hangup button: if you have two calls on your phone and one is on hold, clicking the transfer button will redirect your **active** call, and not the call on hold.

To transfer a call, when talking to someone, click on the target extension (which will be highlighted in red) and then click the transfer button on the toolbar.

To record a conversation to disk, click on the extension button that you want to monitor and then click on the record button on the toolbar.

Action submenu
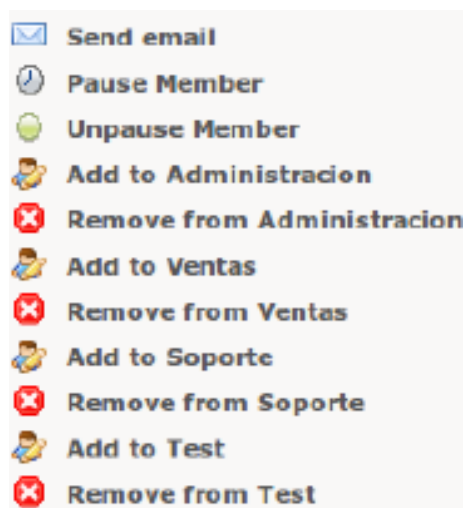
The presence icon for any type of button can be clicked to perform additional actions. A single click on the icon will cause a popup menu to appear, listing the available actions for the selected button. In this manner, you can add, remove or pause members in a queue, send email to users, or pickup parked, queued or trunk calls.

The action submenu for extensions allows you to send emails to users (if the email setting is defined in the button configuration). You can also pause or unpause a queue member, or you can add or remove the extension from any available queue. You cam pickup calls from parking slots, trunks and queues.

The submenu for queue members allows you to remove them, pause, or unpause them.

For conferences you can perform global actions or individual actions, such as mute, unmute, lock, unlock, etc.

Visual Phonebook

At the top left of the main panel window, there is a phonebook icon. When clicked you will be

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presented with a phonebook application where you can add, edit or delete entries. Click on the **Add** button to insert new records, or click the action buttons for each record to view, edit or delete them. There is a search box that allows you to search for any string or number. There is also an **Export** button that lets you export your phonebook in CSV format.

Phonebook entries will be searched in real-time when you type something on the **Dial** box, so

it can be used as a company directory. It will also be searched when an inbound call is received. If the caller id matches one of the entries, it will use the name and picture of the contact for a call notification that will pop up at the lower right hand of the Switchboard screen as depicted below:
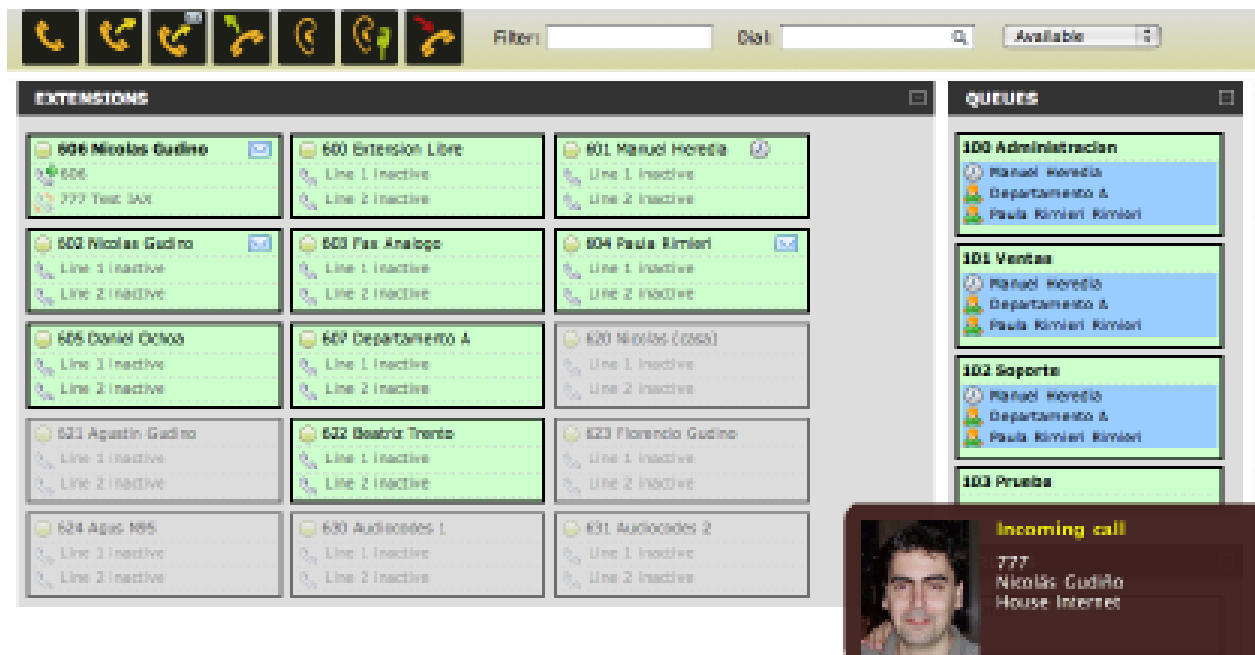
Importing Data

You can import CSV data into the Switchboard phonebook by clicking the Import icon. When you click it, it will show you an extra field where you can type or search for the the csv file to import. The first line on the CSV file must contain the field names, and the following lines the records you want to insert.

A sample file would look like this

firstname,lastname,company,phone1,phone2,private

Nacho,Rodriguez,Telecorp,123900001,125900002,no

John,Doe,Simcorp,55555555,,606,no

Most fields are self-explanatory, such as **firstname** and **lastname**. The **private** field lets you mark whether a record as private or not; private records can only be edited or viewed by their owner.

Queues

Queue buttons have some particularities. Besides the queue name, they will show the list of members or agents belonging to the queue, with a little icon representing the member state, and also the list of calls waiting on the queue with their call counter and timer, as depicted below:
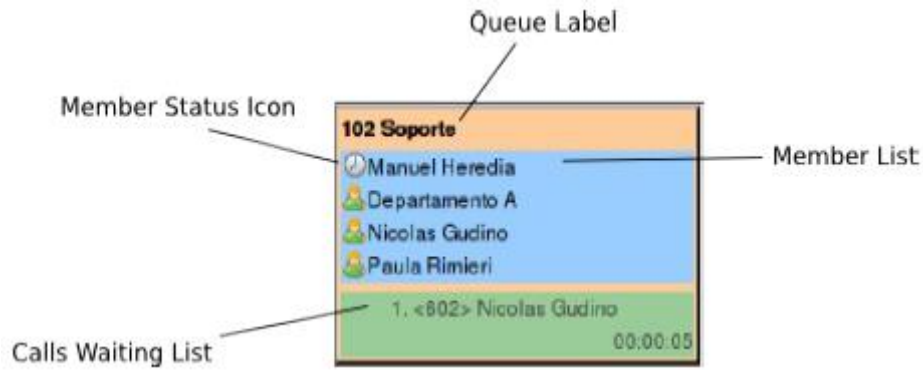
They also can be used to filter out extension buttons. When a queue button is selected, only the extensions that are part of the queue will be displayed. This is particularly useful for call center managers, so that they can focus on a particular queue activity, removing the 'noise' of other queues and extensions.

The state icon for the queue members might be:

 = ready

 = busy

 = unavailable / invalid

 = paused.

The Switchboard can also monitor agent or device names, and rename extension buttons accordingly.

Pickup queued calls

You can click on a call waiting in the queue to display the pickup submenu. If your extension is allowed to pick up calls, you can click on the action and the waiting call will be redirected directly to your extension:

Trunks

Trunk buttons show the number of channels active for the particular trunk. Trunks also display detailed information regarding bridged calls, with channel name and callerid. It is important to note that bridged channels are the ones that are linked to another channel. Calls that are not answered, or that are inside an application, such as IVR, voicemail, etc., will not be displayed.

Click on the channel name to bring up an action submenu, with the option to pick up the call.

Clicking it will redirect the call to the logged in extension.

Conferences

Conference buttons will show all participants in the conference. They also include special actions to be performed on the conference itself or to a selected participant.

There is a little icon to the left of the conference label: click on the icon to display a special action

menu that will let you lock or unlock the conference, or mute (or unmute) all participants in the conference.

To the left of every participant there is also a member icon, it will be a green user for regular participants, or a blue suited user when that participant is the admin user. Click on the icon to open up the action menu for the selected participant, where you can toggle mute for the user or kick them out of the conference.

You must have the **meetme** permission in order to activate these actions.

10. System Protection

Introduction

After a lot of hard work and configuration, we finally have a fully-functional CompletePBX configured precisely the way we need. Now, we need to make sure it stays that way. Even with the best hardware, the failure of a system component is a danger that is always present. Without proper protection and backups, we could wind up without a working PBX and have no way to restore it. In this chapter, we will discuss the following:

- System protection using UPS devices, redundant components, and surge protection
- Making one-time backups
- Configuring recurring backups
- Restoring a backup
- Maintaining backup sets

System protection

There are many ways to protect the components of a server from failure or damage. While dealing with a CompletePBX, these protection methods are even more important. As CompletePBX controls voice communications for a company, downtime often means lost income and angry employees.

While the installation and setup of the equipment listed here is beyond the scope of this book, it is worth keeping them in mind during installation.

Uninterruptible power supplies

An **Uninterruptible Power Supply** (**UPS**) is essential to every VoIP system. A UPS acts as a battery backup in the event of a power failure. If the power supply is cut, anything plugged into the UPS will continue to run until the battery runs out. Most UPS units will also be able to send a shutdown signal to the server when the battery is nearly empty, allowing a clean shutdown.

Some UPS units will also provide a power conditioning service, sending a stable level of power to any attached device. This protects any attached equipment from surges or dips on the power line that can be very damaging. Note that power conditioning is not included in all the makes and models of UPS, so be sure to check before purchasing.

