Racknine VoIP Telephones

A startup guide to your new telephone service

Welcome

Thank you for choosing Racknine VoiP hosted PBX telephone services. We have prepared the following guide to familiarize you with your new service and to answer common questions. Your phone service is highly customizable, though, so we encourage you to contact us with any specific requests you have.

What is a PBX?

PBX stands for Private Branch Exchange. It is a telephone system that switches calls between private users on local extensions.

What does it mean that our PBX is hosted?

Traditional PBX services depend on equipment located on the customer's premises. This means that your telephones require a connection to this equipment on site.

With Racknine's hosted service the PBX is not located on site. It is located at Racknine and accessed through the internet. This means that telephones are not tethered to your site. You can connect your phones anywhere there's an internet connection.   Operate from home just as you would in the office! Transfer a phone call from your office in Edmonton to someone in your Calgary office just as you would if they were in the same building!

The Aastra 6757i VoIP Telephone

Idle Screen

Lot buttons
When a call is Parked it will go to the next available Lot position. The call can then be retrieved from any other extension.

Intercom
Page all other extensions. Paging groups can be customized.

DND
Do Not Disturb. Ignores incoming calls and sends them to voicemail.

Callers
Shows a list of the most recent 200 callers. Use up & down arrow to cycle.

Hangup
Ends current call. Also acts as an escape button when navigating menus.

Options
Options menu where you can modify settings such as call forwarding and phone preferences.

Redial
Most recent list of callees. Cycle with up&down arrow. Press dial to call.

Named Extension buttons
Shortcuts to internal extensions in your organization.

More
Cycles to next page of top on-screen keys. The top keys can support up to 10 menu options (or 2 pages.)

Extension Name
And extension number

Voicemail
Shortcut to extension voicemail

Directory
Custom directory. Use up & down arrows to cycle through or enter first few letters of the contact's name with the number pad.

Lines
Place a call on hold and place a new call on another line. PLEASE NOTE - While functional, the line buttons are generally not used. Park & Lot functions are used instead the majority of the time.

*** On-screen controls are customizable so yours may vary.**

The Aastra 6757i VoIP Telephone

Ring Screen

Ring Indicator & Call Display

Displays information about caller.

Answer

Accept incoming call to your extension.

Ignore

Ignores incoming call and puts extension back to idle. Call will continue to ring on other extensions.

Activity Indicator

Will flash during incoming call.

* On-screen controls are customizable so yours may vary.

The Aastra 6757i VoIP Telephone

During Active Call

Ring Indicator & Call Display
Displays information about caller.

Drop
Terminates current call.

Conf
Conference. Places active call on hold and allows you to call another participant to connect to a conference call.

Xfer
Transfer. Allows you to transfer call to another extension within your organization or to an external line.

Activity Indicator
Will be solid red during an active call.

Call Timer
Displays duration of current active call.

Park
Places call on hold. Call will go to next available “Lot” position, described previously. Corresponding Lot button will light up. Press Lot button to retrieve call.

*** On-screen controls are customizable so yours may vary.**

Basic Functions

How to place a call

Dial external number or extension number. Push dial. If you would like to use the handset, pick it up at this time. If you would prefer to use speaker, leave handset on cradle. Press Drop or Hangup to terminate call.

How to receive a call

If you prefer to use the handset, pick it up to receive call. If you prefer the speaker, press the answer key to receive call. Call will default to Line 1 on your phone. Press Drop or Hangup to terminate call.

How to place a call on hold

Press the Park button. The call will enter the next available parking lot position and the corresponding Lot key will illuminate red. To retrieve the call from any extension, press the corresponding lot key.

How to transfer a call

Press the Xfer key. You can then enter the phone number for an external line or an internal extension number. You may also press a named extension shortcut button if applicable or select a contact from your directory. To complete the transfer, press the Xfer key again.

How to add a call to a conference

You may add another participant to an active call by pressing the Conf key. You can then enter the phone number for an external line or an internal extension number. You may also press an extension shortcut button if applicable or select a contact from your directory. To complete the process, press dial. Wait until the participant answer, then press Conf again.

How to page/intercom a single extension

*80 + the extension number (eg *80101.) If using speaker, you may press drop or hangup to terminate page. Otherwise you can place handset back in cradle to end page. If paging a certain extension is common for you, a shortcut button can be made.

How to page/intercom all phones

Press Icom key. If using speaker, you may press drop or hangup to terminate page. Otherwise you can place handset back in cradle to end page. Custom paging groups can be configured for you so only certain phones receive a page.

How to use the Directory

Press Directory. Use up & down arrow keys to navigate or use your number pad to enter the first few letters of your contact. By default your directory will be empty. To populate your directory, we require that you send us your directory information. Further information on this later.

Do Not Disturb

Press the DND key to enter Do-Not-Disturb mode. While DND is engaged, the red LED beside DND will illuminate. Incoming calls will not ring to your extension during this time but will continue to ring all other extensions. Press DND again to deactivate.

How to use the Callers and Redial lists

Press Callers or Redial. Use the up & down arrow keys to cycle through a list of the most recent callers/callees. Press Dial to call.

Directory

How to submit your desired directory table to Racknine.

To submit your directory to Racknine so it can be accessed from your phones you need to enter the data into a spreadsheet. The spreadsheet must follow the following schema:

Contact Name	Number	Line	Company
Contact Name	Number	Line	Company
Contact Name	Number	Line	Company

Please note that the “Line” will always be 1. Also, please do not include spaces or dashes in the number. For example:

Bill Gates	7805551234	1	Microsoft
Steve Jobs	4035551234	1	Apple
Jane Doe	105	1	MyCompany

Notice that the contact “Jane” has an internal extension number. You can include internal organization contacts in your directory.

Once you have completed your directory spreadsheet, save it as a .csv file (important) and email it to mmeier@racknine.net or dkeiver@incentre.net. We will update your directory and it should then be accessible from your phones.

Voicemail

Using your voicemail for the first time

Most phones will have a Voicemail shortcut button. Press that to access voicemail. If you do not have a shortcut key you may dial *97. You also can access your voicemail from a different extension by dialing *98 + your ext number. (eg *98101) Upon accessing your voicemail you will be prompted for your password. The default password on your mailbox will be 2222.

Change your password

Before doing anything else you will want to change your password. To do so press 0 from the main menu to access mailbox options. Then press 5 and follow the audio instructions.

Record your greetings

There are four types of greetings you can record: Busy, Unavailable, Name, and Temporary. The first greeting you will want to record is your Unavailable greeting. The rest are optional. To do so press 0 from the main menu to access mailbox options. Then press 1 and follow the audio instructions.

You can also record a Busy greeting. (From main menu, 0 and then 2.) If a call gets transferred to your extension while you are already on an active call, the caller will hear your Busy greeting instead of your Unavailable greeting. The Busy greeting is optional, however. If you don't record one, callers will hear your Unavailable greeting instead.

Name Greeting

The Name greeting (from main menu, 0 and then 3) is simply your recorded name. If you have an IVR with a dial-by-name directory for your callers, your Name greeting is what they will hear when the dial your extension.

