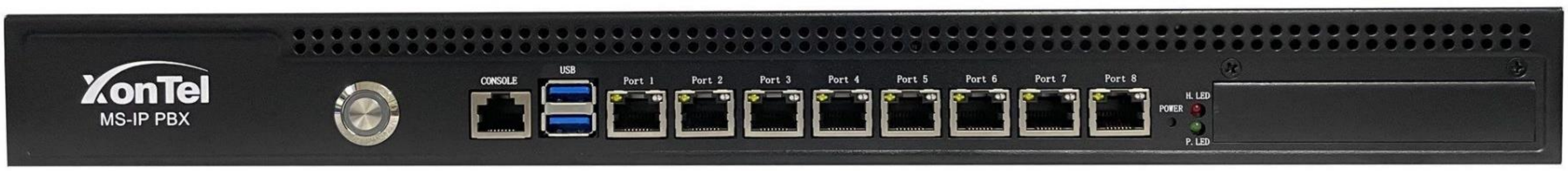
XonTel MS PBX
User Manual



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Contents

1. Overview

1.1 Introduction

The XonTel MS PBX delivers a multi-functional business office telephony system designed for small to medium enterprises. The series integrates functions such as IP phone, fax, and voice recording, and is compatible with multiple service platforms such as Cisco Call Manager, Avaya, Huawei and Asterisk, and terminals. The products are highly reliable, easy to install and deploy, and offer a brand-new experience in mobile offices and communications.

The XonTel MS PBX delivers a full-featured IP Telephony solution. By supporting intelligent communication functions such as mobile phone extensions, instant multi-party conferences, call history, it not only facilitates seamless communication between enterprise employees and customers, but also provides a solid basis for enterprises to analyze core business data.

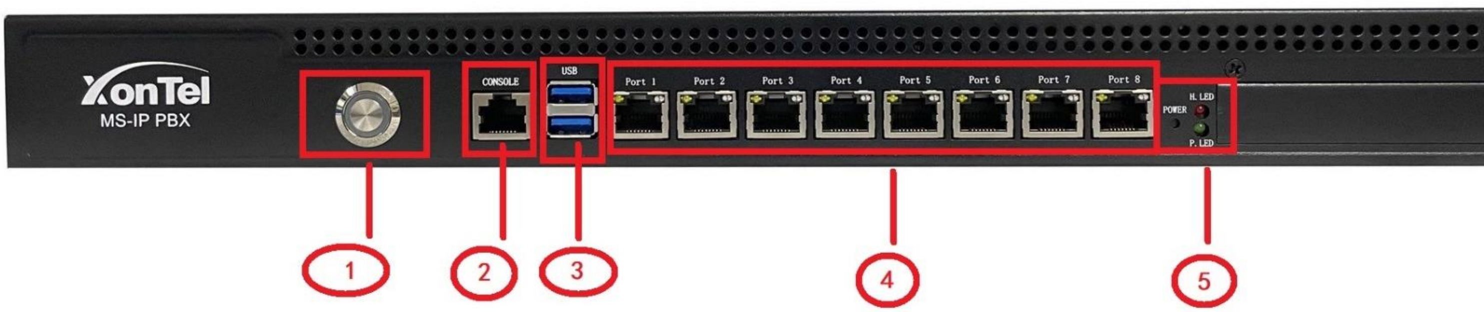
The XonTel MS PBX comes with an asterisk-based system, the IPPBX software, offering not only full PBX functionality, but also a new feature that enables new stability for your unified communication systems.

It can seamlessly integrate VoIP trunks and it has 6 Ethernet ports. MS PBX is developed with a wide selection of codes and signaling protocols, including G711 (alaw/ulaw), G722, OPUS, AMR-NB/WB, SILK, G723.1 G726, G729, GSM, ADPCM, ILBC, H263, H263P, H264, VP8.

Taking full advantages of open-source platform, the XonTel MS appliances support industry standard SIP trunks and IAX2 trunks.

1.2 Hardware Specifications

Front View



Number	Description
1	Power button
2	Console ports
3	USB ports
4	8 Network Ethernet ports
5	System LEDs

Back View

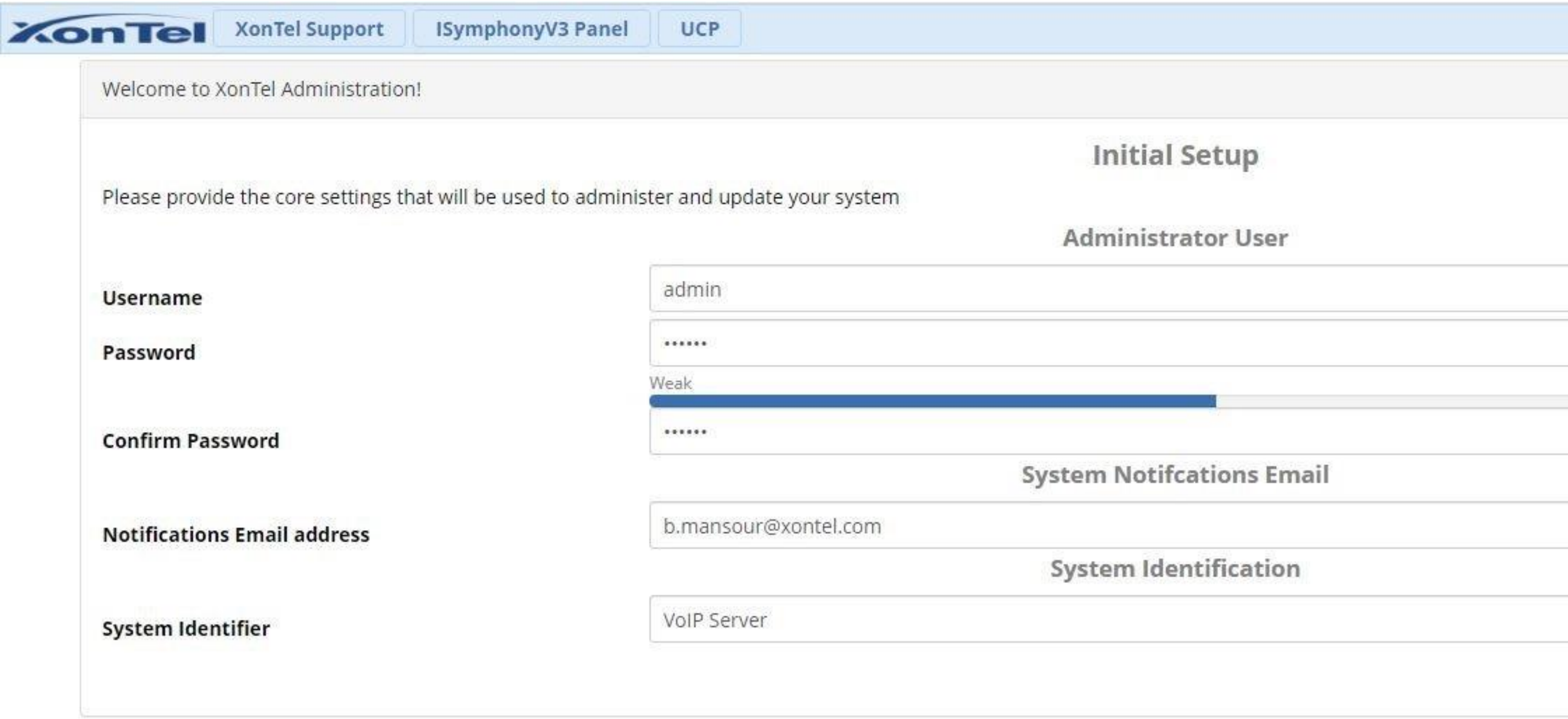
Number	Description
1	**Power Supply 100-240V AC**
2	FAN
3	COM
4	HDMI port

1.3 Log in to the Web GUI

Using another machine on your same network, open a web browser and enter the IP address of your PBX.

Please note the default IP of XonTel MS PBX is DHCP depend on your network router.

The first time you do so, you'll be asked to create the admin username and the admin password. That username and password will be used in the future to access the XonTel PBX configuration screen.



The main XonTel PBX screen will offer you three options:

a. **XonTel PBX Administration** will allow you to configure your PBX. Use the admin username and admin password you configured in the step above to login.

b. **User Control Panel** is where a user can log in to make web calls, set up their phone buttons, view voicemails, send and receive faxes, view conferences, and more, depending on what you have enabled for the user.

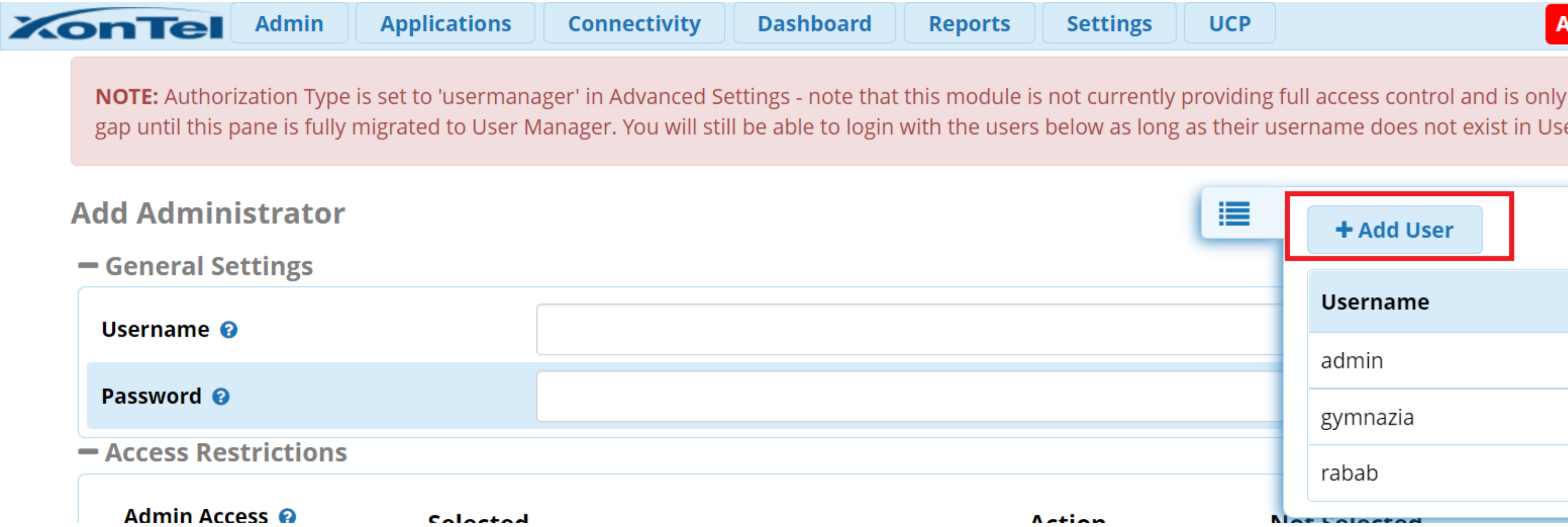
c. **Get Support** takes you to a web page about various official support options for XonTel PBX.