Also note that for a VoIP PBX to continue to be truly effective during a power outage, it must be able to maintain an Internet connection for any VoIP trunk. It is generally a good idea to ensure that the UPS will power not only the PBX but also any modems, routers, and switches required for connectivity.

Redundant components

Most major server manufacturers provide the option for various types of redundant components. The most common redundant component is the hard drive, usually set up in a RAID configuration. This allows a hard drive to fail while the server keeps running. Power supplies are also a common redundant component, again allowing for one to fail without downing the server.

Though redundant components can be costly, they can often prevent downtime for the entire life of a PBX.

Redundant servers

Depending on the budget that is available for the PBX, it may be feasible to have an entirely separate server built and ready to take over call routing, in the event that the primary server fails. The secondary server may be running at all times and have a recent copy of all critical configuration data (known as running a **hot spare**). Alternately, it may be shutdown at all times, and only have a base installation of Linux, CompletePBX, and CompletePBX without any specific configuration data (known as running a **cold spare**).

A hot spare will usually be built using the same steps as the primary server. Once the primary server is configured with users, trunks, and call targets, the configuration data will need to be replicated from the primary server to the backup server. In less complicated setups, this can simply mean taking a backup on the primary server and restoring it on the hot spare. In order to ensure that data is current, the backup and restoration process should happen at least once per day. More complicated setups may have live replication of the MySQL database, CompletePBX configuration files, the voicemail directories, so that when data is changed on the primary server the changes are reflected on the hot spare immediately.

Running a hot spare is typically an expensive endeavor. Initial hardware purchase cost is doubled, and as a hot spare is running at all times, the hardware components will tend to require replacements at the same time as the primary server's components. In addition, the manpower required to maintain data replication between the two servers is often very expensive. Nonetheless, if the budget exists for it, a hot spare will always be the most effective way to reduce downtime.

A cold spare will usually be built using the same steps as the primary server, until the point where the server is being configured with user accounts, trunks, and call targets. A cold spare will be left with a running CompletePBX installation without any configuration on it. Regular backups of the primary server should be taken once it is fully configured. The cold spare will remain turned off during the normal operation of the primary server.

Whenever the system packages or CompletePBX modules are updated on the primary server, the cold spare should be booted up to have the same updates applied to it. In the event that the primary server fails, the cold spare should be booted up and a recent CompletePBX configuration backup should be restored to it. As with a hot spare, the initial hardware cost of running a cold spare is doubled. However, as a cold spare is rarely powered on, the hardware will tend to require replacement far less frequently than a hot spare, reducing the maintenance costs over time. Also, as data replication only happens in the event of a disaster, there is no additional cost for maintaining the data on the cold spare.

A cold spare is a good option if the implementation budget for the PBX allows for two sets of hardware, but maintenance does not allow for the expense of constantly replicating data.

Surge protection

The most common type of surge protection is for power lines, but surges can affect other components of the CompletePBX that will impact functionality. If analog lines are in use with a DAHDI card or ATA, a power surge down the phone lines can ruin equipment. A power surge down a cable or DSL line can take a modem out of operation, or any other routing equipment attached to it. It is important to install surge protection on these types of entry points to CompletePBX.

Backups

CompletePBX allows a system backup of voicemails, system recordings, system configuration, CDRs, and Operator Panel configuration to be taken. Backups can be taken on a one-off basis, or a

scheduled basis.

Backing up voicemail will back up all the voicemail messages and outgoing voicemail greetings for all the users configured on the system. If this is not backed up, then all the users will have to re- record their greetings, and they will be unable to retrieve historical voicemails which were received before the failure.

Backing up system recordings will back up all the voice prompts and custom music on hold recordings. Without this backup, prompts for IVRs, follow-me, queues, and music on hold recordings will all have to be re-recorded or re-uploaded after a system failure.

Backing up system configuration will back up all the configuration data. This includes all of the configuration data for trunks, users, devices, IVRs, queues, time conditions—pretty much all the configuration data on the system. Without this backup, the system will have to be reconfigured all over again after a failure. This will be akin to starting from scratch.

Backing up CDR data will back up all the call detail records as far back as the system has them stored. Without this backup, if the system needs to be rebuilt all CDR records will be lost. There is no way to rebuild CDR records without this backup.

Backing up the Operator Panel will back up all the HTML and configuration data for the Switchboard. Without this backup, any Switchboard customizations will have to be completed again, and the Switchboard will have to be reconfigured.

Configuring backup

In order to set up a backup, navigate to the **Admin** menu and select the **Backup & Restore** dialog, click on **Backup** in the menu on the right, and click on **New Backup**:
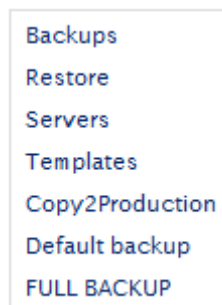
The **Backup** dialog will be displayed as follows:

Backup

Backup Name

Description ❓

Items

Backup Items ❓   **Templates** ❓

➕ CDR's
➕ Config Backup
Type Path/DB Exclude Delete   ➕ Exclude Backup Settings
                                                        ➕ Full Backup
➕   ➕ Recordings
                                                        ➕ System Audio
                                                        ➕ Voicemail

Hooks

Pre-backup Hook ❓

Post-backup Hook ❓

Pre-restore Hook ❓

Post-restore Hook ❓

Backup Server

Backup Server ❓ This server ▾

Storage Locations

Storage Servers ❓   **Available Servers**

                                        ➕ Legacy Backup (local)
                                        ➕ Local Storage (local)

Backup Schedule

Run ❓ Never ▾

Maintenance

Delete after ❓ 0   Minutes ▾

Delete after ❓ 0   runs

Save   Delete

In the first section, select the items that should be backed up in this backup schedule. Note that a single CompletePBX system can have multiple different backup schedules configured. It is possible to back up the **System Configuration** daily, **VoiceMail** weekly, and **System Recordings** and **CDR** monthly, if desired.

In order to create a one-time backup, select **Never** from the **Run** drop-down menu:

In order to set up a scheduled backup, select one of the predefined schedules from the **Run** drop-down menu, or select **Custom** and select the appropriate schedule from the table.

Click on the **Submit Changes** button in order to save the schedule. Once saved, the backup will appear in the menu on the right:

Clicking on the name of an existing backup schedule will load the **Backup** screen, allowing the schedule and items that are being backed up to be edited.

Maintaining and protecting backups

The way in which CompletePBX provides backups is simple and effective, but it presents two major problems: storage and protection. Similar to how call recordings have to be managed or they will fill up the available storage space, CompletePBX backup schedules will continue to back up the system indefinitely. The backups must be maintained or the system will eventually run out of space and cease to function.

Once the problem of backup maintenance is taken care of, the problem of backup protection still exists. The backups taken by CompletePBX reside on the server itself. If the hard disk crashes or the local backup becomes corrupted, there will be no way to restore a failed system.

Maintaining backups

Maintaining backups is best accomplished by a script that simply looks for backups older than a certain threshold and deletes them. The following script will search for all backups older than 90 days and delete them. The script can be adjusted to suit the appropriate backup retention policy for your PBX by changing the value of the *BACKUPEXPIRY* line as follows:

The following script is called *OldBackupDeletion.sh*:

```
#!/bin/bash
# Change this path to reflect your backup storage location
# (default is /var/spool/asterisk/backup)
BACKUPS=/var/spool/asterisk/backup
# Change this number to reflect the maximum age of backups (in days)
BACKUPEXPIRY=90
# Change this number to reflect the maximum age of the # deletion logs (in days)
LOGEXPIRY=365
# Current date
DATE='date'
# Delete recordings older than $EXPIRY days
find $BACKUPS -mtime +$BACKUPEXPIRY -exec rm -rfv > removal-$DATE.log\
# Delete log files older than $LOGEXPRY
find . -mtime +$LOGEXPIRY -exec rm -rf\
```

This script can be run once daily to find backup sets older than the age limit and delete them. Adding the following line to cron will execute the script from */etc/backupdeletion* daily at 5:00 a.m.:

```
0 5 * * * /etc/backupdeletion/OldBackupDeletion.sh
```

Protecting backups

There are a number of ways to protect backups. Each one may protect a particular deployment scenario better than the other. While the specifics of each protection scenario are outside the scope of this book, it is worth keeping them in mind when setting up backups to make sure that a backup is always close at hand. The most common methods for protecting backups are as follows:

- Redundant hardware (specifically, hard disks in a RAID configuration)
- Automating the copy of backups to an external hard disk or network location
- Automating the copy of backups to an off-site backup server

VoIPon   www.voipon.co.uk   sales@voipon.co.uk   Tel: +44 (0)1245 808195   Fax: +44 (0)1245 808299

Restore

Backups taken by CompletePBX can only be restored back to a running CompletePBX system. In the event of a total failure, the operating system will have to be reinstalled along with CompletePBX and MySQL, and backup archives will have to be copied back onto the server before a backup can be restored.

- Restore to the same version of CompletePBX only
- Backups should always be restored to the same version of CompletePBX that created the backup.
- A backup may be restored to a different version without error, but there is a good chance that something will be broken and cause further headaches down the line

Every backup that is taken by CompletePBX is stored in *var/spool/asterisk/backup*. Each backup schedule has its own subfolder in this directory. If a backup is being restored to a newly built system, the original subfolder and all contained backup archives must first be copied to *var/spool/asterisk/backup* before the backup will be accessible from within the CompletePBX interface.