Temporary Greeting

The temporary greeting (from main menu, 0 and then 4) overrides the Busy and Unavailable greetings. It's meant to be used when you need a new greeting for a short time but don't want to lose your regular greetings. Ideal for things like vacations.

How to know when you have voicemail waiting

Your activity light will blink red and there will be a mail icon on your screen.

Listening to your messages

From the main menu press 1 to hear your new messages. For each message you will have the option to press 5 to repeat it, 7 to delete, 8 to forward to another extension/user, and 9 to save.

When you save a message you will be able to save it into a number of folders. New messages, Old messages, Work messages, and Family messages. To listen to saved messages again select 2 from the main menu and listen to the audio instructions to select the relevant folder.

Voicemail to Email

You also have the option to have voicemails sent directly to your email instead of on your phone. To enable this, please have your company director to contact us with the request.

Feature Codes

*30 Blacklist a number

*32 Blacklist the last caller

*31 Remove a number from the blacklist

*21 Findme Follow Toggle

*69 Call Trace

*43 Echo Test

*65 Speak Your Exten Number

*60 Speaking Clock

*80 Intercom prefix

*54 User Intercom Allow

*55 User Intercom Disallow

*85 Pickup Parked Call Prefix

*99 Check Recording

*77 Save Recording

*98 Dial Voicemail

*97 My Voicemail

Advanced Features

The following pages deal with the FreePBX administration site for your PBX phone server. Changes to your server configuration can result in a loss of phone service. The following information is for technical staff only. Please contact Rack9/Internet Centre for your PBX server host name and login credentials.

Backup and Restore

BACKUP

Before making any changes to your PBX configuration it is recommended you backup your current config.

- Under “Admin” click “Backup & Restore.”
- Click “New Backup”
- Enter a name for your backup
- Enter a description of your backup (optional)
- Enter an email address. The PBX will send notifications of backup status here
- Drag each item in “Templates” to the “Backup Items” column to the left
- You can skip “Hooks” and “Backup Server”
- Under Storage Locations, drag “Local Storage” from the Available Servers column on the right to the Storage Servers column on the left
- You can skip the remaining few options
- Click Save
- Click on the name of your backup on the far right
- Scroll all the way to the bottom and click “and run”

RESTORE

To restore a previous configuration backup please contact Racknine / The Internet Centre.

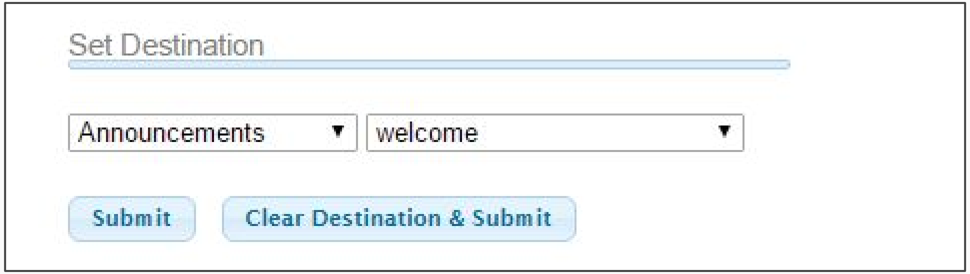
dkeiver@incentre.net
mmeier@racknine.ca
780-450-6787

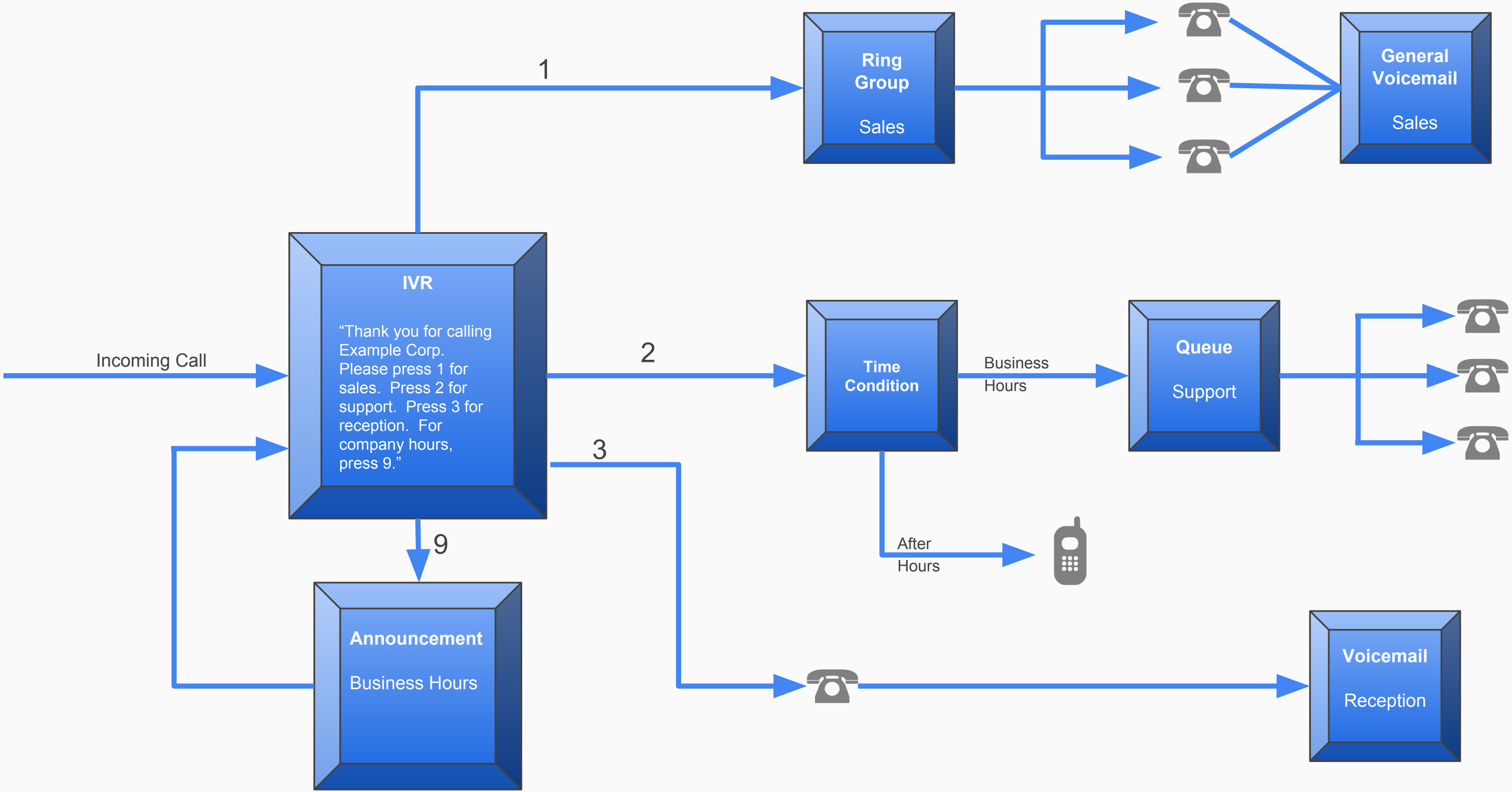
Call Routing

All incoming calls are routed to things like announcements, IVR, ring groups, extensions, etc. It is common to have calls routed to multiple services. For example, someone might have incoming calls going to an Announcement with company information, then to a ring group containing everyone's extension, then to an IVR if no one in the ring group picks up. The IVR then may direct the call to a number of destinations.