2. Administration modules

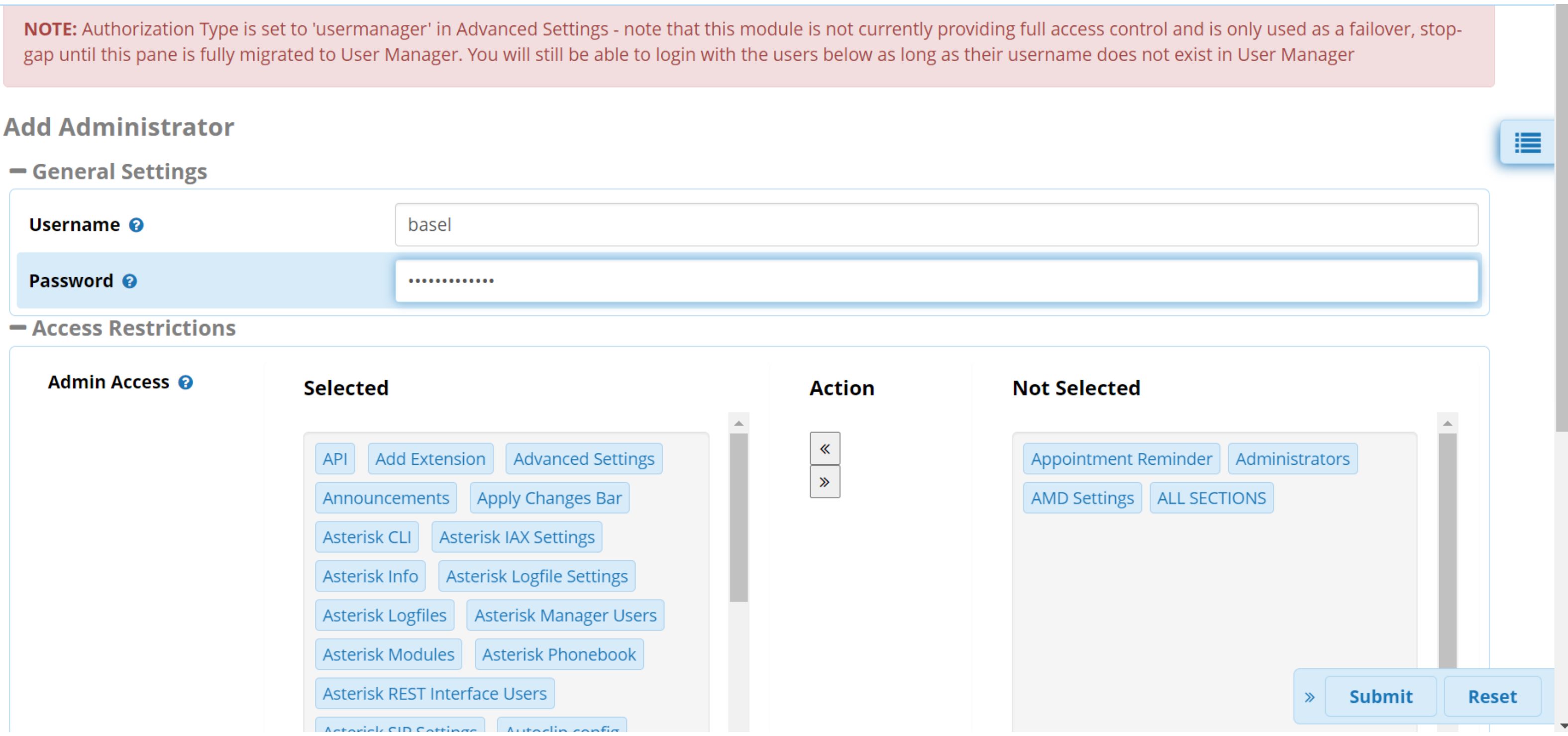
2.1 Administrators

From Administrators module you can create users to access XonTel MS PBX through web interface and choose the desired permissions for these users.

Only what you have to do is click Add User button as shown below.

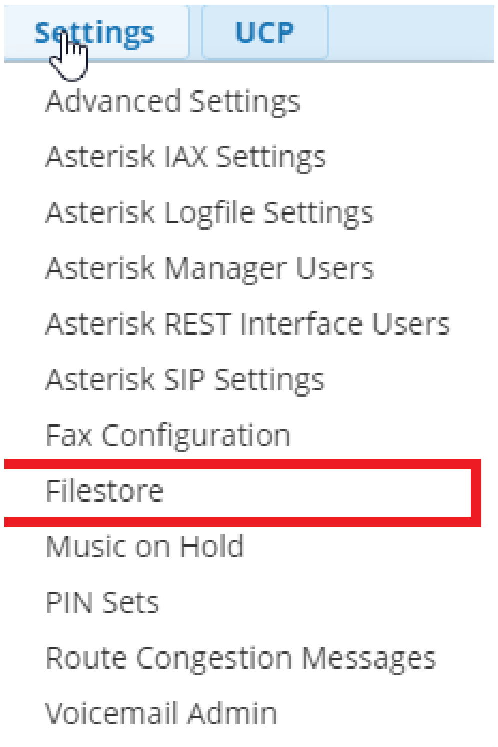


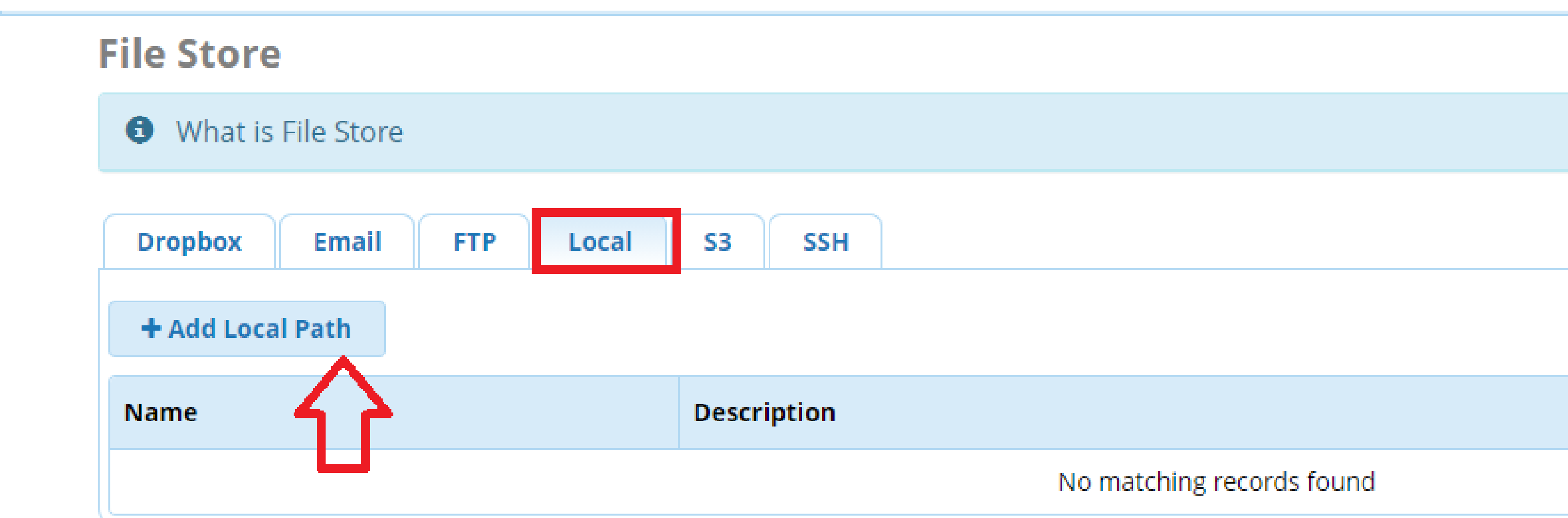
After that set the username and password for the user that you are going to create then set the desired permissions

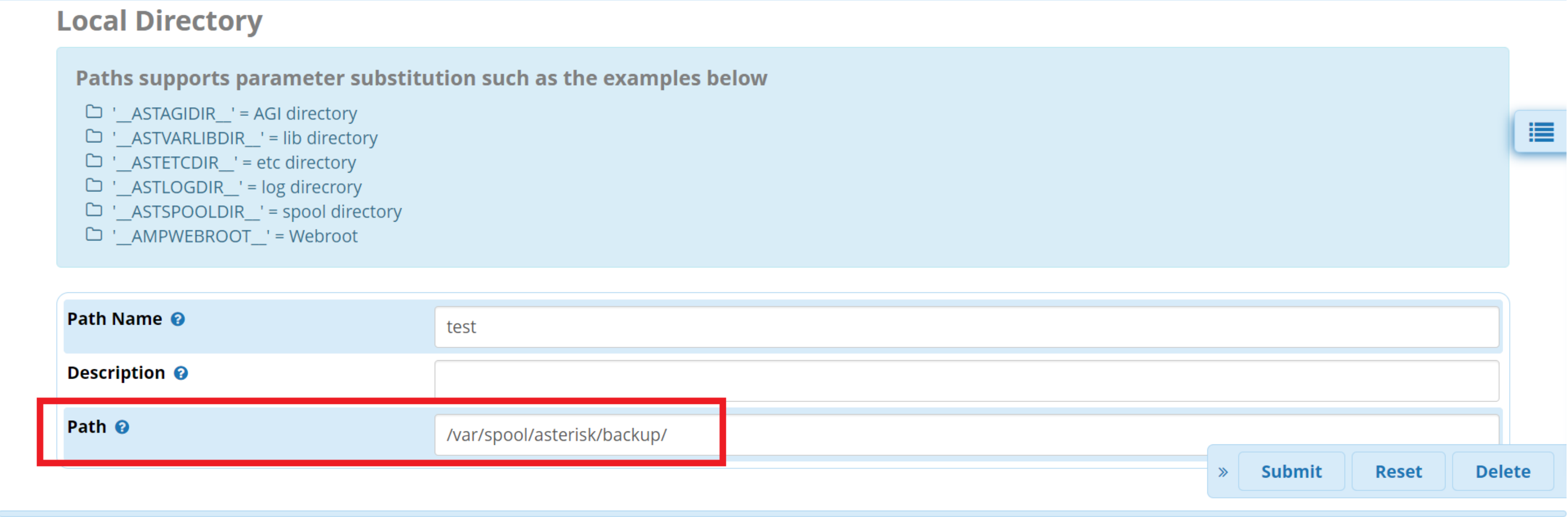

NOTE: Authorization Type is set to 'usermanager' in Advanced Settings - note that this module is not currently providing full access control and is only used as a failover, stop-gap until this pane is fully migrated to User Manager. You will still be able to login with the users below as long as their username does not exist in User Manager
Add Administrator
General Settings
Username
basel
Password
Access Restrictions
Admin Access
Selected
Action
Not Selected
API
Add Extension
Advanced Settings
Announcements
Apply Changes Bar
Asterisk CLI
Asterisk IAX Settings
Asterisk Info
Asterisk Logfile Settings
Asterisk Logfiles
Asterisk Manager Users
Asterisk Modules
Asterisk Phonebook
Asterisk REST Interface Users
Appointment Reminder
Administrators
AMD Settings
ALL SECTIONS
Submit
Reset