In order to restore a backup, navigate to the **Admin** menu and select the **Backup & Restore** dialog, click on **Restore** in the menu on the right. The **Restore** screen will be displayed. If you have uploaded the backup file to your CompletePBX box, you can use the Browse button to select the file from which you want to restore. Alternatively, select one of the backup locations from the right-hand menu.
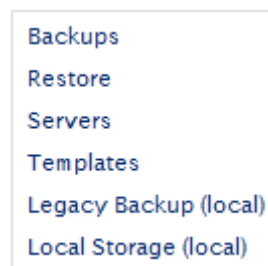
Every backup schedule that has been set up and executed will be present in the restore menu, including both one-time backups and recurring backups. Click on the name of the backup set that is being restored to show a list of all available backup images from that backup schedule:
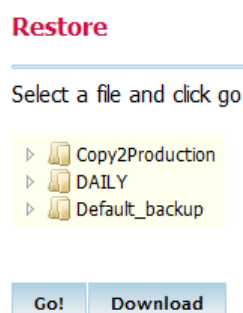
Images are titled by the date and time they were taken. Click on the name of the backup image that should be restored to show all data contained within the image:

Restore

Select a file and click go

▲ 📁 Copy2Production
    20130320-162416-1363789456-346122097.tgz
▲ 📁 DAILY
    20130410-000001-1365541201-1365475570.tgz
    20130411-000001-1365627601-1053508534.tgz
▲ 📁 Default_backup
    20130306-155720-1362578240-222299108.tgz
    20130401-000002-1364763602-905909945.tgz
    20130501-000001-1367355601-528768693.tgz

Name: Copy2Production
Created: Wed Mar 20 2013 16:24:16 GMT+0200 Time)
Files: 5073
Mysql DB's: 1
AstDB's: 1

Go!   **Download**

The various links provide the ability to delete the set, restore the entire set, or selectively restore individual parts of a set. It is possible to restore multiple parts of a particular set individually.

Restore

Select files and databases to restore:

▲ ☐ 📁 etc
    ☐ amportal.conf
    ▷ ☐ 📁 asterisk
    ▲ ☐ 📁 dahdi
        ☐ genconf_parameters
        ☐ init.conf
        ☐ modules
        ☐ modules.rpmsave
        ☐ system.conf
        ☐ system.conf.bak
        ☐ system.conf.rpmsave
    ☐ freepbx.conf
▲ ☐ 📁 var
    ▲ ☐ 📁 lib
        ▷ ☐ 📁 asterisk
    ▲ ☐ 📁 www
        ▷ ☐ 📁 html

PBX Settings ❓ ☐

Restore

In order to start restoration, click on the item that should be restored. CompletePBX will warn about overwriting the item that is being restored. Click on **OK** to proceed or **Cancel** to stop the restore:

Precaution!

It is a good idea to take a full backup of all the CompletePBX configuration data prior to attempting a restore. The restore process will completely overwrite all the existing configurations for the component that is being restored. If the wrong backup set is accidentally restored, then it is not possible to undo the changes.

Once the restore is complete be sure to click on the red-colored **Apply Config** bar at the top of the screen in order to reload all restored changes into the live configuration being used by CompletePBX.

Summary

We are now prepared to handle disaster. We now know the appropriate steps that we should take in order to prevent the failure of CompletePBX. Our CompletePBX system is backing itself up regularly, and we are maintaining and protecting those backups. In the unfortunate event of a systems failure, we are also prepared to restore from one of those backups.

In the next chapter, we will discuss the ins and outs of securing CompletePBX.

11. Security and Access Control

No matter which hardware platform it runs on, at the end of the day a CompletePBX-based phone system is just like any other server. It runs on software that must be secured and maintained to ensure the system does not fall prey to hackers or malicious users. Given the nature of a PBX there is often no leeway afforded for downtime in the event that a system is compromised, so the best practice is to know how to protect the system to start with. In this chapter, we will discuss the following:

- Use strong passwords for SIP and IAX2 extensions
- Define secure SIP settings in the General Settings dialog
- Create a safe dialing plan
- Securing remote access to the PBX
- Securing CompletePBX administration through administrator accounts

SIP/IAX2 extensions configuration.

Make sure that you define long and complicated passwords for SIP and IAX2 extensions. They should consist of a mixture of digits, upper-case, and lower-case letters. The minimum password length should be 8 characters.

Define IP address filters for the local extensions. For example, if you know that extension 201 is assigned to a SIP phone that is installed in the local office then you can define the following IP filter for this extension:

> **deny: 0.0.0.0/0.0.0.0**
> **permit: 192.168.1.0/255.255.255.0**
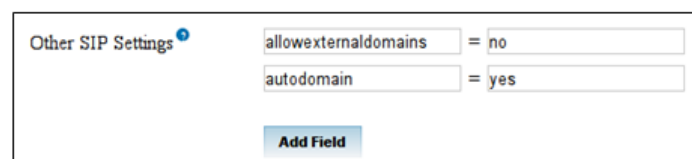
General SIP Settings

Disable inbound calls from unknown sources. This can be done by configuring the **Allow SIP Guests** parameter as "No". You can configure this by navigating to the **Settings** menu, selecting the **SIP Settings** dialog, and navigating down to the **Advanced General Settings** section. This value is the default setting for CompletePBX.

Disable accepting requests for domains not serviced by your CompletePBX. You can configure this by navigating to the **Settings** menu, selecting the **SIP Settings** dialog, navigating down to the **Other SIP Settings** section, and adding the following parameters:

allowexternaldomains=no disables requests from domains not defined in the CompletePBX network
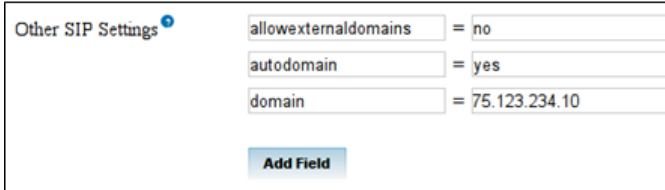
autodomain=yes implicitly adds CompletePBX IP addresses to the list of the local domains.

Note that the external SIP endpoints and SIP servers of Internet Telephony Service Providers (ITSP) will send SIP request to the PBX by using the NAT router external interface IP address. By default, CompletePBX will reject such requests. In order accept inbound SIP requests, click on the **Add** button, and define one or more **domain** parameters.

For example, if the external IP address of your CompletePBX is 75.123.234.10 then you must define:

Make sure that CompletePBX will reject unwanted SIP requests with the same reject reason code, regardless of the real reason. This significantly hampers brute-force attackers from guessing the SIP user name and secrets. This is configured in CompletePBX by default in the */etc/asterisk/sip_general_additional.conf* file, with the parameter *alwaysauthreject=yes*

Dialing Plan Configuration

Do not configure uncontrolled trunk-to-trunk calls. For example, do not define the context **from-internal** for any trunk.

Make sure that DISA and Call-back are well protected by using strong passwords.

You should define passwords (PIN) for all outbound routes that are used for International calls. This will significantly hamper intruders from making malicious calls. Refer to the *PIN Sets* section in Chapter 8,

Personalizing Your PBX for help configuring PIN sets.

Remote access and lock down

The point at which a system is opened up so it can be remotely administered is almost always the point of compromise in an intrusion. It is a good idea to close off the system as much as possible from the outside world, in addition to locking down all network access to the server.

PBX must be installed on a protected LAN, and must not be directly connected to the public Internet. The LAN must be protected by a Firewall/NAT router. The built-in firewall that is installed on CompletePBX could be optionally used as an additional means of protection.

- Do not expose the SIP (5060/udp) or IAX2 (4569/udp) ports if you have not defined any remote extensions.
- Use a non-standard port for SSH. Instead of exposing port 22/tcp, define port forwarding from a not well-known port, for example 4223/tcp, on the external interface to port 22/tcp of CompletePBX.
- Enable the SSH connection from specific IP address only. For example it could be the IP address of the company that provides technical support for you CompletePBX.
- Optionally it is possible to disable the password authentication method in the CompletePBX SSH server configuration and use only private/public key authentication. Refer your SSH client documentation for further details of the private/public key configuration.
- Never expose the HTTP (80/tcp) port at all. Always use SSH tunneling to access the CompletePBX Web interface.

VPN or SSH tunnels

The most secure method of remote access is the one that does not exist directly at all. If the organization where the PBX will be deployed has an existing VPN solution, then the most secure way to access their CompletePBX interface is over a VPN tunnel.

If no VPN exists, it is still possible to reduce the attack surface considerably by forcing Web access to CompletePBX through an SSH tunnel.

SSH tunneling under Linux

In order to generate an SSH tunnel under a Linux operating system, use a command similar to the following:

>   sudo ssh -L 3000:127.0.0.1:80 root@<external PBX IP address>

3000 is the port on your computer that will be forwarded to port 80 on the CompletePBX. You replace this value with any other free port on your local machine.

<external PBX IP address> is the externally accessible IP address of the CompletePBX to which you to connect.

For example, if CompletePBX has an an external IP of 2.2.2.2, our SSH command would look like this:

>   sudo ssh -L 3000:127.0.0.1:80 root@2.2.2.2

The above command will map local port 3000 to port 80 on the target machine. Once the SSH session has been established, opening up a Web browser and pointing it to http://localhost:3000 will load the CompletePBX interface on your machine.

SSH tunneling under Windows

Generating SSH tunnels under Windows can be accomplished using a wonderful utility called PuTTY. PuTTY can be downloaded from *http://www.chiark.greenend.org.uk/~sgtatham/putty/*.