The starting point for call routing is "Inbound Routes." (Connectivity - Inbound Routes.) On the right hand side of the screen will be a list of the phone numbers the PBX handles. Click on a phone number and then scroll to the bottom. You will see under "Set Destination" where the call is routed to.

Example:





EXAMPLE CALL ROUTING

System Recordings

One of the most common things users want to do is record messages for things like Announcements or IVR (Interactive Voice Response.)

“System Recordings” is found under the “Admin” menu.

Record on phone.

This is the simplest way to make a system recording.

- (On web interface) Enter the extension# of the phone you wish to make the recording on and click Go.
- (On phone) Dial *77 and wait for the beep. Record your message. Press # when done and follow the instructions on the phone.
- (On web interface) Type name of system recording and click Save.
- Your recording will appear on the right hand side of the screen. You can click on it to add a description, review, or delete it.

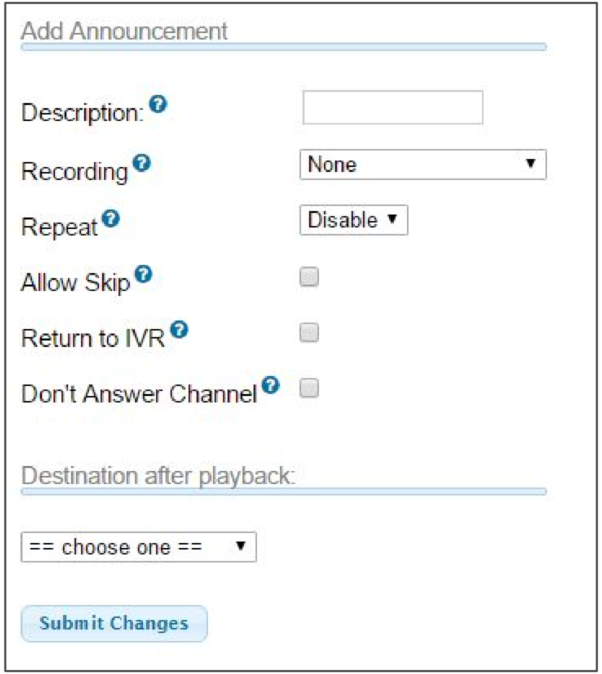
Upload .WAV file.

You may upload a .wav file you recorded yourself. To do so simply click “Choose File.” Navigate to the .wav you wish to upload, then click “Upload.” Your .wav file MUST BE PCM encoded at 16 bits and 8000 hz. Once uploaded, your recording will appear on the right hand side of the screen. You can click on it to add a description, review, or delete it.

Announcements

Use Announcements to play system recordings for incoming callers and then direct the call to another service, group, or extension. (See System Recordings)

- Applications - Announcements
- Enter description of announcement (example, “business hours”)
- From the Recording drop-down menu, select a previously uploaded/recorded System Recording
- If desired, you can have the recording repeat.
- Check “allow skip” if you would like callers to be able to skip the recording by pressing a key
- If callers are being directed to this announcement from an IVR you have the option of returning the caller to the origin IVR. Check the Return to IVR box to do so. This will override the Destination option.
- If “Return to IVR” does not apply, then set a destination for callers to be routed to after announcement is complete.
- Click “Submit Changes” and then the red “Apply Config” button on top.



IVR Menu 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 Announcement on Invalid

Return on Invalid

Invalid Recording: Default

Invalid Destination: == choose one ==

Timeout Retries: 3

Timeout Retry Recording: Default

Append Announcement on Timeout

Return on Timeout

Timeout Recording: Default

Timeout Destination: == choose one ==

Return to IVR after VM

- IVR Entries

Ext	Destination	Return	Delete
digits pressed	== choose one ==		
digits pressed	== choose one ==		
digits pressed	== choose one ==		
digits pressed	== choose one ==		
digits pressed	== choose one ==		

Submit

IVR

IVR (Interactive Voice Response) is a very common feature of PBX systems. It is the interactive voice menu present on my business phone systems. To create your own IVR click Applications - IVR. Refer to the screenshots on the previous page.

- IVR Name - Create a name for your IVR
- IVR Description - Optional, describe your IVR
- Announcement - Specify the system recording your IVR will play to the caller. (See "System Recordings")
- Direct Dial - If you want callers to be able to dial extensions directly from your IVR, select "Extensions" here. Otherwise leave at disabled
- Timeout - The amount of time, in seconds, the IVR will wait for a response from the caller
- Invalid Retries - The number of times the IVR will permit an invalid response from a caller
- Invalid Retry Recording - The audio prompt the IVR will play to the caller when they select an invalid option. By default the system will use pre-recorded prompts but you can select a system recording you created yourself
- Append Announcement on Invalid - If enabled, the IVR will replay your IVR announcement each time your caller selects an invalid option
- Return on Invalid - Applies only if caller was routed here from another IVR. If so, and this option is enabled, the caller will be returned to the previous IVR if they select an invalid option
- Invalid Recording - The audio message played to the caller after they've exceeded the number of invalid retries permitted
- Invalid Destination - Where the call is routed after the caller exceeds the number of invalid retries permitted
- Timeout Retries - The number of times the IVR will wait for a response from the caller
- Timeout Retry Recording - The audio prompt the IVR will play to the caller when they exceed the allowed Timeout time without pressing a key.. By default the system will use pre-recorded prompts
- Append Announcement on Timeout - If enabled, the IVR will replay your IVR announcement each time your caller exceeds the timeout time without pressing a key
- Timeout Recording - The audio message played to the caller after they've exceeded the number of timeout retries permitted
- Where the call is routed after the caller exceeds the number of timeout retries permitted

IVR Entries

Here is where you set your menu options and destinations.
Click the green plus icon to add more fields or the trash can to delete select fields.

Under Ext enter a number key option. Choose the destination beside it. For example, you may want callers to be able to "Press 1 for reception." Enter 1 in Ext and set the Destination to Extension (after which it will prompt you to specify who's extension, reception in this case.)

If you check the "Return" box, it will return the caller to a previous IVR, if applicable, when that number is pressed. This option overrides the destination box.