2.2 Backup & Restore

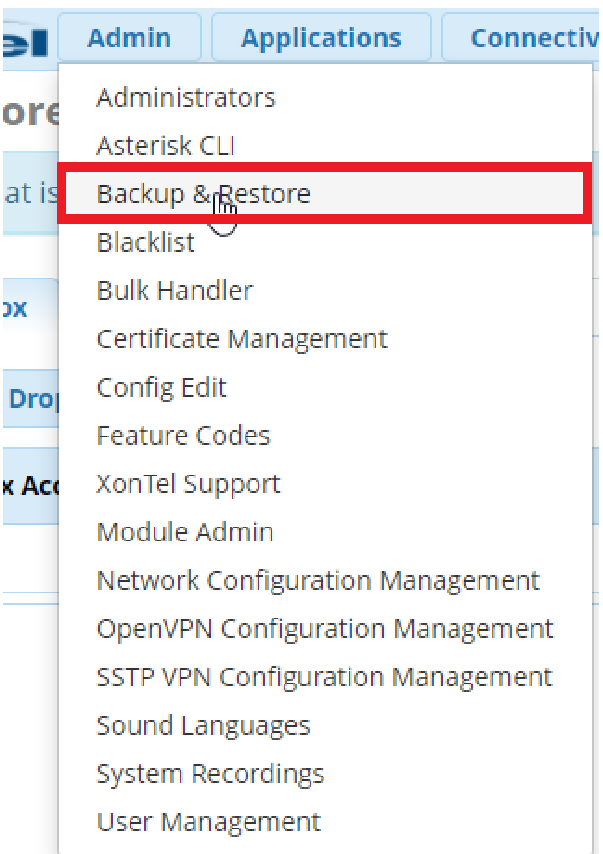
1. Create local directory path as shown in the figures below

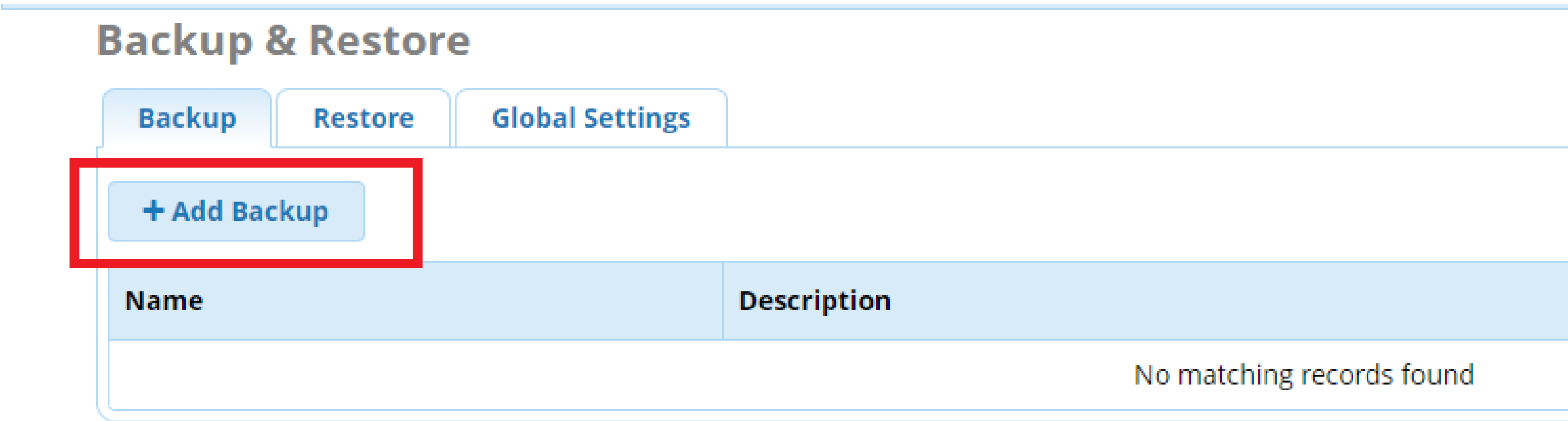


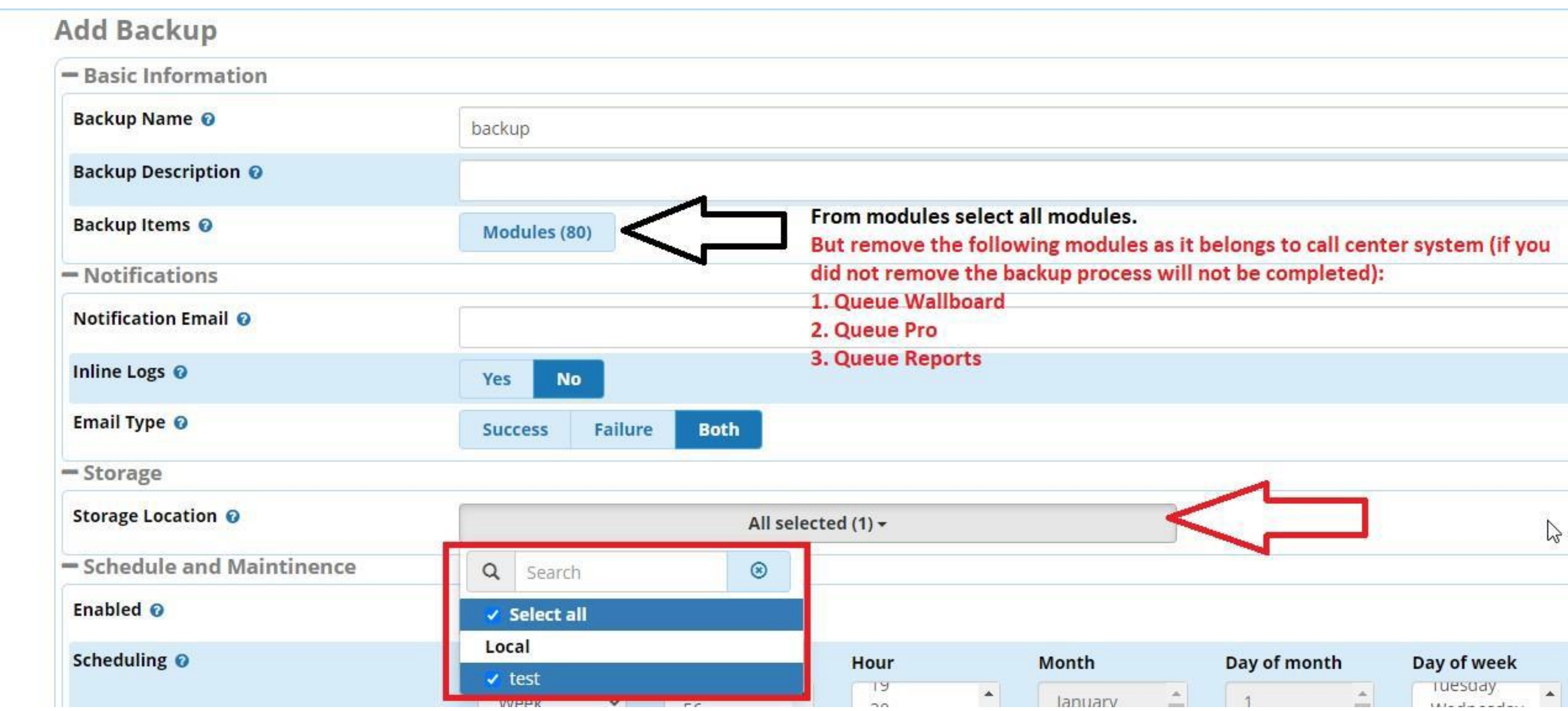




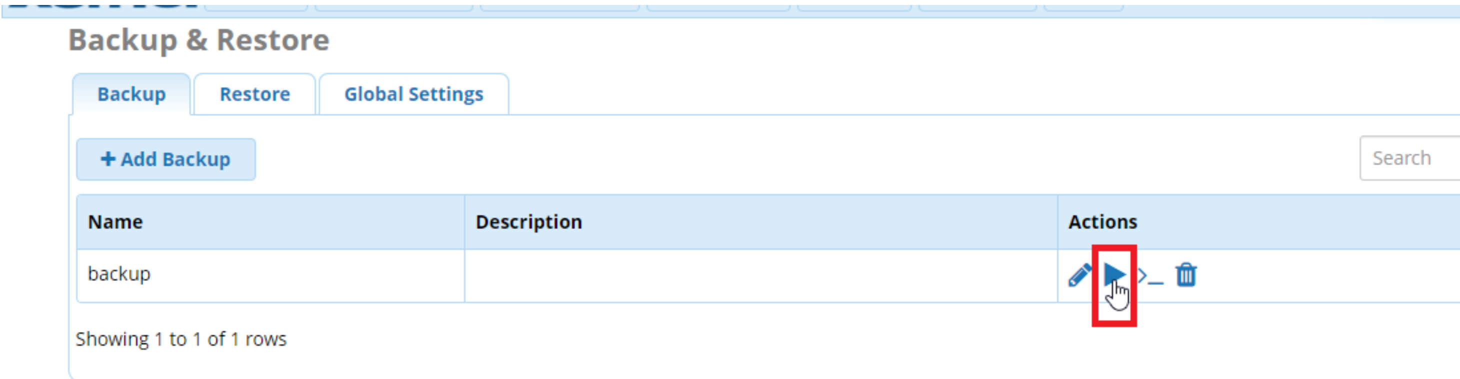
2. Create new backup as shown below

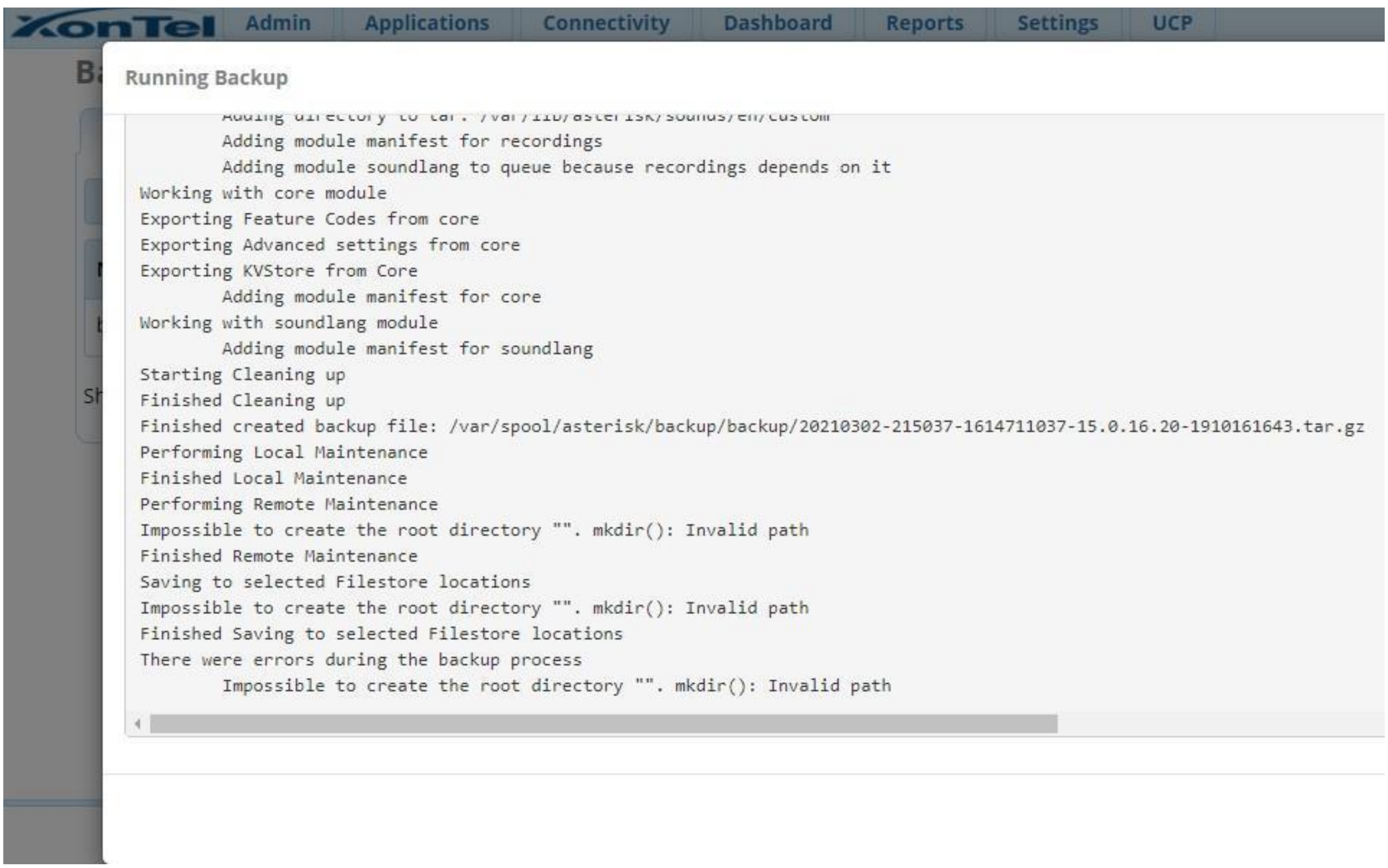



Backup & Restore
Backup
Restore
Global Settings
+ Add Backup
Name
Description
No matching records found