In order to set up an SSH tunnel in PuTTY, first enter the external IP address into the **Host Name (or IP address)** field:
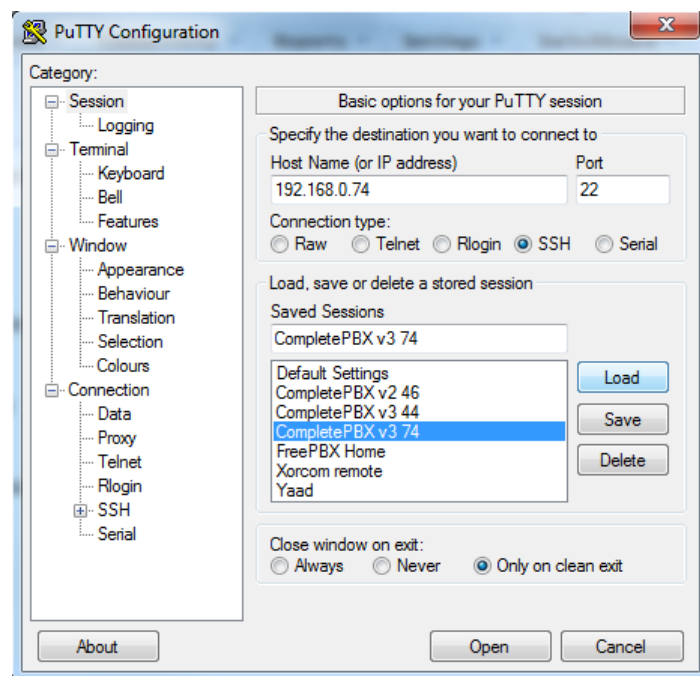
Expand **SSH** in the menu on the left (under **Connection**) and click on **Tunnels**:
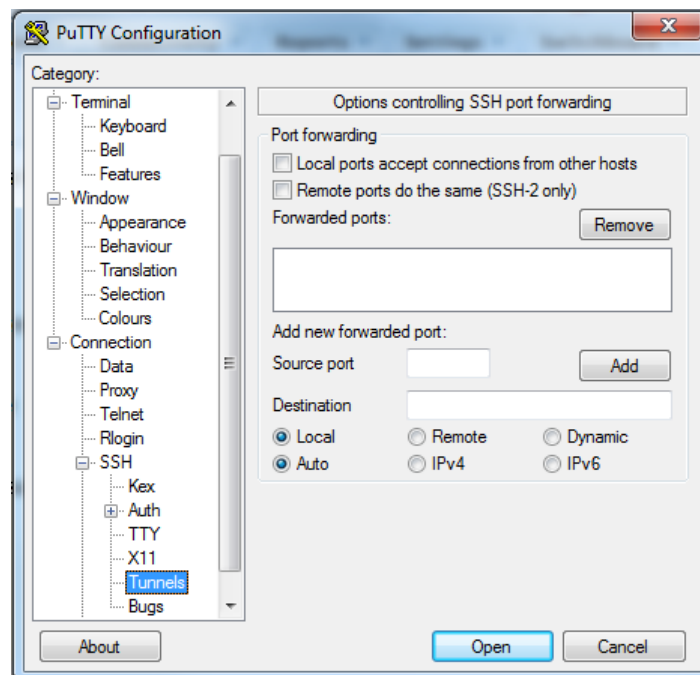
In the **Source port** field, type in 3000.

In the **Destination** field, type in the loopback IP address of the PBX server followed by a colon, and the port on which the Apache server is running.

Leave all other settings at with their default value, and click on the **Add** button. Now you can click on the **Open** button to open the SSH tunnel.
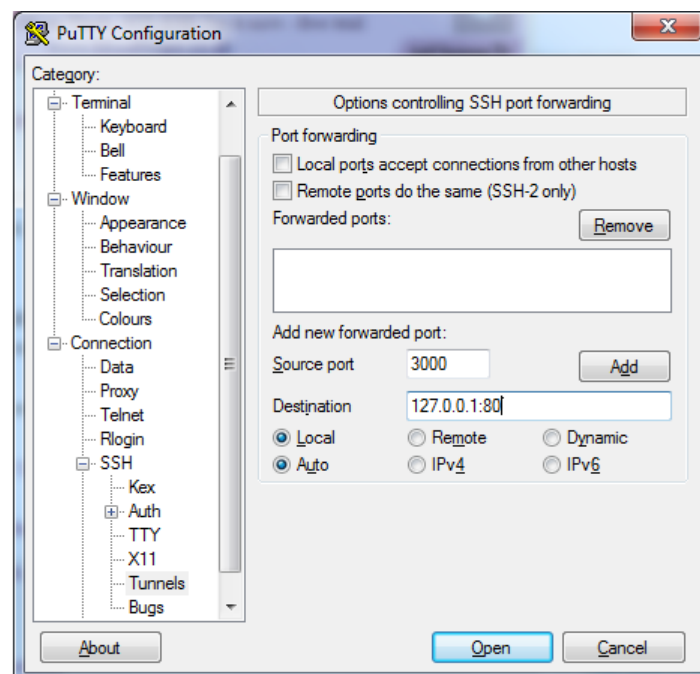
Once the tunnel is authenticated, CompletePBX can be viewed by opening a Web browser and browsing to http://localhost:3000

Administrator accounts in CompletePBX

CompletePBX incorporates the concept of different levels of administrator access. Administrator accounts can have their access restricted to specific departments, a specific extension range, or a specific set of modules. Separate administrator accounts should always be given out to anyone who administers the CompletePBX system. Any staffing changes will then simply require a person's account to be removed to ensure they no longer have access.
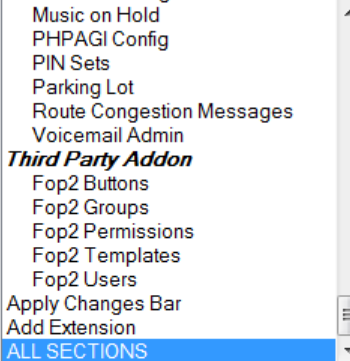
Managing administrator accounts and permissions

Once the Administrator module has been enabled, log in to CompletePBX, navigate to the **Admin** menu and select the **Administrators** dialog. The **Add User** screen is shown:

General Settings

Username ❓ : [   ]
Password ❓ : [   ]

Access Restrictions

Department Name ❓ : [   ]
Extension Range ❓ : [   ] to [   ]
Admin Access ❓ :

```
Music on Hold
PHPAGI Config
PIN Sets
Parking Lot
Route Congestion Messages
Voicemail Admin
Third Party Addon
Fop2 Buttons
Fop2 Groups
Fop2 Permissions
Fop2 Templates
Fop2 Users
Apply Changes Bar
Add Extension
ALL SECTIONS
```

[Submit Changes]

In the **Username** field, enter the account name for the user that is being set up. This is what the user will enter when prompted for username and password.

In the **Password** field, assign a password to the new user. Note that certain characters (such as an exclamation mark) have been known to cause problems in this field. It is best to stick with mixed case letters and numbers. Note that it is possible to create an administrator account that does not have a password. This is a dangerous practice and should be avoided.

The **Department Name** field is a great idea, but at the time of writing is not functional. This field is supposed to allow the user to be assigned to a specific department. When the Administrator module is enabled, certain call targets (such as IVRs and System Recordings) will have an additional field that will allow them to be owned by a specific department. Setting the Department Name field for an administrator will grant them access to all call targets that are in the same department without granting access to any other call targets in the same module. For example, if there is an IVR in the Sales department, an administrator with their Department Name set to sales will be able to change the settings on the Sales IVR but none of the other IVRs. Leaving this field blank will grant the administrator access to the call targets of all departments. Unfortunately, only the groundwork for this exists, the rest is not implemented. Setting this field will have no effect currently.

The **Extension Range** fields allow the administrator to be restricted to altering settings for a specific range of extensions. This is useful if extensions are specific to a department (for example, if the Sales department was extensions 250 through 299, granting a Sales administrator access to alter only these extensions may be useful). Leaving these fields blank allows the administrator to access all extensions.

The **Admin Access** field allows the administrator to be restricted to specific modules. In order to select multiple rows in Windows or Linux, *Ctrl + Click* on each desired row and on Mac, *Command/Apple + Click*. Each module that the administrator should have access to should be selected in this field. Note that **Apply Changes Bar** is a separate access module, so it is possible to give an administrator access to change something without giving them access to actually apply those changes.

Once all fields have been filled out appropriately click on the **Submit Changes** button followed by the red-colored **Apply Config** bar in order to make the new administrator active.
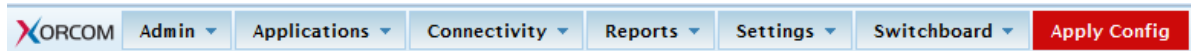
The administrator will show up in the menu on the right, and can be edited by clicking on their username:
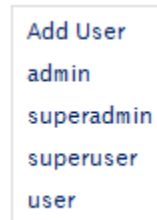
Summary

By now we should be masters of the CompletePBX realm. Our PBX should rival any of the competing closed-source solutions in terms of reliability, feature set, and security. Our system should now be locked down, allowing minimal access from outside the network. Each administrator should have their own login, and should be restricted to just the modules that they need to change.

12. Managing the System

Webmin

Webmin is a Web-based interface for Linux system administration. Using any Internet Web browser, you can manage your Linux system, including system time, timezones, firewall, user accounts, DNS, file sharing and much more. Webmin removes the need to manually edit Linux configuration files and lets you manage the system either from the console or remotely.

Access

To access Webmin, use any Internet browser to access port 6050 of your CompletePBX machine, For example, if the IP address of your CompletePBX installation is 192.168.0.11, then you should type **http://192.168.0.11:6050** in the address bar of your Internet browser.

> **Note!**
>
> By default, access to Webmin is only allowed from computers that are located on the same network.

You can login to Webmin using the username **root** and the root password (The password is documented in the Getting Started Guide, which was packed with your CompletePBX hardware), and clicking on the **Login** button.

This will give you access to the Webmin landing page:

System hostname	localhost.localdomain (127.0.0.1)
Operating system	CentOS Linux 6.2
Webmin version	1.600
Time on system	Thu Jan 24 00:16:34 2013
Kernel and CPU	Linux 2.6.32-220.13.1.el6.i686 on i686
Processor information	Intel(R) Core(TM) i3 CPU M 380 @ 2.53GHz, 1 cores
System uptime	0 hours, 20 minutes
Running processes	97
CPU load averages	0.01 (1 min) 0.07 (5 mins) 0.08 (15 mins)
CPU usage	0% user, 0% kernel, 0% IO, 100% idle
Real memory	1006.77 MB total, 219.71 MB used
Virtual memory	767.99 MB total, 0 bytes used
Local disk space	7.13 GB total, 2.27 GB used
Package updates	59 package updates are available

Webmin IP Access Control Configuration

By default, Webmin access is allowed only from computers located on the same network. It is possible to change this policy by defining additional IP addresses in the Webmin IP Access Control table.