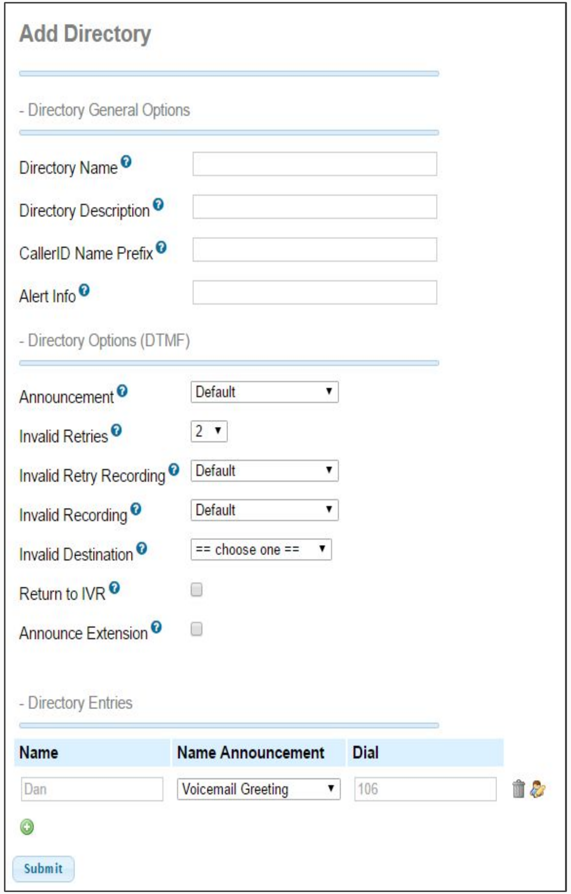
When you're finished creating your IVR, click "Submit" and then the red "Apply Changes" button.

Directory

The Directory is used (typically implemented as an IVR option) to provide callers with a dial-by-name directory of your staff. To add a Directory, click Applications - Directory.

- **Directory Name** - Name of this directory.
- **Directory Description** - Description of this directory.
- **Caller ID Name Prefix** - You can optionally prefix the Caller ID name when callers are dialing from the Directory. ie: If you prefix with "Sales:", a call from John Doe would display as "Sales:John Doe" on the extensions that ring.
- **Announcement** - Greeting to be played on entry to the directory. See System Recordings.
- **Invalid Retries** - Number of times to retry when receiving an invalid/unmatched response from the caller.
- **Invalid Retry Recording** - Prompt to be played when an invalid/unmatched response is received, before prompting the caller to try again. By default a pre-recorded system message is used. You can alternatively use a System Recording you created yourself.
- **Invalid Recording** - Prompt to be played before sending the caller to an alternate destination due to the caller pressing 0 or receiving the maximum amount of invalid/unmatched responses (as determined by Invalid Retries.) By default a pre-recorded system message is used. You can alternatively use a System Recording you created yourself.
- **Invalid Destination** - Destination to send the call to after Invalid Recording is played.
- **Return to IVR** - When selected, if the call passed through an IVR that had "Return to IVR" selected, the call will be returned there instead of the Invalid destination.
- **Announce Extension** - When checked, the extension number being transferred to will be announced prior to the transfer.
- **Directory Entries** - This is where you define which of your extensions callers can "look up" through the directory. Click the green plus to add an extension. Select an extension from the list presented. Select the Name Announcement. By default it will be "Voicemail Greeting" which will play the voicemail NAME greeting the user recorded when setting up their Voicemail. If they haven't recorded a NAME greeting, you can optionally select Text-To-Speech or Spell Name. The Dial field is optional. By default it will contain the number of the extension you selected, but you can choose an alternate number.

When finished, click "Submit" and then the red "Apply Changes" button.



Time Groups and Time Conditions

Time Groups

Time groups are simply scheduled times, defined by you, that are used by Time Conditions. To define a Time Group click on Applications - Time Groups. Enter a name for your Time Group (Example - “Business Hours”) and set the desired time criteria underneath. When done, click “Submit” and then the red “Apply Changes” button.

Time Conditions

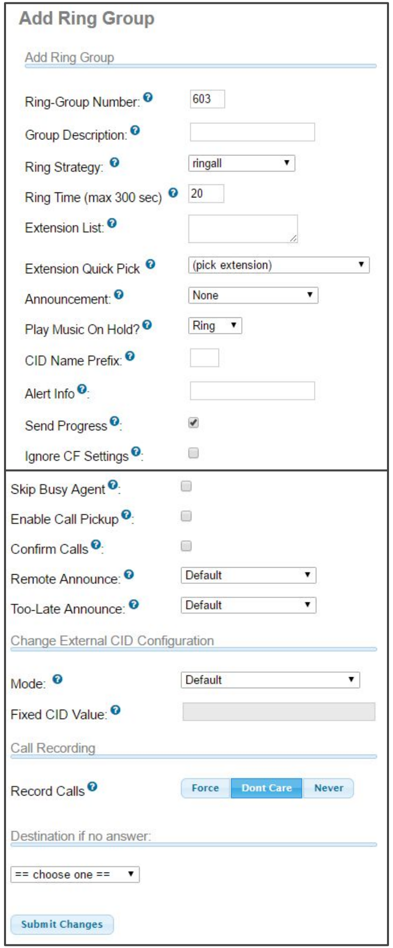
Time conditions are used when you want your PBX to behave different ways at different times of day. For example, you may wish incoming calls to ring in your office during the day and then to an after-hours cell phone when your office is closed. To do this, click on Applications - Time Conditions. Enter a name for your Time Condition. Select the Time Group you created (on the left of this page.) Set the destination to send the call if the time matches your time group and set the destination for when the time doesn’t match. When done, click “Submit” and then the red “Apply Changes” button.

Ring Groups

Ring groups are used to group extensions so you can route callers to them all. To create a ring group click on Applications - Ring Group.

- **Ring-Group Number** - This is the extension number of the ring group. Dialing this extension or routing calls to this ring group number will ring every extension in the ring group
- **Group Description** - Optional
- **Ring Strategy** - Defines which extensions inside the ring group ring.
  - **Ringall**: Ring all available channels until one answers
  - **Hunt**: Take turns ringing each available extension
  - **Memory Hunt**: Ring first extension in the list, then ring the 1st and 2nd extension, then ring 1st 2nd and 3rd extension in the list.... Etc
  - ***-prim**: These modes act as described above. However, if the primary extension (first in list) is occupied, the other extensions will not be rung. If the primary is FreePBX DND, it won't be rung. If the primary is FreePBX CF unconditional, then all will be rung
  - **Firstavailable**: ring only the first available channel
  - **Firstnotonphone**: ring only the first channel which is not offhook - ignore CW
  - **Random**: Makes a call could hop between the included extensions without a predefined priority to ensure that calls in the groups are (almost) evenly spread. Simulates a Queue when a Queue can not otherwise be used.
- **Ring Time** - Time in seconds that the phones will ring. For all hunt style ring strategies, this is the time for each iteration of phone(s) that are rung
- **Extension List** - List extensions to ring, one per line. You can include an extension on a remote system, or an external number by suffixing a number with a '#'. ex: 7804506787# would dial 7804506787 externally. Extensions without a '#' will not ring a user's Follow-Me. To dial Follow-Me, Queues and other numbers that are not extensions, put a '#' at the end
- **Extension quick pick** - Choose an extension to append to the end of the extension list above.
- **Announcement** - Message to be played to the caller before dialing this group. (See System Recordings)
- **Play Music On Hold?** - If you select a Music on Hold class to play, instead of 'Ring', they will hear that instead of Ringing while they are waiting for someone to pick up.
- **CID Name Prefix** - You can optionally prefix the CallerID name when ringing extensions in this group. ie: If you prefix with "Sales:", a call from John Doe would display as "Sales:John Doe" on the extensions that ring.
- **Confirm Calls -** Enable this if you're calling external numbers that need confirmation - eg, a mobile phone may go to voicemail which will pick up the call. Enabling this requires the remote side push 1 on their phone before the call is put through. This feature only works with the ringall ring strategy
- **Remote Announce** - Message to be played to the person RECEIVING the call, if 'Confirm Calls' is enabled. Default is pre-recorded system message, but you can also use System Recordings
- **Too-Late Announce** - Message to be played to the person RECEIVING the call, if the call has already been accepted before they push 1. Default is pre-recorded system message, but you can also use System Recordings
- **Record Calls** - You can always record calls that come into this ring group (Force), never record them (Never), or allow the extension that answers to do on-demand recording (Dont Care). If recording is denied then one-touch on demand recording will be blocked, unless they have the "Override" call recording setting.
- **Destination if no answer** - Where to direct the call if no one in the ring group answers the call. Mandatory.