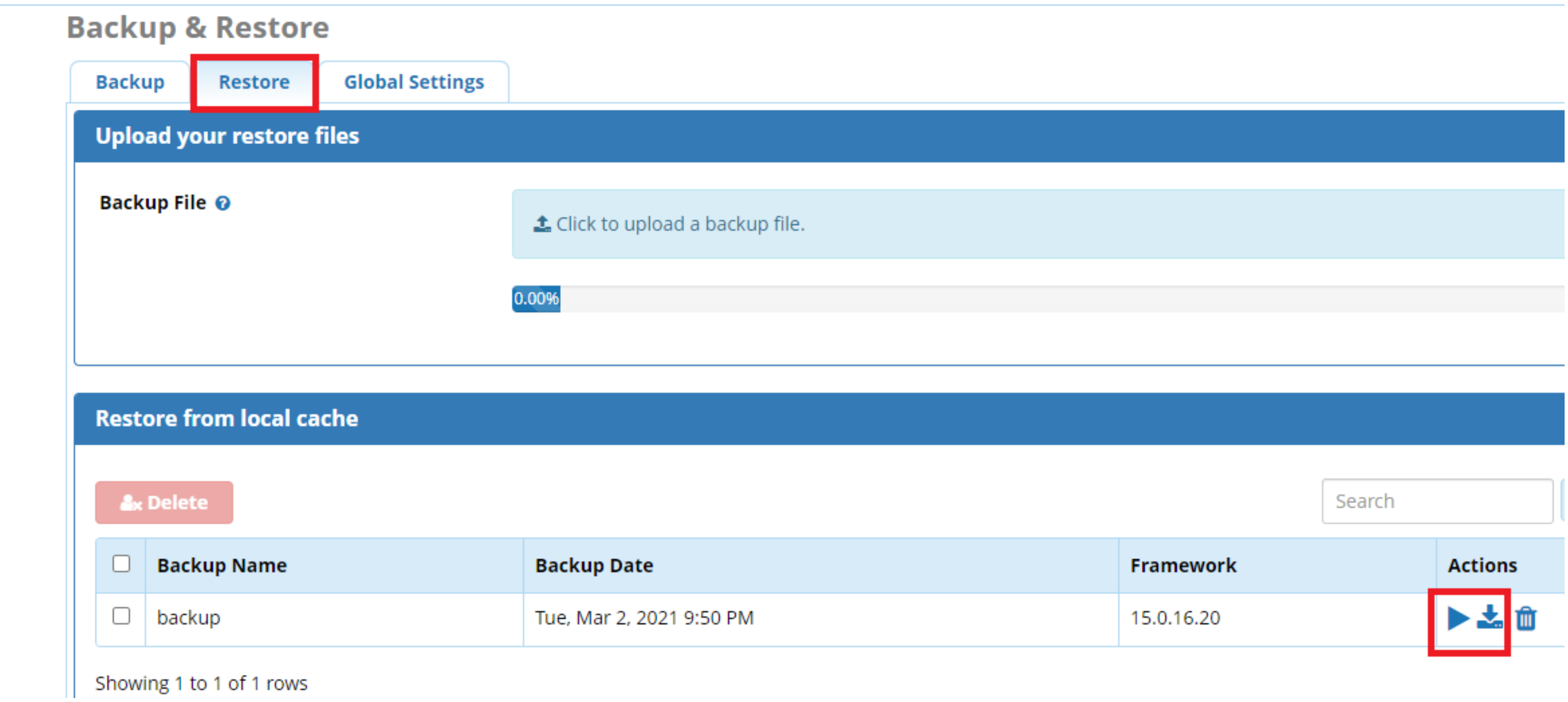

Add Backup
Basic Information
Backup Name
backup
Backup Description
Backup Items
Modules (80)
From modules select all modules.
But remove the following modules as it belongs to call center system (if you did not remove the backup process will not be completed):
1. Queue Wallboard
2. Queue Pro
3. Queue Reports
Notifications
Notification Email
Inline Logs
Yes
No
Email Type
Success
Failure
Both
Storage
Storage Location
All selected (1)
Search
Select all
Local
test
Schedule and Maintinence
Enabled
Scheduling
Hour
Month
Day of month
Day of week

3. Run the backup generation process that you created to generate the backup file and wait till generation process completed.



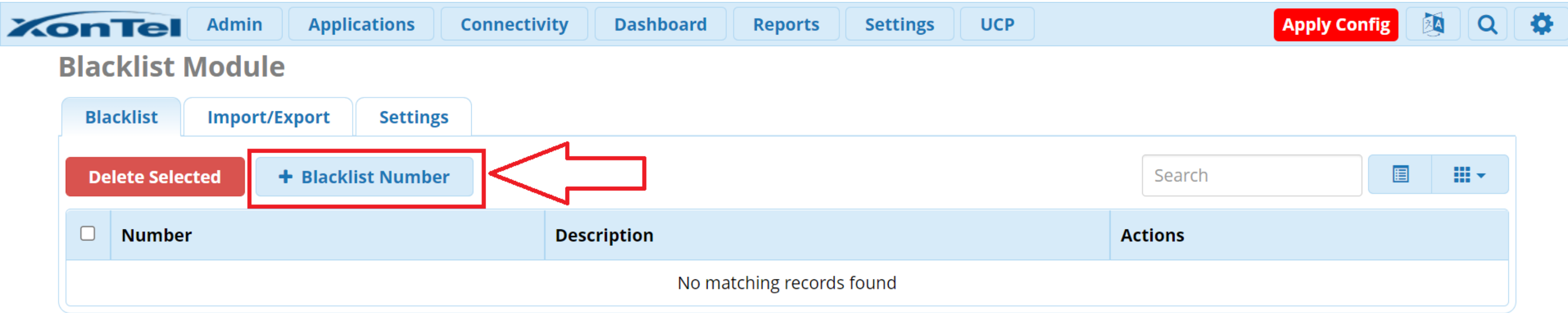


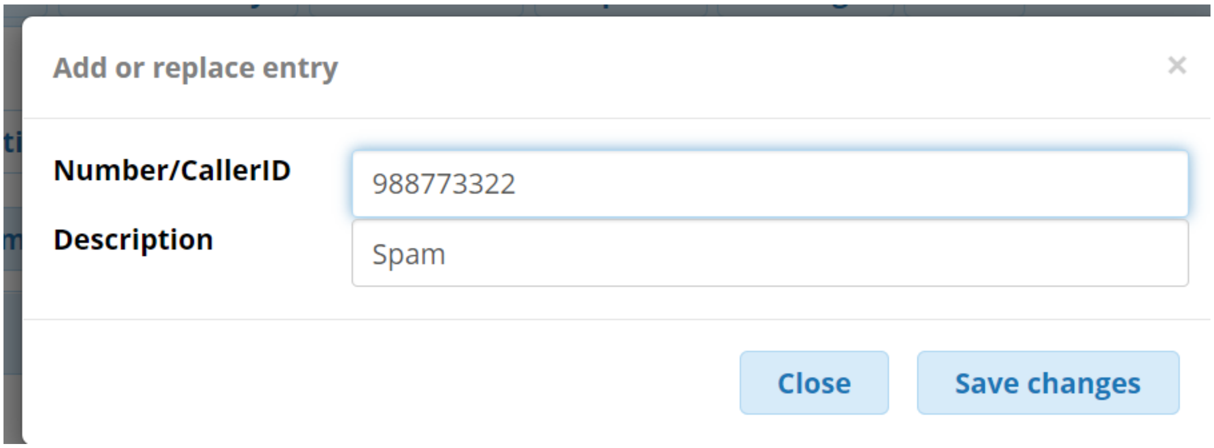
4. As you see here you can download backup to your PC, restore PBX to the current backup that you created or upload backup from your PC then restore it in PBX



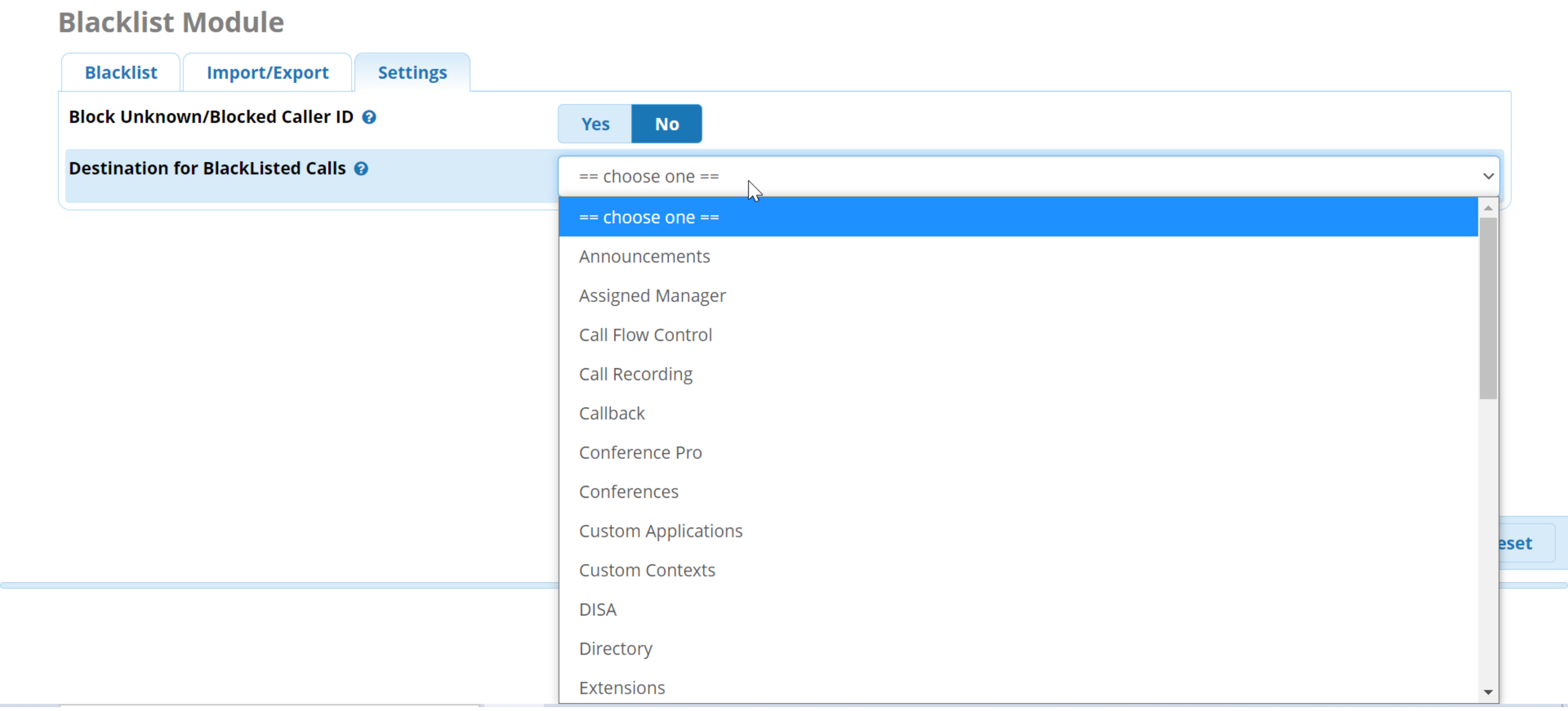
2.3 Blacklist

From here you can blacklist a number in XonTel PBX as shown below.