Enter the Webmin console (as described earlier in this chapter), navigate to the **Webmin/Webmin Configuration** menu (on the left-hand side), and click on the **IP Access Control** icon.

For example, if it is necessary to enable access to Webmin from any host belonging to the 192.168.0.0 class C network and from 192.168.1.178 host, then the following should be defined:

When you have completed entering the list of allowed IP addresses, click on the **Save** button. Webmin will be automatically restarted and the new settings applied.

System Time Module

All Linux systems have an internal clock to keep track of the current time, even when the system is powered off. Linux systems effectively have two clocks:

- One that is maintained by the Kernel when the system is running,
- A separate hardware clock that runs all the time

The kernel's system time is set based on the hardware time when the kernel is loaded, so they should be synchronized. However, if one of the clocks is slower than the other it is possible for the hardware and system times to fall out of sync on a system that has been running for a long time.

All Linux systems store the time internally as the number of seconds since 1st January, 1970 GMT. For display, this is converted to a human-readable local time based on the system's configured time zone. All Linux distributions allow you to choose your time zone at install time, and include a tool for changing it later.

System Time

You can adjust both the system and hardware times by entering the webmin console (as described earlier in this chapter) and using the **System Time** module, which can be found under the **Hardware** category.

To set the system time, navigate to left-hand menu to the **Hardware** menu, select the **System Time** dialog, and select the **Set time** tab. Set the system time with current time information, by typing appropriate data into the **Date**, **Month**, **Year**, **Hour**, **Minutes**, and **Second** fields. When you have completed entering the data, click on **Apply** to permanently save the information.

It is also possible to force the system time to be set to the current hardware time, by clicking the **Set system time to hardware time** button. Either way, any change will immediately be visible to all programs running on your system, such as desktop clocks, syslog and mail clients.

Time zone

All UNIX systems store the time internally as a number of seconds since 1st Jan 1970, GMT time. When a program (such as the `date` command) displays the current time to the user, it needs to convert this to the correct time zone, which involves applying the correct time offset and adjusting for daylight savings.

For this reason, your system must have the time zone set correctly to display and log times properly.

You can manage the time zone by entering the webmin console (as described earlier in this chapter), navigating to the **Hardware** menu, selecting the **System Time** dialog, and selecting the **Change**

timezone tab. Click on current time zone to see a dropdown showing all available time zones, which are ordered by continent first, then by city or country. Select the time zone that you want to use, and click **Save** to permanently save the change.

NTP Server

The System Time module can also be used to set the system or hardware time based on the system time of another server. The other server must be either running an NTP (Network Time Protocol) server, or have the time protocol enabled in inetd.

Before you set the NTP server, you should visit **http://www.pool.ntp.org/en/** to get address of a public time server that is located in your region.

You can manage the NTP Server by entering the webmin console (as described earlier in this chapter), navigating to the **Hardware** menu, selecting the **System Time** dialog, and selecting the **Time server sync** tab. You can now type the hostname of IP address of the public NTP server that you found from the Internet into the **Timeserver hostnames or addresses** field. It is always better to choose a server that is close by, so that the effect of network latency is minimized. Multiple servers may be entered.

If you want to have the sync done regularly, change **Synchronize on schedule?** to **Yes** and select the times to sync at using the fields below it. Once per day is typically good enough.

Click the **Sync and Apply** button to permanently save your changes.

If the server cannot be contacted or does not support the NTP or time protocols, an error message will be displayed. Otherwise the time or times will be set and the page re-displayed.

Firewall

Shoreline (Shorewall) Firewall

Shoreline (also referred to as Shorewall) is used in network installations (as opposed to a personal computer firewall), since most of its strength lies in its ability to work with "zones", such as the DMZ or a 'net' zone. Each zone can have different rules, making it easy to have relaxed rules for the company intranet, yet clamp down on traffic coming in from the Internet.

You can access the Shoreline Firewall by entering the Webmin console (as described earlier in this chapter) and selecting the **Shoreline Firewall** from the **Networking** menu.

Shoreline Firewall
Shorewall version 4.5.4

Basic Firewall Management Operations

Buttons on the bottom of the page allows you perform some basic firewall management actions such as Stop Firewall, Clear Firewall, Show Status etc.

It is important to understand the difference between Stop Firewall and Clear Firewall buttons.

Button	Action
Stop Firewall	This button causes the firewall to block access from all hosts. Note that any TCP connections that were established before the Stop Firewall button was clicked will not be disconnected.
	You can define a list of hosts that will not be blocked, even after Stop Firewall has been clicked, in the **When Stopped** table.
	It is important to understand that button does not disable the firewall protection.
Clear Firewall	This button removes all firewall restrictions.
Apply Configuration	Applies the configuration changes that were made.

Firewall Configuration

Detailed Shorewall documentation can be found on the Shorewall Web site at http://shorewall.net/.

CompletePBX is supplied with a pre-configured and activated firewall. The most important firewall settings are located on the top four icons: Network Zones, Network Interfaces, Default Polices and Firewall Rules.

Network Zones

The firewall has two pre-configured network zones:

- **fw** – the PBX host itself
- **net** – everything outside the PBX

The zones listed on this page represent different networks reachable from your system. However, the entries here do no actually have any effect on the firewall - instead, they just define zone names and descriptions.

Select all. | Invert selection. | Add a new network zone.

	Zone ID	Parent zone	Zone type	Comment	Move	Add
☐	net		IPv4		↓	⊤ ⊥
☐	fw		Firewall system		↑	⊤ ⊥

Select all. | Invert selection. | Add a new network zone.

Delete Selected

- Manually Edit File   Click this button to manually edit the Shorewall file /etc/shorewall/zones, in which the entries above are stored.

Network Interfaces

Some CompletePBX models support multiple Ethernet interfaces. On machines that have multiple Ethernet interfaces installed, both eth0 and eth1 are assigned to the **net** zone.

Each of the network interfaces on your system that you want Shorewall to manage should be listed on this page, and associated with the zone that it is connected to. The loopback interface lo must never be listed.

Select all. | Invert selection. | Add a new network interface.

	Interface	Zone name	Broadcast address	Options	Move	Add
☐	eth1	net	None	optional,dhcp	↓	⊤ ⊥
☐	eth0	net	None	dhcp	↑	⊤ ⊥

Select all. | Invert selection. | Add a new network interface.

Delete Selected

Manually Edit File   Click this button to manually edit the Shorewall file /etc/shorewall/interfaces, in which the entries above are stored.

Default Policies

Default Policies enable all outgoing traffic from CompletePBX (the **fw** zone) to the outside (the **net** zone), and blocks all traffic in the opposite direction, i.e. all incoming traffic from the outside towards CompletePBX is blocked.

This page allows you to configure the default actions for traffic between different firewall zones. They can be overridden for particular hosts or types of traffic on the Firewall Rules page.

Select all. | Invert selection. | Add a new default policy.

	Source zone	Destination zone	Policy	Syslog level	Traffic limit	Move	Add
☐	Firewall	net	ACCEPT	None	None	↓	⊤ ⊥
☐	net	Any	DROP	info	None	↑ ↓	⊤ ⊥
☐	Any	Any	REJECT	info	None	↑	⊤ ⊥

Select all. | Invert selection. | Add a new default policy.

Delete Selected

Manually Edit File   Click this button to manually edit the Shorewall file /etc/shorewall/policy, in which the entries above are stored.

Firewall Rules

The Firewall Rules define 'exceptions' from the default policies. This allows you to enable inbound traffic for some particular services such as SIP, HTTP etc.

Firewall rules can also be defined in Shorewall macros (refer to http://shorewall.net/Macros.html for more detailed information). Most of the macros that appear in the Firewall Rules table are provided by Xorcom shorewall-cpbx-<version>.noarch.rpm.

This table lists exceptions to the default policies for certain types of traffic, sources or destinations. The chosen action will be applied to packets matching the chosen criteria instead of the default.

Select all. | Invert selection. | Add a new firewall rule. | Add a new comment.

Action	Source	Destination	Protocol	Source ports	Destination ports	Move	Add
TFTP/ACCEPT	Any	Any		Any		↓	T ↓
NTPbi/ACCEPT	Any	Any		Any		↑ ↓	T ↓
DNS/ACCEPT	Any	Any		Any		↑ ↓	T ↓
DRBD/ACCEPT	Any	Any		Any		↑ ↓	T ↓
HTTP/ACCEPT	Any	Any		Any		↑ ↓	T ↓
SSH/ACCEPT	Any	Any		Any		↑ ↓	T ↓
SIP-UDP/ACCEPT	Any	Any		Any		↑ ↓	T ↓
SIP-TCP/ACCEPT	Any	Any		Any		↑ ↓	T ↓
SIP-TLS/ACCEPT	Any	Any		Any		↑ ↓	T ↓
RTP/ACCEPT	Any	Any		Any		↑ ↓	T ↓
IAX2/ACCEPT	Any	Any		Any		↑ ↓	T ↓
Asterisk-AMI/DROP	Any	Any		Any		↑ ↓	T ↓
SwitchBoard/ACCEPT	Any	Any		Any		↑ ↓	T ↓
Webmin-cPBX/ACCEPT	Any	Any		Any		↑ ↓	T ↓
Ping/ACCEPT	Any	Any		Any		↑	T ↓

Select all. | Invert selection. | Add a new firewall rule. | Add a new comment.

Delete Selected

Manually Edit File   Click this button to manually edit the Shorewall file /etc/shorewall/rules, in which the entries above are stored.

In the **Action** column you can see the *macro-name/action-parameter*. For example:

HTTP/ACCEPT means that the IP packets that fulfill the conditions described in the macro HTTP (*/usr/share/shorewall/macro.HTTP*) will be accepted from outside and can also be sent from the CompletePBX to the outside. An action that has either the action-parameter DROP or REJECT will cause traffic to be prohibited for this service.