When finished click “Submit Changes” and then the red “Apply Changes” button.

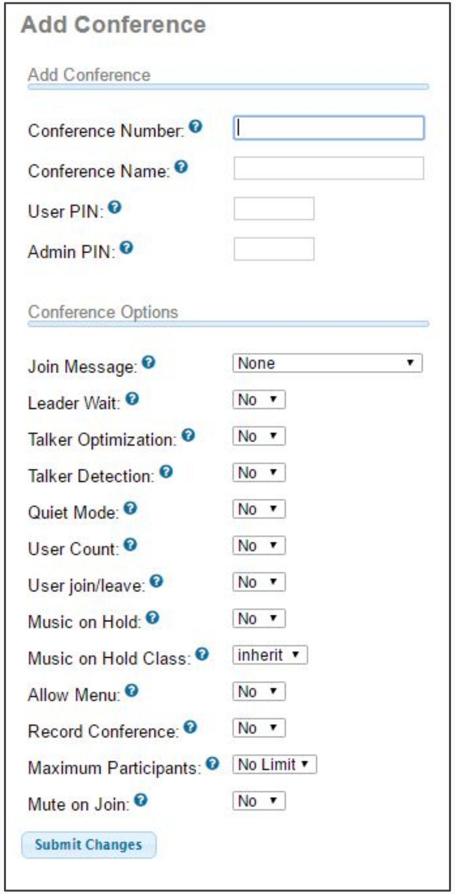


Conference

Set up conference calls by clicking Applications - Conference. Once your conference call is set up you can dial into it from local extensions or route calls to your conference from Inbound Routes, IVR, etc. Also, once a conference is created, it can be dialed into at any time.

- **Conference Number** - Use this number to dial into the conference.
- **Conference Name** - Give this conference a brief name to help you identify it.
- **User PIN** - You can require callers to enter a password before they can enter this conference. This setting is optional. If either PIN is entered, the user will be prompted to enter a PIN.
- **Admin PIN** - Enter a PIN number for the admin user. This setting is optional unless the 'leader wait' option is in use, then this PIN will identify the leader.
- **Join Message** - Message to be played to the caller before joining the conference.
- **Leader Wait** - Wait until the conference leader (admin user) arrives before starting the conference.
- **Talker Optimization** - Turns on talker optimization. With talker optimization, the PBX treats talkers who are not speaking as being muted, meaning that no encoding is done on transmission and that received audio that is not registered as talking is omitted, causing no buildup in background noise.
- **Quiet Mode** - Quiet mode (do not play enter/leave sounds)
- **User Count** - Announce user(s) count on joining conference
- **User join/leave** - Announce users joining or leaving the conference
- **Music on Hold** - Enable Music On Hold when the conference has a single caller
- **Record Conference** - Record the conference call. Recording will later be available in CDR Reports
- **Maximum Participants** - Maximum Number of users allowed to join this conference.
- **Mute on Join** - Mute everyone when they initially join the conference. Please note that if you do not have 'Leader Wait' set to yes you must have 'Allow Menu' set to Yes to unmute yourself

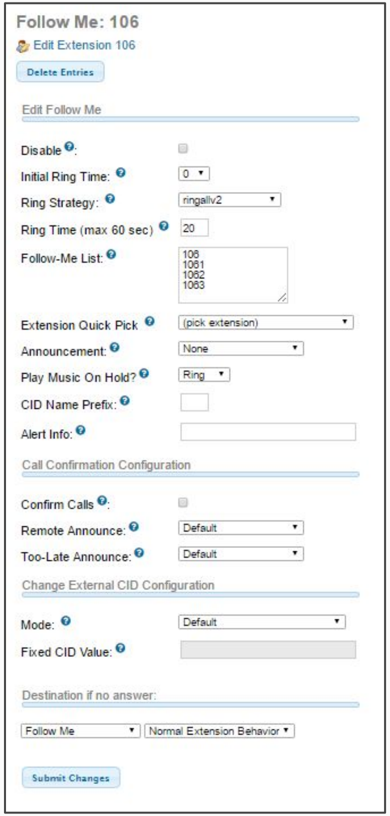
Once Conference setup is complete, click “Submit Changes” and then the red “Apply Changes” button.