Please note that from Blacklist module settings you can block unknown caller ID and set special destination for the blocked caller ID as explained before


Blacklist Module
Blacklist
Import/Export
Settings
Block Unknown/Blocked Caller ID
Yes
No
Destination for BlackListed Calls
== choose one ==
== choose one ==
Announcements
Assigned Manager
Call Flow Control
Call Recording
Callback
Conference Pro
Conferences
Custom Applications
Custom Contexts
DISA
Directory
Extensions
eset

2.4 Features Codes

Blacklist codes

Name	Code	Description
Blacklist a number	*30	Add a new number to the black list. All callers from blocked numbers will hear the corresponding audio recording.
Blacklist the last caller	* 32	Add the last caller to the IP - PBX in the black list
Remove a number from the blacklist	* 31	Delete the number from the black list. Number is entered manually

Call forwarding codes

Name	Code	Description
Call Forward All Activate	* 72	Forward all incoming to the extension to another number.
Call Forward All Deactivate	* 73	Turn off call forwarding.
Call Forward All Prompting Activate	* 93	Asks the caller to enter the number on which you want to enable call forwarding.
Call Forward All Prompting Deactivate	* 74	Asks the caller to enter the number on which to disable call forwarding
Call Forward Busy Activate	* 90	Enables call forwarding if the called number is busy.
Call Forward Busy Deactivate	* 91	Disables call forwarding if the called number is busy.
Call Forward Busy Prompting Activate	* 94	Prompts to enter the number on which you want to enable call forwarding by the result of "Busy"
Call Forward Busy Prompting Deactivate	* 92	Prompts to enter the number on which you want to disable redirection call by the result of "Busy"

Name	Code	Description
Call Forward No Answer / Unavailable Activate	* 52	Activates call forwarding in case the user is unavailable or does not answer the call
Call Forward No Answer / Unavailable Deactivate	* 53	Deactivates call forwarding if the user is unavailable or does not answer the call
Call Forward No Answer / Unavailable Prompting Activate	* 95	Prompts to enter the number on which you want to connect call forwarding by no answer or unavailability
Call Forward Toggle	* 96	Enables or disables the call forwarding mode. The first call to *** 96** will turn off the function, and the second will turn it on. And so on.

Call Waiting Codes

Name	Code	Description
Call Waiting - Activate	* 70	This feature allows you to configure the receiving of a call, even if the subscriber is already in a talk state. This option is supported only on telephones, which have the ability to accept multiple calls.
Call Waiting - Deactivate	* 71	Disables your specified functionality

System core codes (core)

Name	Code	Description
Asterisk General Call Pickup	*8	Dial service code data to intercept a call that rings on the other phone to make this function work, make sure: - The caller must have the **Call Group** configured in the extension settings. - On the number from which you want to intercept the call, the **Pickup Group** field should be configured in the phone number settings.
ChanSpy	555	Very - very convenient function? Allows you to listen to the conversations of employees, while giving tips to the subscriber of their network, which the caller from the city will not hear. This is convenient if you are training a new employee, and, when communicating with a client, you want to tell him some nuances in real time
Directed Call Pickup	**	Dial the service code of this function, and then the extension number from which you want to intercept the call. The function allows you to intercept calls

Name	Code	Description
		from even numbers that do not have a common **Call Group** and **Pickup Group**
In-Call Asterisk Attended Transfer	* 2	This feature allows you to make a consultative call transfer, ie a transfer, in which the operator initially dials to the subscriber to whom the call should be transferred, speaks to him, and then connects the caller to this subscriber.
In-Call Asterisk Blind Transfer	##	Dial this code to make a "blind", then a transfer, without prior consultation.
In-Call Asterisk Disconnect Code	**	Instant reset of incoming call.
In-Call Asterisk Toggle Call Recording	*1	In fact, this service code activates the call recording "on demand". In the world, this type of record is known as **On Demand** or **Prerecording** . On-demand recording means that by default, the call is not recorded, but if you need to record a conversation, simply press the specified service code.
Simulate Incoming Call	7777	IP - Asterisk makes a test incoming call to an internal number
User Logoff	*12	This code allows the user to be logged off from the phone. This option is only available if the PBX is configured to use Device & User Mode.
User Logon	*eleven	Allows to make login to the user in the case described above
ZapBarge	888	This code allows the user to monitor audio on the drivers of the E1 interface, that is, on Zaptel or Dahdi drivers.

Do Not Disturb (DND) control codes

Name	Code	Description
DND Activate	* 78	This service code places the extension number in the Do Not Disturb state. This means that all callers to the number of subscribers will either hear the signal busy, or they will be sent to the voicemail.
DND Deactivate	* 79	Disables DND mode on the number
DND	* 76	Enables / disables DND activation for an extension

Name	Code	Description
Toggle		

Follow Me

Name	Code	Description
Findme Follow Toggle	* 21	The code allows you to enable or disable Follow Me settings for the extension number.

Information Services

Name	Code	Description
Call Trace	* 69	The system announces the Caller ID of the last caller to this extension number.
Echo Test	* 43	This function is used to check the quality of the connection, including the microphone, the speaker of the device and so on.
Speak Your Exten Number	* 65	The system pronounces the extension number configured on the telephone in use.
Speaking Clock	* 60	The system says the current server time.

The functions of Paging and Intercom (Intercom)

Name	Code	Description
Intercom prefix	* 80	This function is necessary in order that instead of the usual dialing to the number, you did not wait for the buzzer, and with the help of the loudspeaker the message was pronounced. Here is an example of how it works: The user types this service code, followed by the extension number. Further, all subsequent calls to this number will be accepted immediately without the caller being contacted and through the speakerphone, the caller will be able to deliver his message.
User Intercom Allow	* 54	Enable receiving of intercom messages (speakerphone, as described above).
User Intercom Disallow	* 55	Disables the above function.

Call Park

Name	Code	Description
Pickup ParkedCall Prefix	* 85	When the administrator has configured the slot for the parking call, the user can park this call by transfer to the parking number - by default, this number is 70. Even if this slot is occupied, the number of possible slots can be indicated in the **Parking** module configuration. The system will automatically park the call on an available slot and say its number. This service code is responsible for raising the call from the parking slot.

Queues

Name	Code	Description
Allow Dynamic Members of a Queue to login or logout. See the Queues Module for how to assign a Dynamic Member to a Queue.	* 45	This option allows dynamic queue members to connect and disconnect from it
Playback Queue Caller Count	* 47	To say the number of people in the queue
Queue Pause Toggle	* 46	Take a pause in the queue and do not take calls. Reactivation will return the user to the queue.

Time Conditions

The specified service code, and by default this *** 27** allows you to manage the settings of the time condition. Within the system, each new Time Condition generates its own service code, which is *** 27X** , where X is the time condition number.

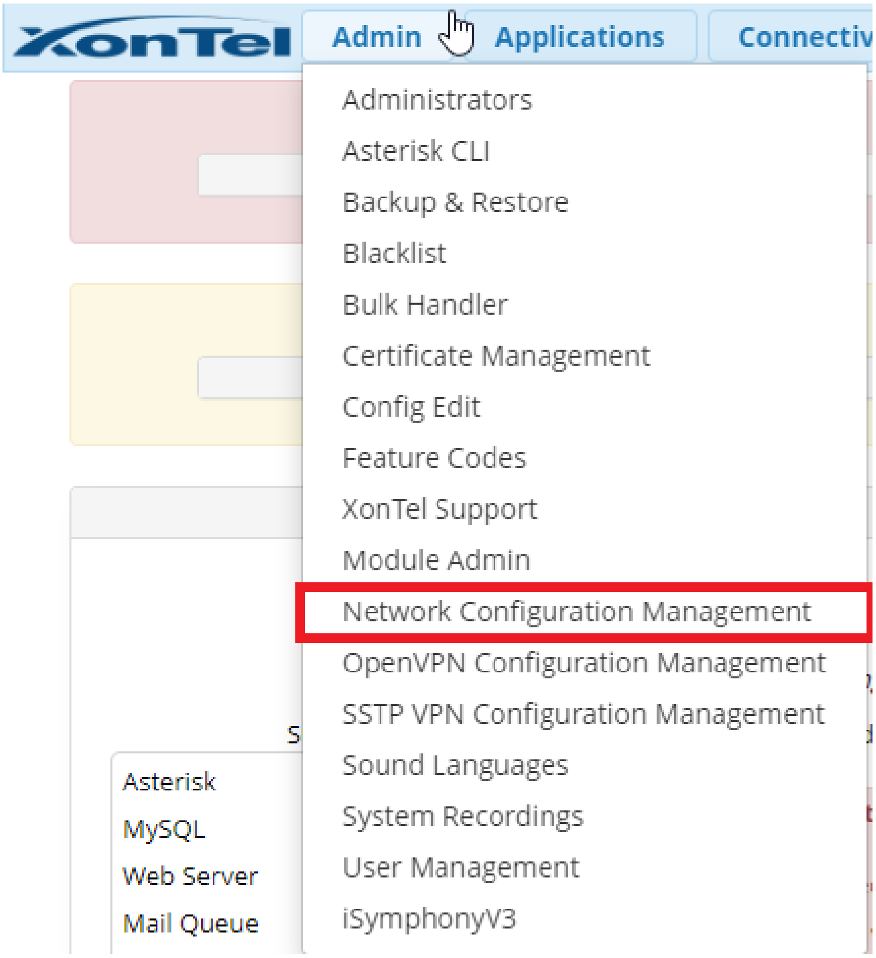
Voicemail

Name	Code	Description
Dial Voicemail	* 98	By dialing this service code, you will be prompted to enter the voice mailbox number and listen to it.
My Voicemail	* 97	Access to the voice mailbox, which refers to the number from which this code is dialed (listening to your own entries)