Asterisk-AMI/DROP means that communication to the Asterisk Management Interface (AMI) is blocked as well as any Computer Telephony Integration (CTI) applications. This is done for security reasons. If you want to allow such communication, then the action-parameter should be changed to ACCEPT. It is recommended to restrict access to AMI to specific hosts only. For example, if we want to enable the access to AMI for the host on which the company CRM system is running (e.g., 192.168.0.130):

- Click on Asterisk-AMI/DROP
- Check on the "**Only hosts in zone with addresses,** and write the 192.168.0.130 address in the next field
- Click on the **Save** button

Now you will see the changes in the rules table:

- Click on Return to list of tables button
- Click on the **Apply Configuration** button.

Useful CLI Commands

Defining wrong firewall settings may block any IP communication with CompletePBX. In such cases it will be necessary to perform some actions from the CompletePBX Linux console.

Command	Action
shorewall clear	Clear all firewall restrictions.
shorewall start	Firewall start to work according the current configuration.
shorewall safe-start	Shorewall is started normally. You will then be prompted asking if everything went all right. If you answer "n" or if you fail to answer within 60 seconds, a *shorewall clear* is performed for you.
shorewall logwatch	See the dropped/rejected packet log in real-time.
shorewall show connections	Displays the IP connections currently being tracked by the firewall.
shorewall show macro *name*	Displays the file that implements the specified macro.

13. Feature Codes

With all modules installed, CompletePBX has more than 45 feature codes that users can dial to toggle features or run applications. Feature codes can be customized from the **Feature Code Admin** page (they can be located by clicking on the **Feature Codes** option under the **Basic** menu in the CompletePBX interface). The default feature codes and their actions are listed as follows:

Blacklist

Feature Code	Default	Action
Blacklist a number	*30	Prompts the user to enter a telephone number. The entered number is then added to the user's blacklist. Inbound calls will not ring an extension if they are on that extension's blacklist. Blacklisted callers will be told that the number they dialed is no longer in service.
Blacklist the last caller	*32	Adds the last number that called the user to the blacklist.
Remove a number from the blacklist	*31	Prompts the user to enter a telephone number. The entered number is removed from the user's blacklist.

Call Forward

Feature Code	Default	Action
Call Forward All Activate	*72	Prompts the user to enter a phone number or extension number. Any calls to the user's extension will be forwarded to the entered number until call forwarding is deactivated.
Call Forward All Deactivate	*73	Deactivates call forwarding on the extension of the user placing the call.
Call Forward All Prompting Deactivate	*74	Prompts the user to enter an extension number. Call forwarding will then be disabled on the entered extension. This feature code enables a user to remove call forwarding on their extension from another extension on the system, or from any phone if the users have DISA access.
Call Forward Busy Activate	*90	Prompts the user to enter a phone number or extension number. If the user is on the phone from that point onward, inbound calls will be forwarded to the entered number. If the user is not on the phone when an inbound call comes in, the user's extension will ring as normal.

Feature Code	Default	Action
Call Forward Busy Deactivate	*91	Disables call forward on busy service, on the extension of the user placing the call.
Call Forward Busy Prompting Deactivate	*92	Prompts the user to enter an extension number. Call forward on busy service will then be disabled on the entered extension. This feature code enables a user to remove call forward on busy service from their extension using another extension on the system, or from any phone if the users have DISA access.
Call Forward No Answer/Unavailable Activate	*52	Prompts the user to enter a telephone number or extension number. If the user does not answer an inbound call or the user's telephone is not registered with CompletePBX, inbound calls to the user will be forwarded to the entered number. If the user is on the phone when an inbound call comes in, the call will be processed normally (either by initiating call waiting or by failing over to another destination).
Call Forward No Answer/Unavailable Deactivate	*53	Disables call forward on no answer or unavailable service on the extension of the user placing the call.

Call Waiting

Feature Code	Default	Action
Call Waiting Activate	*70	Enables call waiting service on the extension of the user placing the call. If an inbound call comes into an extension that is already on the line and call waiting is enabled, then the user receiving the call will hear a beep and may see the caller ID of the new caller (depending on their telephone). The user receiving the call can then place their current call on hold and pick up the new caller. Without call waiting, the new caller would either receive a busy signal or would immediately be dropped to voicemail without the extension ringing.
Call Waiting Deactivate	*71	Deactivates call waiting on the extension of the user placing the call.

Core

Feature Code	Default	Action
General Call Pickup	*8	Allows the user to pick up a call that is ringing another phone as long as the user's pickup group matches the ringing extension's call group. For example, if extension 5000 is in call group 1 and extension 5001 is in pickup group 1, the user at extension 5001 can pick up a call that is ringing extension 5000 by dialing *8. If the user's pickup group does not match the call group of the ringing phone, dialing this feature code will have no effect.
ChanSpy	555	Allows the user to listen to an active call. After initiating the ChanSpy application, the user can dial the *key to cycle through all active channels.
Dial System FAX	666	Dials the extension that is defined as the default fax extension. The default fax extension is set on the General Settings page by changing the extension of the fax machine for the receiving faxes drop-down menu.
Directed Call Pickup	**(ext)	Allows the user to pick up a call directed to another extension. The (ext) token should be replaced with the extension being picked up. For example, if extension 5001 is ringing, another user could dial **5001to pick up the call.
In-call Attended Transfer	*2	When on a call, a user can dial this feature code to initiate an attended transfer. The user will hear a dial tone and at this point, they can enter an extension or telephone number. The users will be connected with the number they dialed. When the user hangs up the phone, the person they are transferring will be connected with the person the user called. Attended transfers are used when the person transferring the call wishes to speak to the person receiving the call before the call is transferred. A common scenario for this would be a call center agent transferring a call to their manager. The agent needs to first brief the manager on the problem before the call is transferred.
In-call Blind Transfer	##	When on a call, a user can dial this feature code to initiate a blind transfer. The user will hear a dial tone and at this point, they can enter an extension or telephone number. The person the user was speaking with will be immediately transferred to the number the user entered.
In-call Disconnect Code		When on a call, a user can dial this feature code to immediately terminate the call.

Feature Code	Default	Action
In-Call Toggle Call Recording	*1	When on a call, a user can dial this feature code to start the recording of the call using the CompletePBX MixMonitor application. This feature code works if the user's **Recording Options** are set to **On Demand** on their configuration page in the CompletePBX interface. Recorded calls are saved to the */var/spool/asterisk/monitor* directory by default.
Simulate Incoming Call	7777	Simulates an inbound call to the PBX. The call will be matched against any inbound route rules in the same way as an incoming call through a trunk would be matched.
User Logon	*11	If CompletePBX is configured in the DeviceAndUser mode, this feature code will allow a user to log in to a telephone with their extension and password. Once a user is logged in, any calls to the user's extension will ring the phone the user logged into.
User Logoff	*12	If CompletePBX is configured in DeviceAndUser mode, this feature code will allow a user to log out of a phone they are currently logged into. If a user is not logged in to any device when a call comes into their extension, the call will immediately proceed to the user's failover destination (this is normally voicemail, but it can be other extensions or telephone numbers if follow-me is configured).
ZapBarge	888	Prompts the user for a channel number. The user will then be able to listen to an active call on the DAHDI channel, which they entered. ZapBarge has been renamed to DAHDIBarge in newer releases due to trademark violations.

Dictation

Feature Code	Default	Action
Perform Dictation	*34	Prompts the user to enter a numeric filename followed by the pound (#) key. The user is then connected to the dictation application where whatever they say will be recorded. The dictation application allows recording to be paused at will, and can be switched to playback mode to hear the current recording.
Email completed dictation	*35	Prompts the user to enter a numeric filename followed by the pound (#) key. If a dictation recording with that filename exists, it will be emailed to the email address specified in the user's dictation options in the CompletePBX interface.

Do-Not-Disturb (DND)

Feature Code	Default	Action
DND Activate	*78	Activates the Do-Not-Disturb (DND) mode for the extension placing the call. Any calls directed to an extension with DND enabled will not ring any of the phones associated with that extension. Calls to an extension that has DND enabled will immediately continue to the extension's failover destination (this is usually voicemail, but could be other extensions or phone numbers if follow me is enabled).
DND Deactivate	*79	Deactivates the Do-Not-Disturb (DND) mode for the extension placing the call.
DND Toggle	*76	Toggles the state of Do-Not-Disturb (DND) mode for the extension placing the call.

Follow Me

Feature Code	Default	Action
Findme Follow Toggle	*21	Toggles follow-me mode on the extension placing the call. Any calls to an extension with follow me enabled will follow the call flow defined in the extension's Follow Me configuration page in the CompletePBX interface. If follow me is disabled, calls to the extension will ring any associated phones as normal, and fail over to voicemail (if enabled) or eventually disconnect the call (if voicemail is not enabled).

Info Services

Feature Code	Default	Action
Call Trace	*69	Plays back the telephone number or extension number of the last caller to the phone dialing the feature code.
Directory	#	Connects the user to the company directory. The directory is populated with all users who have voicemail boxes. The directory can be searched by first name and last name (this is a configurable option on the CompletePBX General Settings page). When a user finds the person they are looking for, they can opt to be transferred to that person.

Feature Code	Default	Action
Echo Test	*43	Plays back whatever the user is saying as they are saying it. The echo test is designed to provide a sense of the latency between the telephone dialing the feature code and the CompletePBX server.
Speak Your Extension Number	*65	Plays back the extension number of the extension placing the call.
Speaking Clock	*60	Plays back the current time on the CompletePBX server.