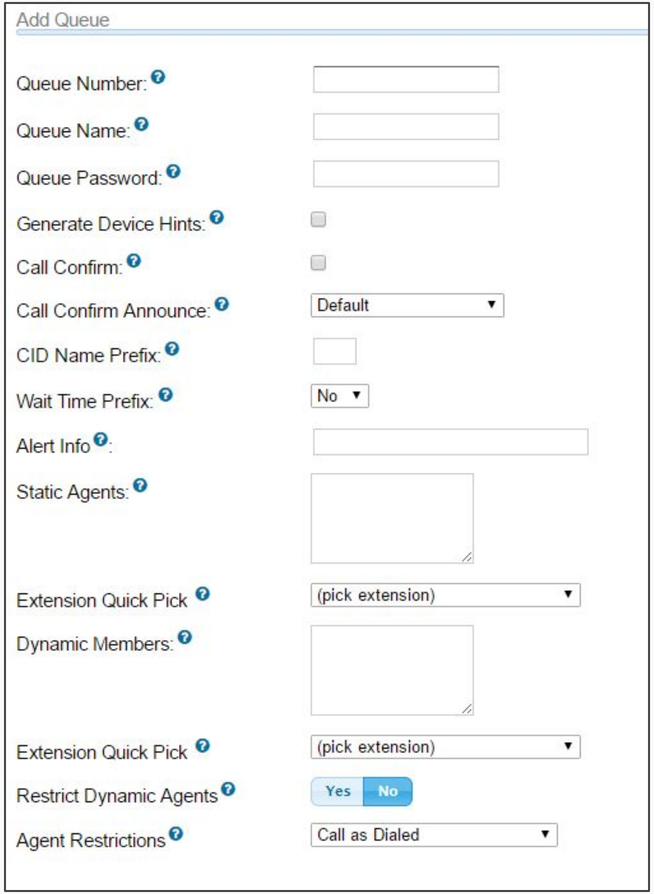
Follow Me

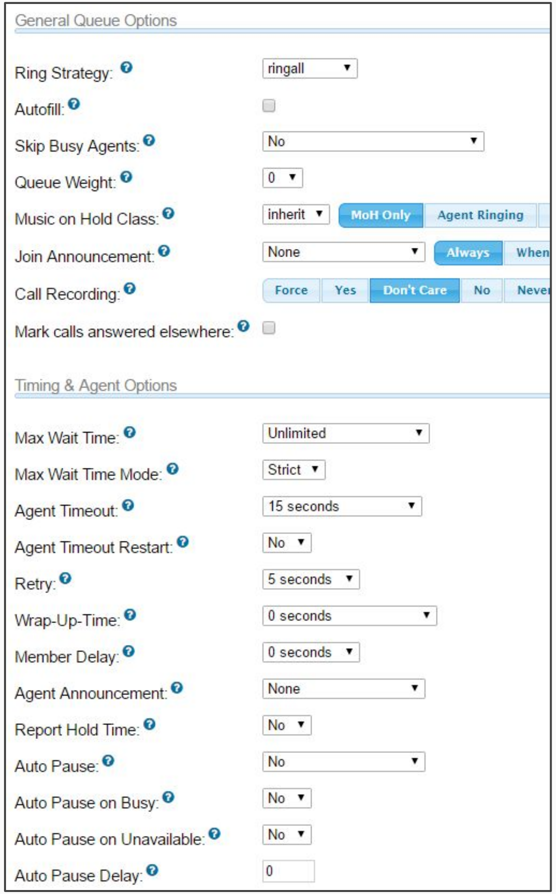
Follow Me is a method to associate multiple extensions to ring together. An example use for this would be to have a user's cell phone associate with their desk extension so their cell phone will right when their desk phone rings. Or you can set it so their cell phone will ring 5 seconds after their desk phone. To set up a Follow Me click on Applications - Follow Me. Then click the extension on the right that you wish to add a Follow Me on.

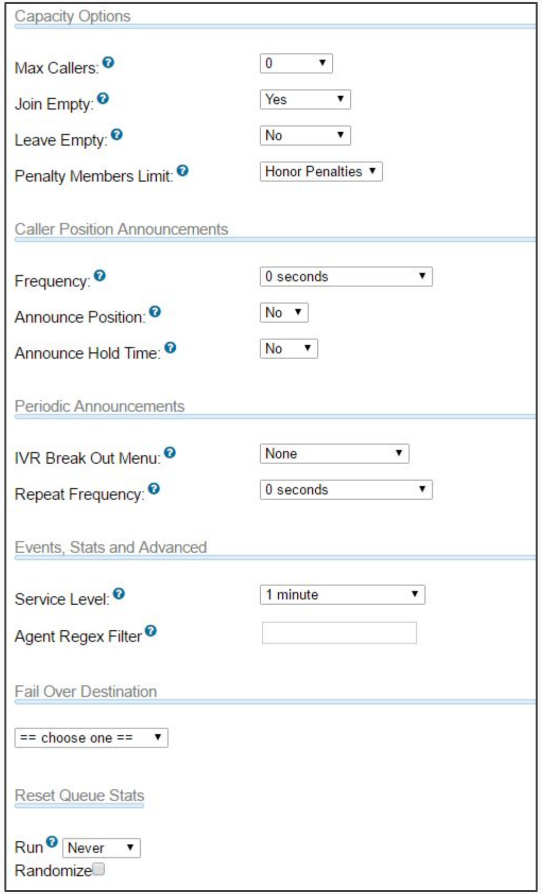
- **Disable** - By default (not checked) any call to this extension will go to this Follow-Me instead, including directory calls by name from IVRs. If checked, calls will go only to the extension. However, destinations that specify FollowMe will come here. Checking this box is often used in conjunction with VmX Locater, where you want a call to ring the extension, and then only if the caller chooses to find you do you want it to come here.
- **Initial Ring Time** - This is the number of seconds to ring the primary extension prior to proceeding to the follow-me list. The extension can also be included in the follow-me list. A 0 setting will bypass this.
- **Ring Strategy** -
  - **Ringallv2**: ring Extension for duration set in Initial Ring Time, and then, while continuing call to extension, ring Follow-Me List for duration set in Ring Time.
  - **Ringall**: ring Extension for duration set in Initial Ring Time, and then terminate call to Extension and ring Follow-Me List for duration set in Ring Time.
  - **Hunt**: take turns ringing each available extension.
  - **Memory Hunt**: ring first extension in the list, then ring the 1st and 2nd extension, then ring 1st 2nd and 3rd extension in the list.... Etc.
  - ***-prim:** these modes act as described above. However, if the primary extension (first in list) is occupied, the other extensions will not be rung. If the primary is FreePBX DND, it won't be rung. If the primary is FreePBX CF unconditional, then all will be rung.
  - **Firstavailable**: ring only the first available channel.
  - **Firstnotonphone**: ring only the first channel which is not off hook - ignore CW
- **Ring Time** - Time in seconds that the phones will ring. For all hunt style ring strategies, this is the time for each iteration of phone(s) that are rung
- **Follow-Me List** - List extensions to ring, one per line, or use the Extension Quick Pick below. You can include an extension on a remote system, or an external number by suffixing a number with a pound (#). ex: 7804506787#
- **Extension Quick Pick** - Choose an extension to append to the end of the extension list above.
- **Announcement**: Message to be played to the caller before dialing this group. (See System Recordings)
- **Play Music On Hold?** - If you select a Music on Hold class to play, instead of 'Ring', they will hear that instead of Ringing while they are waiting for someone to pick up.
- **CID Name Prefix** - You can optionally prefix the Caller ID name when ringing extensions in this group. ie: If you prefix with "Sales:", a call from John Doe would display as "Sales:John Doe" on the extensions that ring.
- **Confirm Calls** - Enable this if you're calling external numbers that need confirmation - eg, a mobile phone may go to voicemail which will pick up the call. Enabling this requires the remote side push 1 on their phone before the call is put through. This feature only works with the ringall/ringall-prim ring strategy
- **Remote Announce** - Message to be played to the person RECEIVING the call, if 'Confirm Calls' is enabled. See System Recordings.
- **Too-Late Announce** - Message to be played to the person RECEIVING the call, if the call has already been accepted before they push 1. See System Recordings.
- **Destination if No Answer** - Where the call is directed if no extension in this Follow Me is answered. Leaving this at default (Normal Extension Behavior) lets the main extension dictate the destination.



QUEUE PAGE SCREENSHOT







QUEUE

A queue is similar to a ring group. However, whereas a Ring Group will route a call elsewhere if its member extensions are busy, a queue will keep callers on hold until the next available extension is ready. Queues are found under Applications - Queues. To create a queue, refer to the screenshots on the previous page. Descriptions on this page.