2.5 Network Configuration Module

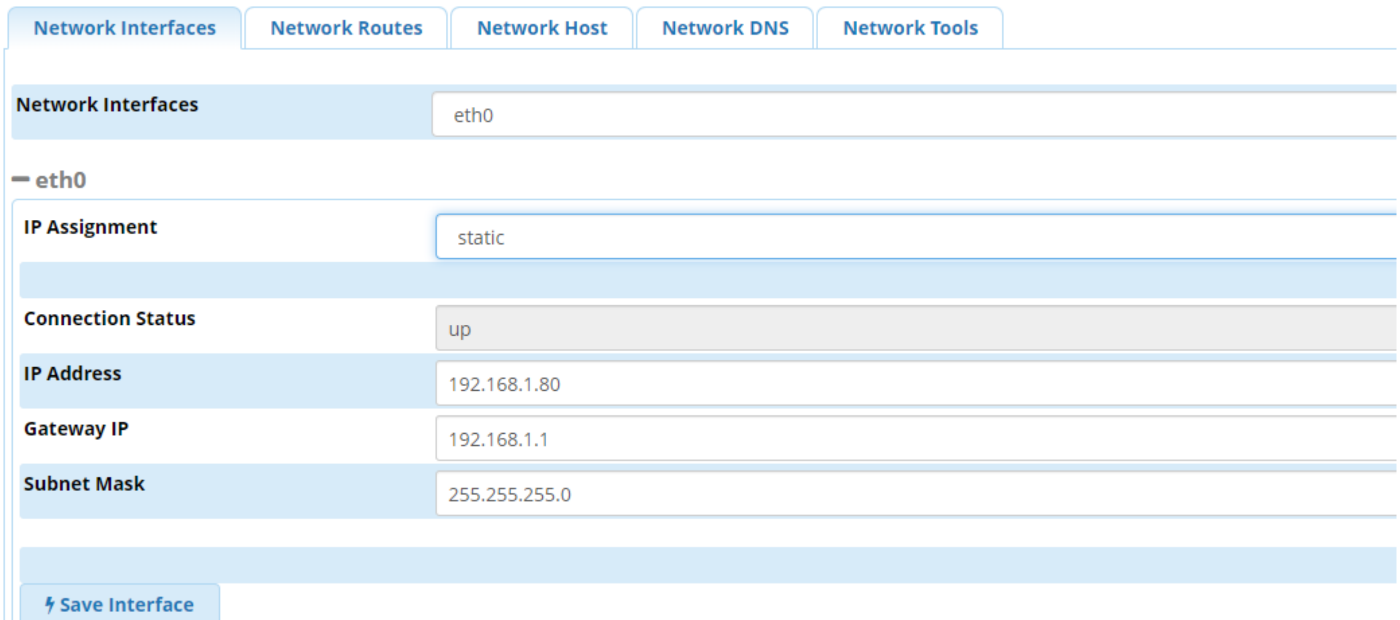
From here you configure XonTel MS PBX network settings

Go to " **Network Configuration Management** " from **Admin** Menu.



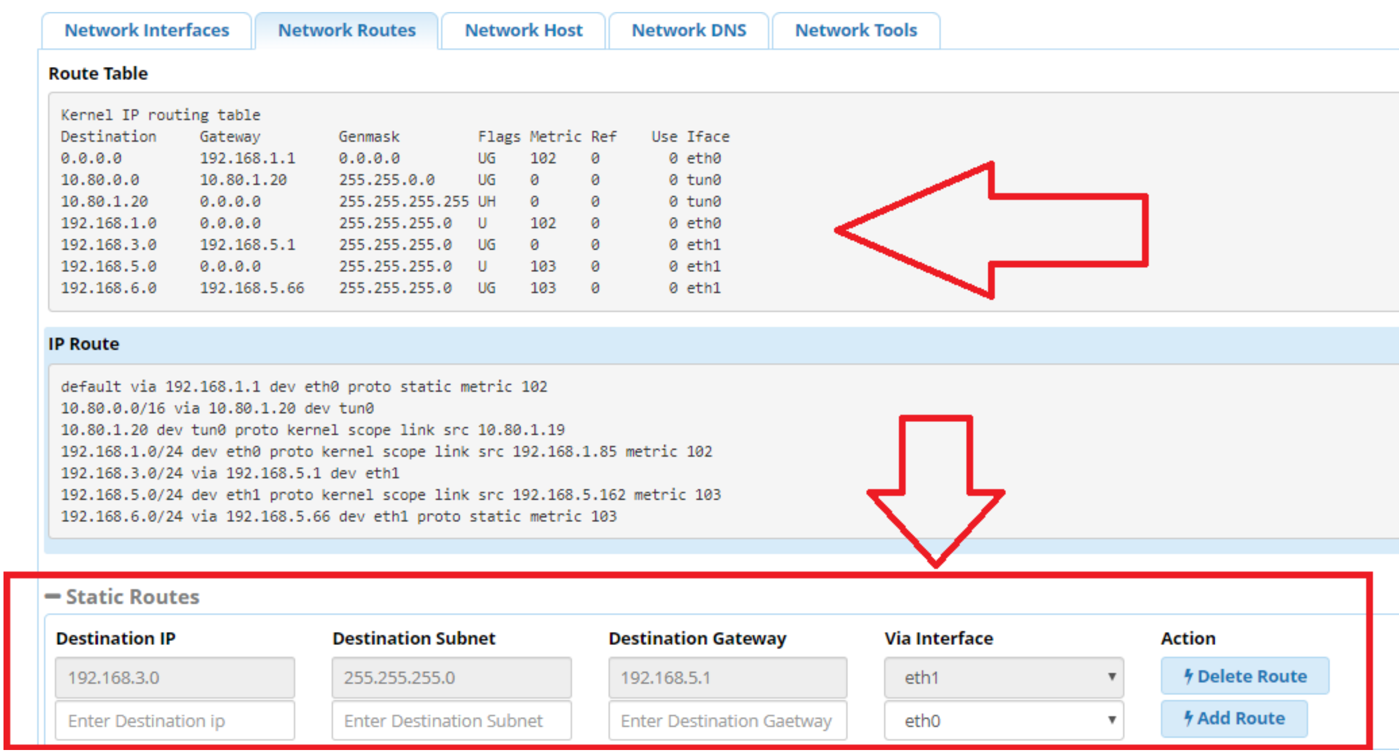
The figures below are the configuration options for **Network Configuration Manager**.

Network Interfaces configuration

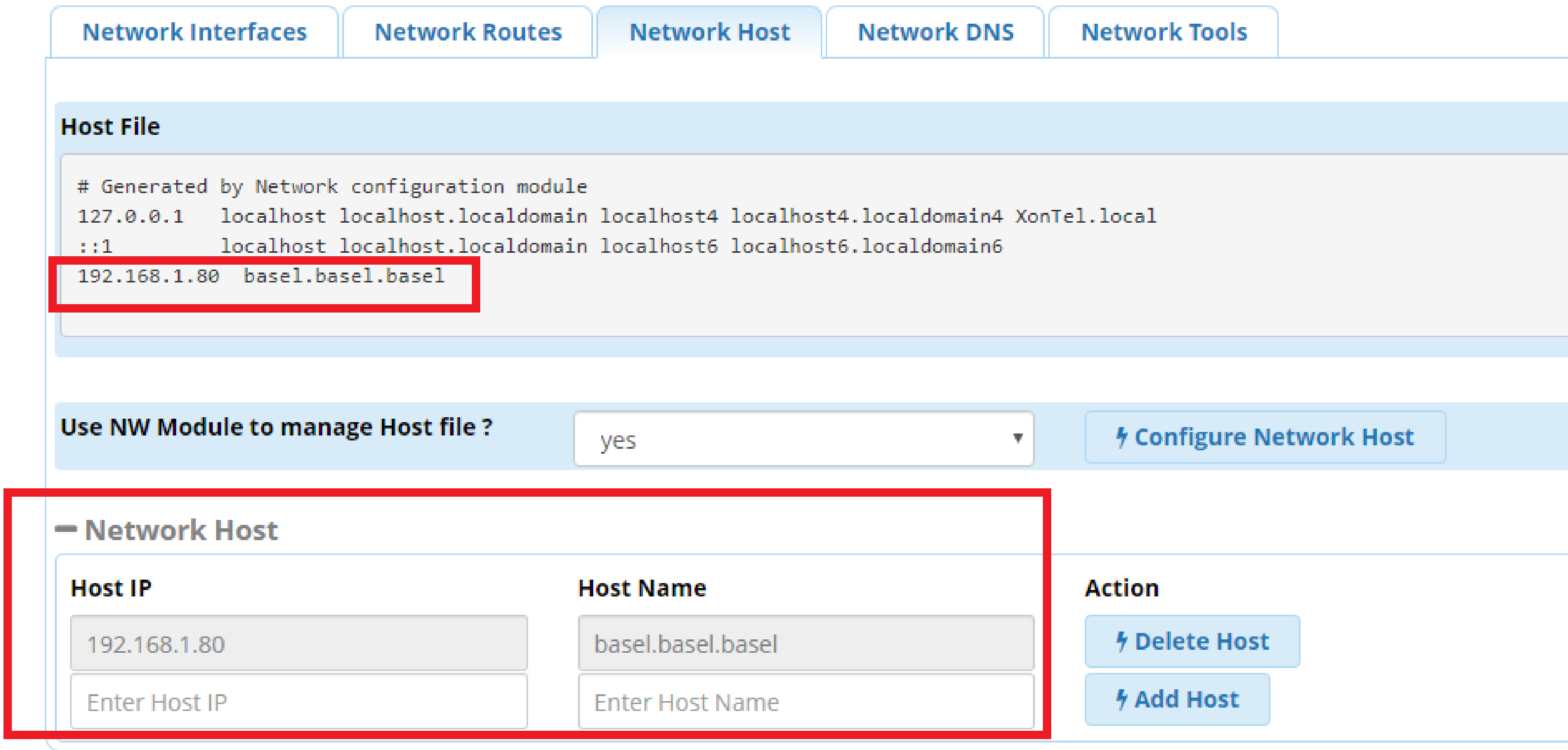


Network Routes Configuration

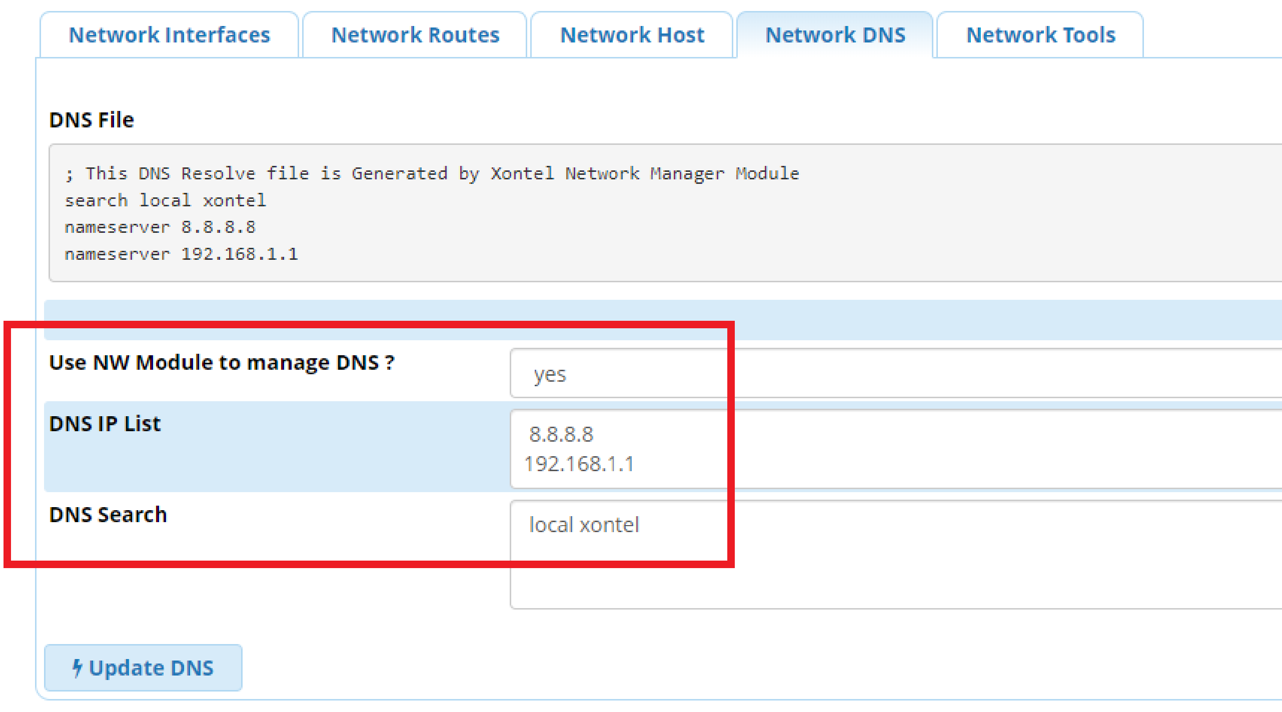
We can add new Static routes and manager their addition/deletion.