Paging and Intercom

Feature Code	Default	Action
Intercom Prefix	*80 (ext)	Connects the user to another extension via intercom functionality (the called phone does not ring, but immediately answers the call and places the call on speaker phone). The (ext) token should be replaced with the extension that the user is trying to connect to. For example, to start an intercom session with extension 5001, dial *805001. The extension being called must have intercom functionality enabled, and the phone must support auto-answer via SIP info headers. Most SIP phones provide this support.
User Intercom Allow	*54	Enables intercom functionality on the extension placing the call. Any user can initiate an intercom session with an extension that has this functionality enabled.
User Intercom Disallow	*55	Disables intercom functionality on the extension placing the call. If intercom functionality is disabled on an extension, any user who tries to initiate an intercom session with that extension will hear a message indicating that all circuits are busy, and hence the session will not be established.

Phonebook Directory

Feature Code	Default	Action
Phonebook dial-by-name directory	411	Works in conjunction with the CompletePBX Phonebook module. The user is prompted to enter three letters on their touch tone phone to make a search in the phonebook. When the desired entry is found, the user is given the option to press 1to dial the phone number associated with the matching entry.

Recordings

Feature Code	Default	Action
Save Recording	*77	Plays a single beep after which whatever the user says is recorded to a temporary file on the PBX. This is used in conjunction with the CompletePBX System Recordings module. The CompletePBX interface is used to name the temporary recording and transfer it to a more permanent location.
Check Recording	*99	Plays back the most recent temporary recording that was made by the user placing the call. It is used to verify that a recording sounds the way it was meant to. If the user is not satisfied with the recording, they can dial the Save Recording feature code again to re-record it.

Speed Dial Functions

Feature Code	Default	Action
Set user speed dial	*75	Allows the user to configure a speed dial entry. The user is prompted for a speed dial location followed by the pound (#) key. The speed dial location can be any number. The user is then prompted for the telephone number or extension number, which they wish to add to speed dial followed by the pound (#) key. For example, to add the phone number 555-555-1234 to the speed dial location 1000, dial *75, wait for the prompt, then dial 1000# and wait for the prompt, and now dial 5555551234#.
Speeddial prefix	*0 (num)	The (num) token should be replaced by the speed dial location being dialed. Dialing this feature code connects the user with the speed dial phone number associated with the speed dial location they entered. For example, to dial the telephone associated with speed dial location 1000, dial *01000.

Voicemail

Feature Code	Default	Action
Dial Voicemail	*98	Prompts the user to enter a voicemail box number and the password associated with that mailbox. If the user is authenticated successfully, they are transferred to the messaging center where they can listen to their voicemails and record their outgoing voicemail messages.

Feature Code	Default	Action
My Voicemail	*97	Assumes the user is calling to check messages in the voicemail box associated with the extension they are calling from. It prompts the user for a password. If the user is authenticated successfully, they are transferred to the messaging center where they can listen to their voicemails and record their outgoing voicemail messages.

14. CompletePBX Modules

CompletePBX has more than 50 optional modules available for installation from the online repository. A listing of all of the available modules and their functions as of the time of writing is as follows:

Basic

Module	Function
Built-in/Core	Provides basic call routing functionality and it is a built-in module. This includes Inbound Routes, Outbound Routes, Devices, Users, Extensions, and General Settings. The built-in module cannot be removed, and is installed by default when CompletePBX is first installed.
Feature Code Admin	Provides the ability to enable/disable feature codes, or change the numbers that are dialed to access certain feature codes. Feature codes are extensions dialed by users to access certain PBX features such as call forwarding activation, voicemail, and call waiting activation.
CompletePBX Framework	Core framework for CompletePBX. This is the framework that all of the other modules plug into. The framework encompasses the module administration page and the CDR/reporting pages.
CompletePBX Localization Updates	Provides the ability to install translations of the CompletePBX interface in localized languages.

Module	Function
System Dashboard	Provides a landing page that is shown whenever logging into the CompletePBX Administration. The landing page shows the following pieces of information: CompletePBX Notices Notices when new modules are available for update Warnings when routes point to non-existent call targets Warnings when trunks or extensions have weak passwords Warnings when default MySQL or CompletePBX passwords are in use Errors encountered by cron that are relevant to CompletePBX v1.1 (such as when a statistics gathering task fails) CompletePBX Statistics Total active calls Active internal calls (extension to extension) External calls (calls that are using an external trunk) Total active channels (how many people are actually speaking on phones IP phones online IP trunks online IP trunk registrations Uptime System (operating system) uptime CompletePBX uptime Last time CompletePBX was reloaded System Statistics Processor load average Processor current percent usage Physical memory in use Swap space in use Usage of all of the disk mount points Current send and receive statistics for all of the network interfaces

CID and number management

Module	Function
Phonebook Directory	Provides an audio phonebook that users can search through from their telephones. The phonebook is populated from the CompletePBX Web interface. Users can

Module	Function
	search by name and are given the associated phone number when their search matches an entry.
Speed Dial Functions	Works in conjunction with the Phonebook Directory module. Entries in the phonebook can be assigned short speed dial codes that users can dial. CompletePBX will then dial the associated full number of the phonebook entry and connect the call with the dialing user.

Inbound call control

Module	Function
Announcement Targets	Plays back a selected audio recording and then routes the call to another call target.
Blacklist	Provides the ability to build a list of numbers that are not allowed to call the system. Callers that are listed in the blacklist will hear a prompt informing them that the number they called is not in service.
Follow Me	Provides the ability to create a personalized ring group for each user on the system. A set of phone numbers is configured to ring when the user's extension is rung, and a fail over destination is specified.
IVR	Provides digital receptionist functionality. The IVR module answers a call, plays back a sound clip, and waits for the caller to push digits on their phone in order to route the call.
Queue Priorities	Provides the ability to give a particular caller higher priority when entering a queue. Callers with higher priority will be placed at the front of the queue and have their calls answered faster.
Queues	Allows callers to be routed in a call queue where they wait on hold until someone is available to answer their call. Callers are placed into the queue in the order they called in (unless their call order is altered by a queue priority).
Ring Groups	Creates a call target that will ring a predefined set of extensions. Extensions can either be rung all at the same time or in various hunt sequences that ring one extension at a time.
Time Conditions	Provides the ability to route calls to one of two call targets depending on the day of the week, month, date, or time.

Internal options and configuration

Module	Function
Call Forward	Allows users to set up call forwarding from their extension to another extension by dialing a feature code.
Call Waiting	Allows users to toggle call waiting by dialing a feature code. When enabled, the user's phone will give an audio or visual indication that another call is coming in, if the user is already on the phone.
Callback	Allows a trigger to be configured that calls back a specific number and provides the callee with access to a module or application. The callback number can be predefined, or can be the number that triggered the callback. This is mostly used to provide a user's cell phone with a dial tone on the CompletePBX system, giving them access to long distance calling that is cheaper than dialing directly from their cell phone.
Conferences	Provides the ability to set up conference rooms in which two or more users can join together in a single call. Conference rooms can be password protected.
DISA	Provides Direct Inward System Access (DISA) as a call target. This will give the caller a dial tone on the system allowing them to dial any of the features of the system, or initiate an outbound call through the trunks of the system.
Dictation	Allows a user to dial an extension, speak, and then have a recording of what they just said emailed to them.
Do-Not-Disturb (DND)	Allows a user to enable Do-Not-Disturb (DND) mode on their extension. If DND is enabled, calls to that extension will go straight to voicemail and the extension will not ring.
Info Services	Allows access to several information services through feature code for users. These services include: • Company directory • Call trace • Echo test • Speaking clock • Name current extension
Languages	Allows a call to be tagged with a specific language. The call can then be routed to a specific person or a set of people who speak the same language as the caller.

Module	Function
Misc Applications	Allows an administrator to configure arbitrary feature codes that direct the caller to a specific call target.
Misc Destinations	Allows an administrator to configure call targets that dial any local number (such as an extension, feature code, or external phone number). These destinations can then be set up as the target for inbound routes or as the failover destination for other call targets.
Music on Hold	Provides the ability to upload custom sound files or define streaming audio sources that will be played to callers while they are on hold.
PIN Sets	Allows an administrator to define a set of identification codes that can be used to grant access to outbound routes or conferences.
Paging and Intercom	Allows an administrator to define paging groups. When a paging group is called the phones in the group will automatically answer the call and play the caller's audio through their speakers. It also allows the creation of an intercom feature code that allows paging functionality to be directed to one phone specifically.
Parking Lot	Allows a call to be "parked". When a call is parked, the person parking the call is given a parking space number. Anyone who dials that number will be connected with the parked caller.
Recordings	Allows users to record, upload, and merge audio recordings. Recordings can be used in IVRs and as confirmation clips for ring groups.
Voicemail Blasting	Allows an administrator to create a group of extensions that can be "blasted" with a single voicemail message. When this functionality is invoked the caller records a message that is dropped into the voicemail box of all of the extensions that are members of the blast group.

Support

Module	Function
CompletePBX Logfiles	Allows an administrator to view the CompletePBX log files through the CompletePBX Web interface.
Online Support	

System administration

Module	Function
CompletePBX API	Provides the ability to create and edit users that can access the CompletePBX **My Extension** Web page.
CompletePBX CLI	Allows commands to be executed on the CompletePBX Command Line Interface (CompletePBX CLI) from the CompletePBX Web interface. CLI output is displayed in the Web interface.
CompletePBX Info	Provides the following information from CompletePBX: • Summary • Active SIP and IAX2 channels • Number of SIP and IAX2 registrations • Number of online and offline SIP and IAX2 peers • Registries • Active SIP and IAX2 registrations (including registration host, username, refresh interval, and registration state) • Channels • Active channel summary • Active number of SIP and IAX2 channels • Peers • Shows all of the defined SIP and IAX2 extensions/peers • SIP Info • Specific SIP registry and peer information • IAX Info • Specific IAX2 registry and peer information • Conferences • Shows details on conferences that are currently active • Subscriptions • Shows registered dial plan hints • Voicemail users • Shows a list of users that have a voicemail box and the number of waiting messages for those users • Full report • Shows a full listing of all of the information contained within the other information sections

Module	Function
Backup & Restore	Provides the administrator with the ability to set up one-time or recurring backups of all of the CompletePBX configuration data, recordings, and voicemails, and the ability to restore those backups.
Custom Applications	Allows customer applications and extensions to be registered with CompletePBX so that they can be accessed by other CompletePBX v1.1 applications. These applications would be written into the extensions_custom.conf file.
Phonebook Directory	Provides a phonebook that can be used as a lookup source for the Caller ID Lookup module. The phonebook is populated through the CompletePBX Web interface.
Weak Password Detection	Shows an administrator any user accounts, extensions, or trunks that have weak registration passwords. Accounts with weak passwords represent a security hole and should be updated as soon as possible.