- **Queue Number** - Use this number to dial into the queue, or transfer callers to this number to put them into the queue. Agents will dial this queue number plus * to log onto the queue, and this queue number plus ** to log out of the queue. For example, if the queue number is 123: 123* = log in 123** = log out.
- **Queue Name** - Give this queue a brief name to help you identify it.
- **Queue Password** - You can require agents to enter a password before they can log in to this queue. This setting is optional. The password is only used when logging in with the legacy queueno* code. When using the toggle codes, you must use the Restrict Dynamic Agents option in conjunction with the Dynamic Members list to control access.
- **Call Confirm** - If checked, any queue member that is actually an outside telephone number, or any extensions Follow-Me or call forwarding that are pursued and leave the PBX will be forced into Call Confirmation mode where the member must acknowledge the call before it is answered and delivered.
- **Call Confirm Announce** - Announcement played to the Queue Member announcing the Queue call and requesting confirmation prior to answering. If set to default, the standard call confirmation default message will be played unless the member is reached through a Follow-Me and there is an alternate message provided in the Follow-Me. This message will override any other message specified. To add additional recordings please use the "System Recordings" MENU.
- **CID Name Prefix** - You can optionally prefix the CallerID name of callers to the queue. ie: If you prefix with "Sales:", a call from John Doe would display as "Sales:John Doe" on the extensions that ring.
- **Wait Time Prefix** - When set to Yes, the CID Name will be prefixed with the total wait time in the queue so the answering agent is aware how long they have waited. It will be rounded to the nearest minute, in the form of Mnn: where nn is the number of minutes. If the call is subsequently transferred, the wait time will reflect the time since it first entered the queue or reset if the call is transferred to another queue with this feature set.
- **Static Agents** - Static agents are extensions that are assumed to always be on the queue. Static agents do not need to 'log in' to the queue, and cannot 'log out' of the queue. List extensions to ring, one per line. You can include an extension on a remote system, or an external number.
- **Extension Quick Pick** - Choose an extension to append to the end of the static agents list above.
- **Dynamic Members** - Dynamic Members are extensions or callback numbers that can log in and out of the queue. When a member logs in to a queue, their penalty in the queue will be as specified here. Extensions included here will NOT automatically be logged in to the queue.
- **Restrict Dynamic Agents** - Restrict dynamic queue member logins to only those listed in the Dynamic Members list above. When set to Yes, members not listed will be DENIED ACCESS to the queue.
- **Agent Restrictions** - When set to 'Call as Dialed' the queue will call an extension just as if the queue were another user. Any Follow-Me or Call Forward states active on the extension will result in the queue call following these call paths. This behavior has been the standard queue behavior on past FreePBX versions. When set to 'No Follow-Me or Call Forward', all agents that are extensions on the system will be limited to ringing their extensions only. Follow-Me and Call Forward settings will be ignored. Any other agent will be called as dialed. This behavior is similar to how extensions are dialed in ringgroups. When set to 'Extensions Only' the queue will dial Extensions as described for 'No Follow-Me or Call Forward'. Any other number entered for an agent that is NOT a valid extension will be ignored. No error checking is provided when entering a static agent or when logging on as a dynamic agent, the call will simply be blocked when the queue tries to call it.
- **Ring Strategy** -
  - **Ringall**: ring all available agents until one answers
  - **Leastrecent**: ring agent which was least recently called by this queue
  - **Fewest Calls**: ring the agent with fewest completed calls from this queue
  - **Random**: ring random agent
  - **RRMemory**: round robin with memory, remember where we left off last ring pass
  - **RROrdered**: same as rrmemory, except the queue member order from config file is preserved
  - **Linear**: rings agents in the order specified, for dynamic agents in the order they logged in
  - **Wrandom**: random using the member's penalty as a weighting factor, see asterisk documentation for specifics
- **Autofill** - if this is checked, and multiple agents are available, Asterisk will send one call to each waiting agent (depending on the ring strategy). Otherwise, it will hold all calls while it tries to find an agent for the top call in the queue making other calls wait.

QUEUE - Continued

- **Skip Busy Agents** - When set to 'Yes' agents who are on an occupied phone will be skipped as if the line were returning busy. This means that Call Waiting or multi-line phones will not be presented with the call and in the various hunt style ring strategies, the next agent will be attempted. When set to 'Yes + (ringinuse=no)' the queue configuration flag 'ringinuse=no' is set for this queue in addition to the phone's device status being monitored. This results in the queue tracking remote agents (agents who are a remote PSTN phone, called through Follow-Me, and other means) as well as PBX connected agents, so the queue will not attempt to send another call if they are already on a call from any queue. When set to 'Queue calls only (ringinuse=no)' the queue configuration flag 'ringinuse=no' is set for this queue also but the device status of locally connected agents is not monitored. The behavior is to limit an agent belonging to one or more queues to a single queue call. If they are occupied from other calls, such as outbound calls they initiated, the queue will consider them available and ring them since the device state is not monitored with this option. WARNING: When using the settings that set the 'ringinuse=no' flag, there is a NEGATIVE side effect. An agent who transfers a queue call will remain unavailable by any queue until that call is terminated as the call still appears as 'inuse' to the queue UNLESS 'Agent Restrictions' is set to 'Extensions Only'.
- **Queue Weight** - Gives queues a 'weight' option, to ensure calls waiting in a higher priority queue will deliver its calls first if there are agents common to both queues.
- **Music On Hold Class** - Music (MoH) played to the caller while they wait in line for an available agent. Choose "inherit" if you want the MoH class to be what is currently selected, such as by the inbound route. MoH Only will play music until the agent answers. Agent Ringing will play MoH until an agent's phone is presented with the call and is ringing. If they don't answer, MoH will return. Ring Only makes callers hear a ringing tone instead of MoH ignoring any MoH Class selected as well as any configured periodic announcements. This music is defined in the "Music on Hold" Menu.
- **Join Announcement** - Announcement played to callers prior to joining the queue. This can be skipped if there are agents ready to answer a call (meaning they still may be wrapping up from a previous call) or when they are free to answer the call right now. To add additional recordings please use the "System Recordings" MENU.
- **Call Recording** - Incoming calls to agents can be recorded. If 'never' is selected, then in-call on demand recording is blocked.
- **Mark calls answered elsewhere** - Enabling this option, all calls are marked as 'answered elsewhere' when cancelled. The effect is that missed queue calls are *not* shown on the phone (if the phone supports it)
- **Max wait time** - The maximum number of seconds a caller can wait in a queue before being pulled out. (0 for unlimited).
- **Max wait time mode** - Asterisk timeoutpriority. In 'Strict' mode, when the 'Max Wait Time' of a caller is hit, they will be pulled out of the queue immediately. In 'Loose' mode, if a queue member is currently ringing with this call, then we will wait until the queue stops ringing this queue member or otherwise the call is rejected by the queue member before taking the caller out of the queue. This means that the 'Max Wait Time' could be as long as 'Max Wait Time' + 'Agent Timeout' combined.
- **Agent Timeout** - The number of seconds an agent's phone can ring before we consider it a timeout. Unlimited or other timeout values may still be limited by system ringtime or individual extension defaults.
- **Agent Timeout Restart** - If timeoutrestart is set to yes, then the time out for an agent to answer is reset if a BUSY or CONGESTION is received. This can be useful if agents are able to cancel a call with reject or similar.
- **Retry** - The number of seconds we wait before trying all the phones again. Choosing "No Retry" will exit the Queue and go to the fail-over destination as soon as the first attempted agent times-out, additional agents will not be attempted.
- **Wrap-up Time** - After a successful call, how many seconds to wait before sending a potentially free agent another call (default is 0, or no delay)
- **Member Delay** - If you wish to have a delay before the member is connected to the caller (or before the member hears any announcement messages), set this to the number of seconds to delay.
- **Agent Announcement** - Announcement played to the Agent prior to bridging in the caller. Example: "the Following call is from the Sales Queue" or "This call is from the Technical Support Queue". To add additional recordings please use the "System Recordings" MENU.
- **Report Hold Time** - If you wish to report the caller's hold time to the member before they are connected to the caller, set this to yes.
- **Auto Pause** - Auto Pause an agent in this queue (or all queues they are a member of) if they don't answer a call. Specific behavior can be modified by the Auto Pause Delay as well as Auto Pause Busy/Unavailable settings if supported on this version of Asterisk.
- **Auto Pause on Busy** - When set to Yes agents devices that report busy upon a call attempt will be considered as a missed call and auto paused immediately or after the auto pause delay if configured.
- **Auto Pause on Unavailable** - When set to Yes agents devices that report congestion upon a call attempt will be considered as a missed call and auto paused immediately or after the auto pause delay if configured.