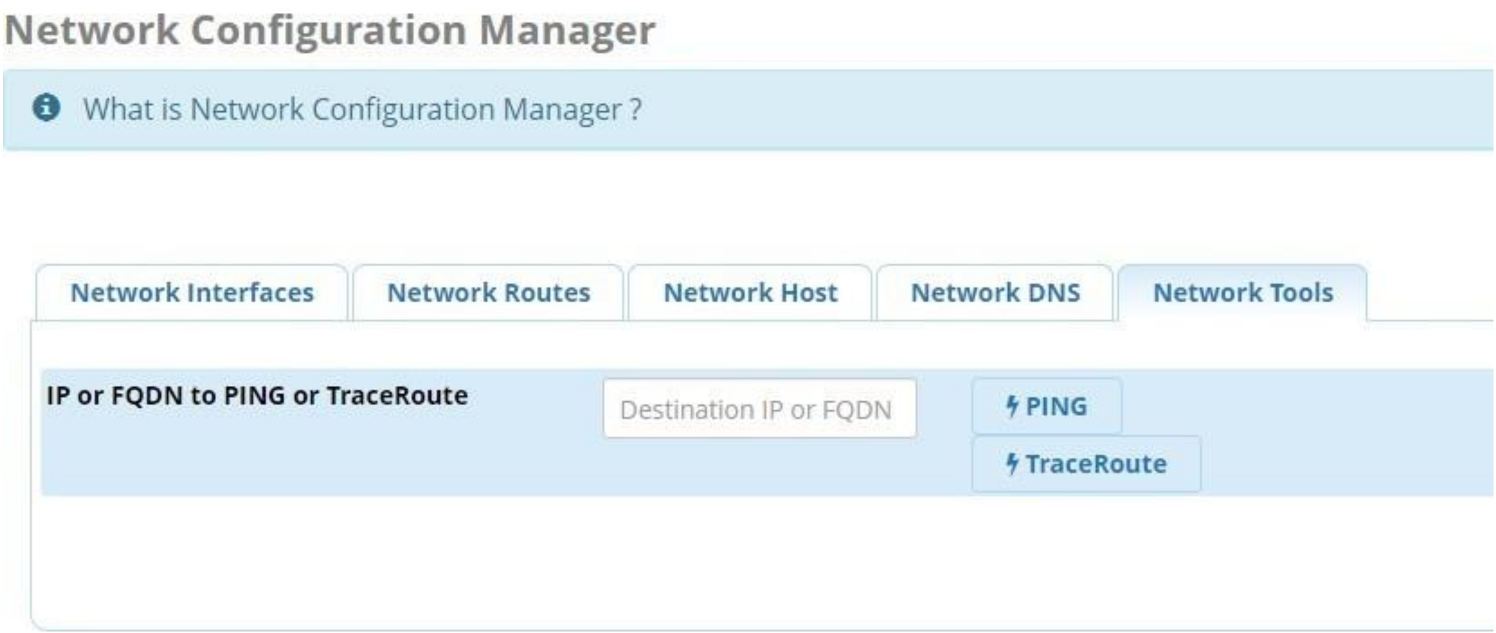
Network Host configuration



Network DNS configuration

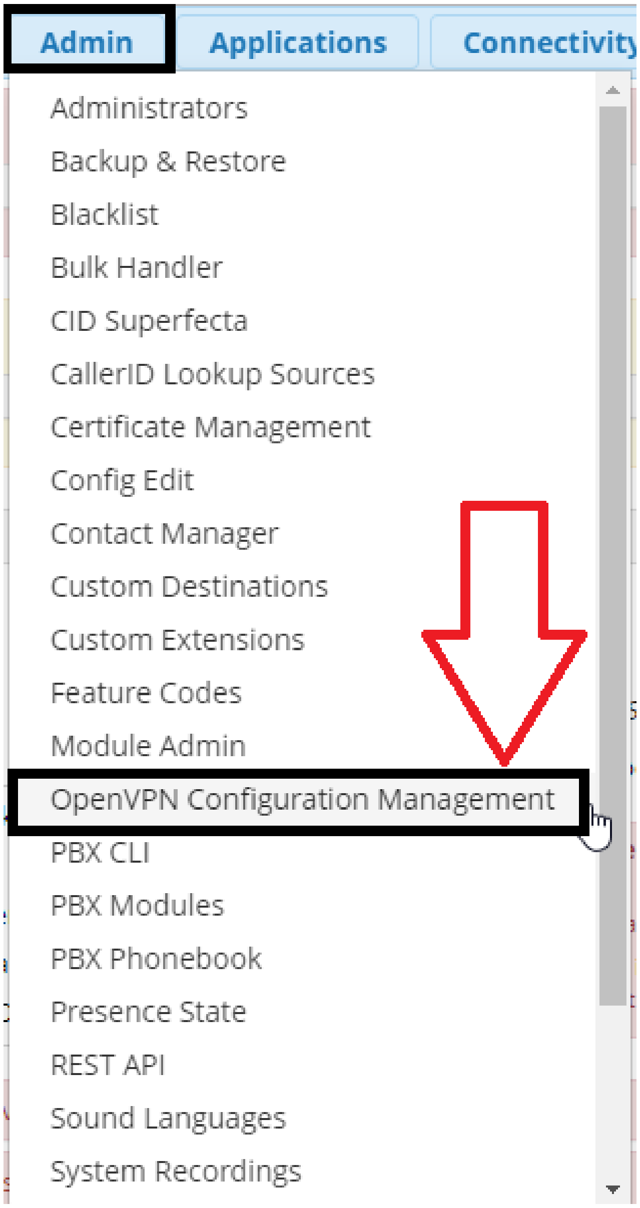


Network Tools (Ping and Traceroute) configuration.



2.6 Open VPN Module

From here you can connect XonTel MS PBX with an Open VPN server by uploading Open VPN certificate as shown below



Below are the configuration options for “ **OpenVPN Configuration Management** ” module.

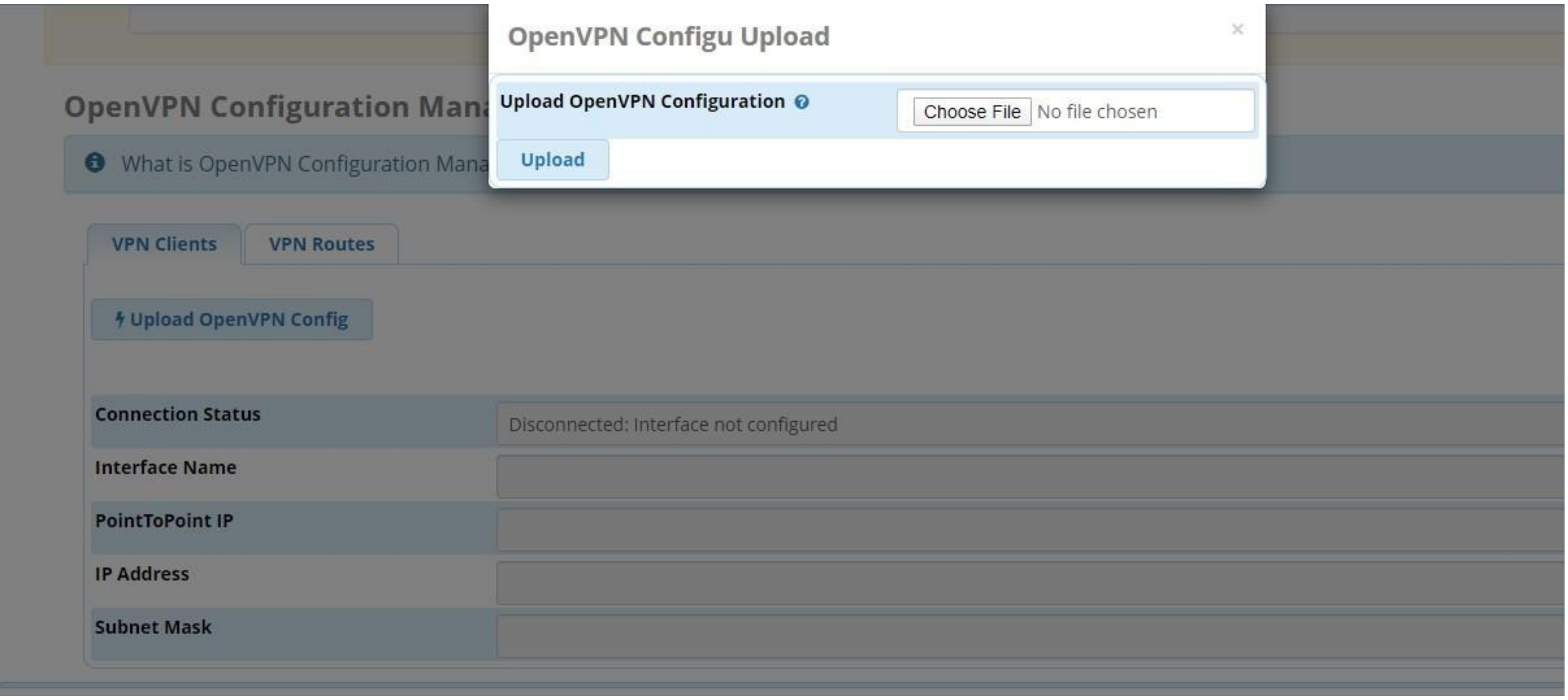
Two tabs are present:

A) **VPN Clients** Tab will be used to upload VPN client configuration.

B) **VPN Routes** tab to view the routing table (for debugging/quick view).