Third-party Add-ons

Module	Function
Customer DB	If multiple companies are hosted on a single CompletePBX system, the Customer DB module provides a way to track which company owns a particular trunk or SIP device. This module is simply for tracking purposes and does not affect configuration or call routing.
Gabcast	Provides integration with a third-party podcasting service—Gabcast. The Gabcast module adds the Gabcast service as an available call target for inbound routes or fail over destinations.
Inventory	Provides an inventory of all of the employees within a given company. The Inventory module provides places to enter the MAC address, IP address, and serial number of an employee's IP phone. This module is simply for tracking purposes and does not affect configuration or call routing.
Print Extensions	Provides a printable list of all of the configured extensions, conferences, queues, and feature codes.

15. Voicemail.conf Options

When setting up an extension (Extensions mode) or a user (DeviceAndUser mode), CompletePBX provides a field called **VM Options** under the **Voicemail & Directory** group of options. This field allows any option to be entered that could be included as a mailbox option in */etc/asterisk/voicemail.conf*. These options affect the behavior of a mailbox and the way the voicemail messages are received and processed. Options should be listed using the syntax *optionname=value* (where *optionname* is replaced with the name of the option being configured, and *value* is replaced with a valid value for that option). Multiple options can be specified by placing a pipe character (/) between each option (for example,

attach=yes|attachfmt=gsm).

Option	Valid values	Purpose
attach	yes, no	If voicemail to email notifications are enabled, the attach option allows a copy of the voicemail to be attached to the email notification. If this option is not configured, the default value of "no" will be used.
		Note that this option can be configured within the CompletePBX interface using the Email Attachment field and should not be manually configured. Manually configuring this option could result in conflicting configurations and unpredictable result.
attachfmt	gsm, wav, wav49	Specifies the format that the sound file attached to the notification email should be in. GSM files are raw GSM encoded, WAV files are 16-bit linear WAV encoded, and WAV49 files are GSM encoded WAV format files. Windows PCs can play back WAV files by default, and most stock Linux desktop PCs should be able to play back GSM files. Any operating system should be able to play any of these formats, but additional software may be required. If this option is not configured, the default value of "wav" will be used.

Option	Valid values	Purpose
callback	(Any valid context)	The CompletePBX voicemail applications provide the ability for a user to call back the telephone number or extension number of the person who left them a voicemail. The callback option configures the context that CompletePBX will use for placing the outbound call. By default, users created under CompletePBX use the from-internal context to place their calls, and this option can be safely set to this value.
		The only time this option typically differs from the context that the user normally utilizes is when a user's outbound calling ability is restricted. If a user can typically call local telephone numbers, but not long distance, they would be unable to return a voicemail from anyone who is not within the local calling area. It is prudent for a company to allow callbacks to someone who has left a message. A company may choose to create a special context specifically for the purpose of voicemail callbacks in which calls to long distance numbers are allowed. In this case, the callback option would be set to the name of the custom context that allows long distance calling.
		If this option is not configured, no default value will be used and the callback feature of the voicemail application will be disabled.
delete	yes, no	If voicemail to email notifications are enabled, the delete option provides the ability to delete voicemail messages from the CompletePBX server as soon as the notification email is sent. This option is intended to be used by users who wish to receive their voicemails through email only. If this option is not configured, the default value of "no" will be used by CompletePBX.
		Note that this option can be configured within the CompletePBX interface using the Delete Voicemail field and should not be manually configured. Manually configuring this option could result in conflicting configurations and unpredictable result.
envelope	yes, no	If set to "yes", the user will hear the date and time a message was left prior to the message being played when they check their messages.

Option	Valid values	Purpose
forcegreetings	yes, no	If set to "yes", new users will be prompted to record their busy and unavailable outgoing messages the first time they log into their mailbox. The voicemail application determines that a user is new if their voicemail password is the same as their extension. If this option is not configured, the default value of "no" will be used.
forcename	yes, no	If set to "yes", new users will be prompted to record their name the first time they log into their mailbox. The voicemail application determines that a user is new if their voicemail password is the same as their extension. If this option is not configured, the default value of "no" will be used.
hidefromdir	yes, no	If set to "yes", the user will be hidden from the company directory. If this option is not configured, the default value of "no" will be used.
imapuser/imappassword	(IMAP account credentials)	The voicemail application allows voicemail messages to be stored on an IMAP server. Storing messages on an IMAP server allows voicemail to be managed either through a phone or an email client. Changes made using one management method are reflected in the other. For IMAP storage to work, the file */etc/asterisk/vm_general.inc* must be edited in order to include a line that reads imapserver=servername (where servername is replaced with the actual hostname of the IMAP server being used). Once the imapserver configuration option is defined in the vm_general.inc file, the imapuser and imappassword options must be configured to use valid user account credentials for the IMAP server. If these options are not configured, there are no default values used, and IMAP voicemail storage will not be used.
maxmsg	(whole number)	Defines the maximum number of messages that the user can have in each one of their voicemail folders. If this option is not configured, then there is no default value used and no limit will be placed on the amount of voicemail messages a user can store.

Option	Valid values	Purpose
operator	yes, no	If set to "yes", the person leaving the message is able to press *0* to return to the call target they came from before dialing an extension. For example, if a caller is directed to an IVR and then dials a user's extension and receives voicemail, they can press *0* to return to the IVR. If the caller did not come from a previous call target (they were routed directly to the user's extension), pressing *0* will route their call to the extension defined in the **Operator Extension** field on the CompletePBX **General Settings** page. If this option is not configured, the default value of "no" will be used.
review	yes, no	If set to "yes", the person leaving the message is able to press the pound (#) key after leaving their message in order to review the message they just recorded. They are given the option to re-record their message if they choose to. If this option is not configured, the default value of "no" will be used.
sayid	yes, no	If set to "yes", the user will hear the caller ID number of the person who left the voicemail prior to the voicemail being played when they check their messages. If this option is not configured, the default value of "no" will be used. Note that this option can be configured within the CompletePBX interface using the **Play CID** field and should not be manually configured. Manually configuring this option could result in conflicting configurations and unpredictable results.
sayduration	yes, no	If set to "yes", the user will hear the duration of the voicemail message prior to the voicemail being played when they check their messages. If this option is not configured, the default value of "no" will be used.

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Option	Valid values	Purpose
saydurationm	(Minimum time, in whole number of minutes)	This option restricts when the voicemail application will say the duration of a message. If this value is set, then the voicemail application will only announce the duration of a message if the message is longer than the value of saydurationm. For example, if this option is set to "2", the application will only announce the duration of messages that are longer than two minutes. This option only takes effect when the sayduration option is set to "yes". If this option is not configured, no default value will be used and all message durations will be announced when the sayduration option is set to "yes".
sendvoicemail	yes, no	If set to "yes", the user is able to forward a voicemail message they have received to another user. Otherwise, this functionality is disabled. If this option is not configured, the default value of "no" will be used.
serveremail	(Email address)	This option sets the email address from which voicemail to email notification messages appear to come. For example, this might be voicemail@example.com.
tempgreetwarn	yes, no	If set to "yes" and the user has a temporary greeting enabled, they will be notified that the greeting is enabled each time they check their voicemail messages. If this option is not configured, the default value of "no" will be used.
tz	(Valid timezone)	This option specifies which time zone a user is in, so that the envelope time data is played in the user's local time instead of the server's local time. This option must be a valid time zone. In order to find the valid time zones on the CompletePBX server, check the contents of the /usr/share/zoneinfo folder. The files in this folder are valid time zones; the subfolders contain country-specific time zones. If this option is not configured, the default time zone will be the local time zone of the CompletePBX server.

Option	Valid values	Purpose
volgain	Positive or negative numbers, rounded to one decimal place)	Specifies the gain that should be used when recording messages for the user. If messages are generally too quiet, the value should be higher (for example, 1.5). If messages are too loud, the number should be lower (for example, -2.1). If this option is not configured, the default value of "0" will be used and no gain will be applied to voicemail messages.

16. Trunk Configurations

The trunk configuration settings for common VoIP providers are as follows. Simply replace the required information tokens in the following configurations with their appropriate values to configure a fully working trunk.

Required information tokens that may be used are listed as follows:

Token	Replace with
(Name)	The name of the owner of the trunk. This might be a company's name or a person's name. Letters, numbers, and spaces are allowed. No other special characters may be used.
(TelNumber)	The telephone number associated with the trunk that is being configured. For PSTN trunks, this will be a number in the format of the local calling area (for example, nine digits for North America). Trunks to other VoIP providers may have different formats. For example, the **Free World Dialup (FWD)** network uses six-digit telephone numbers.
(Username)	The username or user ID assigned by the VoIP provider. This is usually a numeric user ID.
(Password)	The password or "secret" assigned by the VoIP provider.
(MaxChannels)	The maximum number of active channels that the provider can support. Some providers have a set limit (in which case that limit will be listed). Some providers allow the purchase of additional channels for which a monthly fee is charged (in this case, the (MaxChannels) token is used).
(FailureScript)	The script that is used to monitor trunk failures, if one exists. See the Monitoring trunk status with CompletePBX failure scripts section of Chapter 3, **Trunks** for sample monitoring scripts.
(VoIPHost)	The SIP or IAX2 server supplied by the provider. This might be referred to as the "registration server" or simply the "host" by the provider.