QUEUE - Continued

- **Auto Pause on Delay** - This setting will delay the auto pause of an agent by auto pause delay seconds from when it last took a call. For example, if this were set to 120 seconds, and a new call is presented to the agent 90 seconds after they last took a call, they will not be auto paused if they don't answer the call. If presented with a call 120 seconds or later after answering the last call, they will then be auto paused. If they have taken no calls, this will have no affect.
- **Max Callers** - Maximum number of people waiting in the queue (0 for unlimited)
- **Join Empty** - Determines if new callers will be admitted to the Queue, if not, the failover destination will be immediately pursued. The options include: **Yes** - Always allows the caller to join the Queue. **Strict** - Same as Yes but more strict. Simply speaking, if no agent could answer the phone then don't admit them. If agents are inuse or ringing someone else, caller will still be admitted. **Ultra Strict** - Same as Strict plus a queue member must be able to answer the phone 'now' to let them in. Simply speaking, any 'available' agents that could answer but are currently on the phone or ringing on behalf of another caller will be considered unavailable. **No** - Callers will not be admitted if all agents are paused, show an invalid state for their device, or have penalty values less then QUEUE_MAX_PENALTY (not currently set in FreePBX dialplan). **Loose** - Same as No except Callers will be admitted if their are paused agents who could become available.
- **Leave Empty** - Determines if callers should be exited prematurely from the queue in situations where it appears no one is currently available to take the call. The options include: **Yes** - Callers will exit if all agents are paused, show an invalid state for their device or have penalty values less then QUEUE_MAX_PENALTY (not currently set in FreePBX dialplan) **Strict** - Same as Yes but more strict. Simply speaking, if no agent could answer the phone then have them leave the queue. If agents are inuse or ringing someone else, caller will still be held. **Ultra Strict** - Same as Strict plus a queue member must be able to answer the phone 'now' to let them remain. Simply speaking, any 'available' agents that could answer but are currently on the phone or ringing on behalf of another caller will be considered unavailable. **Loose** - Same as Yes except Callers will remain in the Queue if their are paused agents who could become available. **No** - Never have a caller leave the Queue until the Max Wait Time has expired.
- **Penalty Members Limit** - A limit can be set to disregard penalty settings, allowing all members to be tried, when the queue has too few members. No penalty will be weighed in if there are only X or fewer queue members.
- **Frequency** - How often to announce queue position and estimated holdtime (0 to Disable Announcements)
- **Announce Position** - Announce position of caller in the queue?
- **Announce Hold Time** - Should we include estimated hold time in position announcements? Either yes, no, or only once; hold time will not be announced if <1 minute.
- **IVR Break Out Menu** - You can optionally present an existing IVR as a 'break out' menu.

This IVR must only contain single-digit 'dialed options'. The Recording set for the IVR will be played at intervals specified in 'Repeat Frequency', below.
- **Repeat Frequency** - How often to announce a voice menu to the caller (0 to Disable Announcements).
- **Service Level** - Used for service level statistics (calls answered within service level time frame)
- **Fail-Over Destination** - Destination to send call to upon exiting queue
- **Reset Queue Stats** - Select how often to reset queue stats. The following schedule will be followed for all but custom:
    - **Hourly** - Run once an hour, beginning of hour
    - **Daily** - Run once a day, at midnight
    - **Monthly** - Run once a month, midnight, first of month
    - **Annually** - Run once a year, midnight, Jan. 1
    - **Reboot** - Run at startup of the server OR of the cron daemon (after every: service cron restart)
    - If **Randomize** is selcted, a similar frequency will be followed, only the exact times will be randomized (avoiding peak business hours, when possible). Please note: randomized schedules will be rescheduled (randomly) every time ANY backup is saved
    - **Never** will never reset the queue stats automatically. If a custom schedule is selected, any section not specficed will be considered to be 'any' (aka: wildcard). I.e. if Day of Month is set to 12 and Day of Week is not set, the queue stats will be reset on ANY 12th of the month - regardless of the day of the week. If Day of Week is set to, say, Monday, the queue stats will be reset ONLY on a Monday, and ONLY if it's the 12th of the month.

Paging and Intercom

Intercom allows you to immediately page all phones in the paging group. Applications - Paging & Intercom. To add a new paging group, click “New Paging Group” in the options on the top right corner.

- **Paging Extension** - The number users will dial to page this group.
- **Group Description** - Provide a descriptive title for this Page Group.
- **Device List** - Add devices to this paging group by click and dragging users from the “Not Selected” group on the right to the “Selected” group on the left.
- **Busy Extensions** -
  - **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 page (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 page being "sent to the device's earpiece "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 too much sense to choose duplex if using Whisper mode.
- **Duplex** - Paging is typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all. This makes it like an "instant conference"

Parking

This module is used to configure parking lots. You can transfer a call to the parking lot extension and the call will enter one of a number of available “Lot” positions, making it available for any extension to pick up. Applications - Parking Lots. Click Default Lot to configure/make changes to your Lot.

- **Parking Lot Extension** - This is the extension where you will transfer a call to park it.
- **Parking Lot Name** - Provide a descriptive title for this Parking Lot.
- **Parking Lot Starting Position** - The starting postion of the parking lot
- **Number of Slots** - The total number of parking lot spaces to configure. Example, if 70 is the extension and 8 slots are configured, the parking slots will be 71-78. Users can transfer a call directly into a parking slot.
- **Parking Timeout** - The timeout period in seconds that a parked call will attempt to ring back the original parker if not answered.
- **Parking Music Class** - This 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.
- **BLF Capabilities** - Enable this to have Asterisk 'hints' generated to use with BLF buttons.
- **Find Slot** - Next: If you want the parking lot to seek the next sequential parking slot relative to the the last parked call instead of seeking the first available slot. First: Use the first parking lot slot available.
- **Pickup Courtesy Tone** - Whom to play the courtesy tone to when a parked call is retrieved.
- **Announcement** - Optional message to be played to the call prior to sending back to the Originator or the Alternate Destination.
- **Come Back To Origin** - Where to send a parked call that has timed out. If set to yes then the parked call will be sent back to the originating device that sent the call to this parking lot. If the origin is busy then we will send the call to the Destination selected below. If set to no then we will send the call directly to the destination selected below.