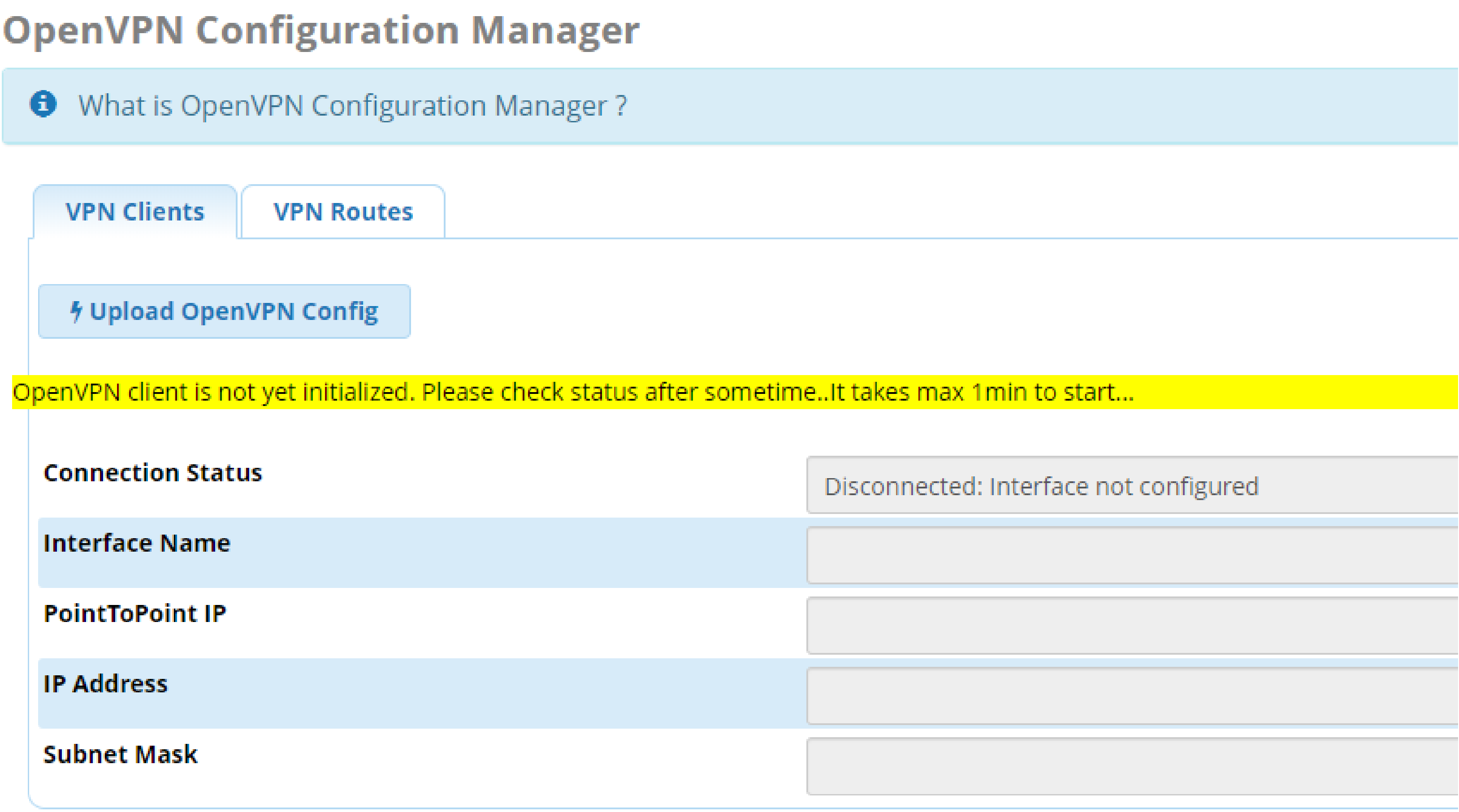
- On **VPN Clients** tab click on “ **Upload OpenVPN Config** “ to upload the OpenVPN certificate to the PBX.

Validations are added to ensure we can only upload “OpenVPNConfig.tar ” file.



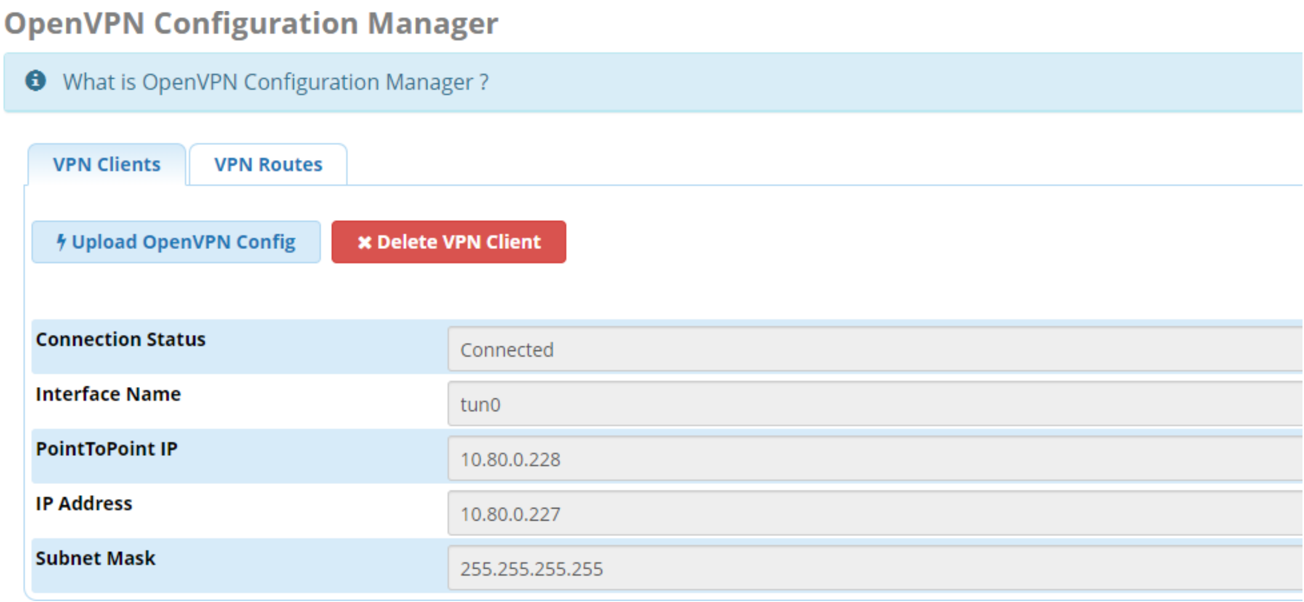
Upload configuration and VPN client initialization will take around one minute time. We need to refresh the page to view status.

While VPN Client initialization is going on, you are allowed to do any other MS-PBX configuration. You can come back to OpenVPN page after sometime to check the status again.

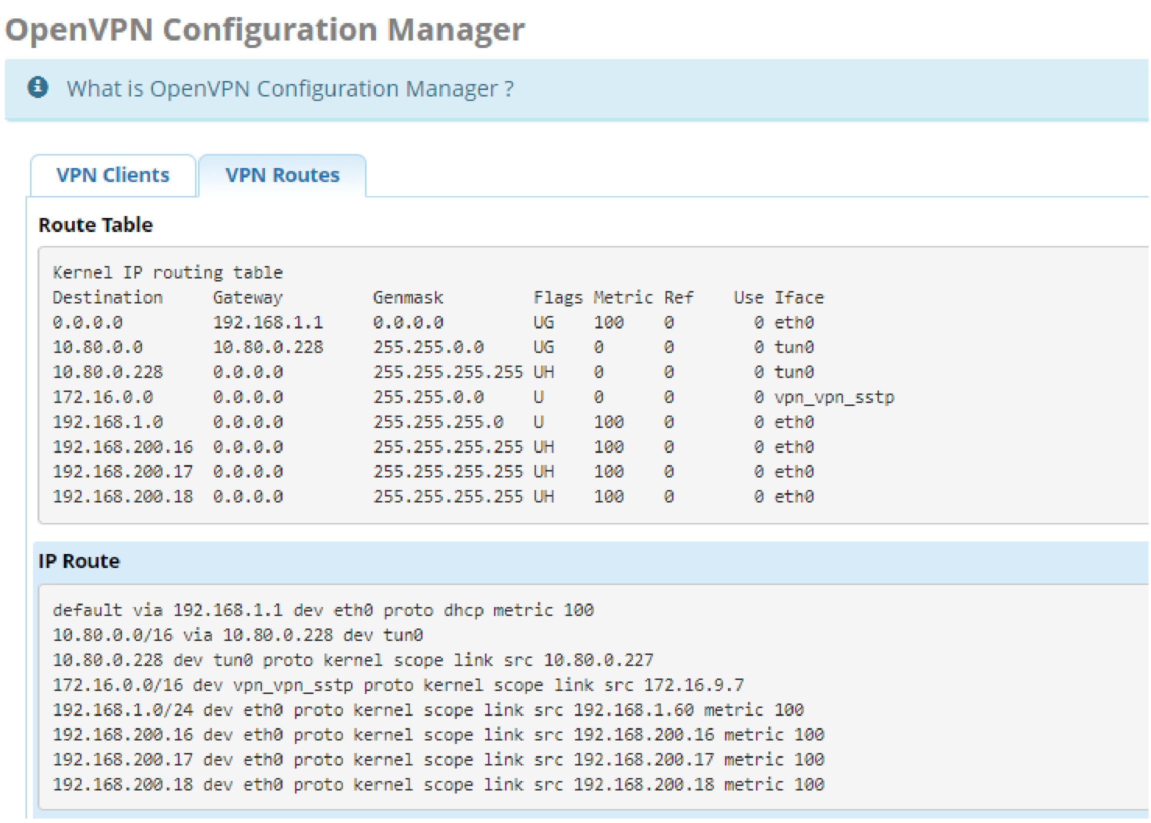


Once connection is initialized properly, we can see below status in **VPN Clients** tab.

Please note – Now " Delete VPN Client " option is visible in case we want to remove the VPN from this system



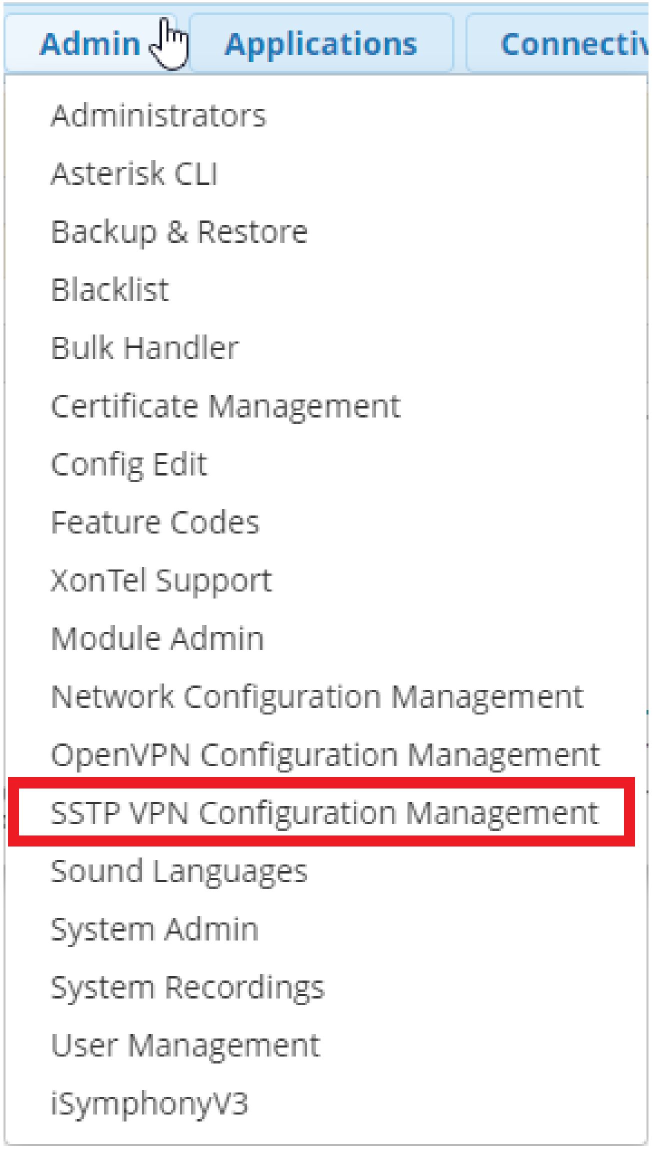
VPN Routes tab will display the VPN routes as well as the default route.



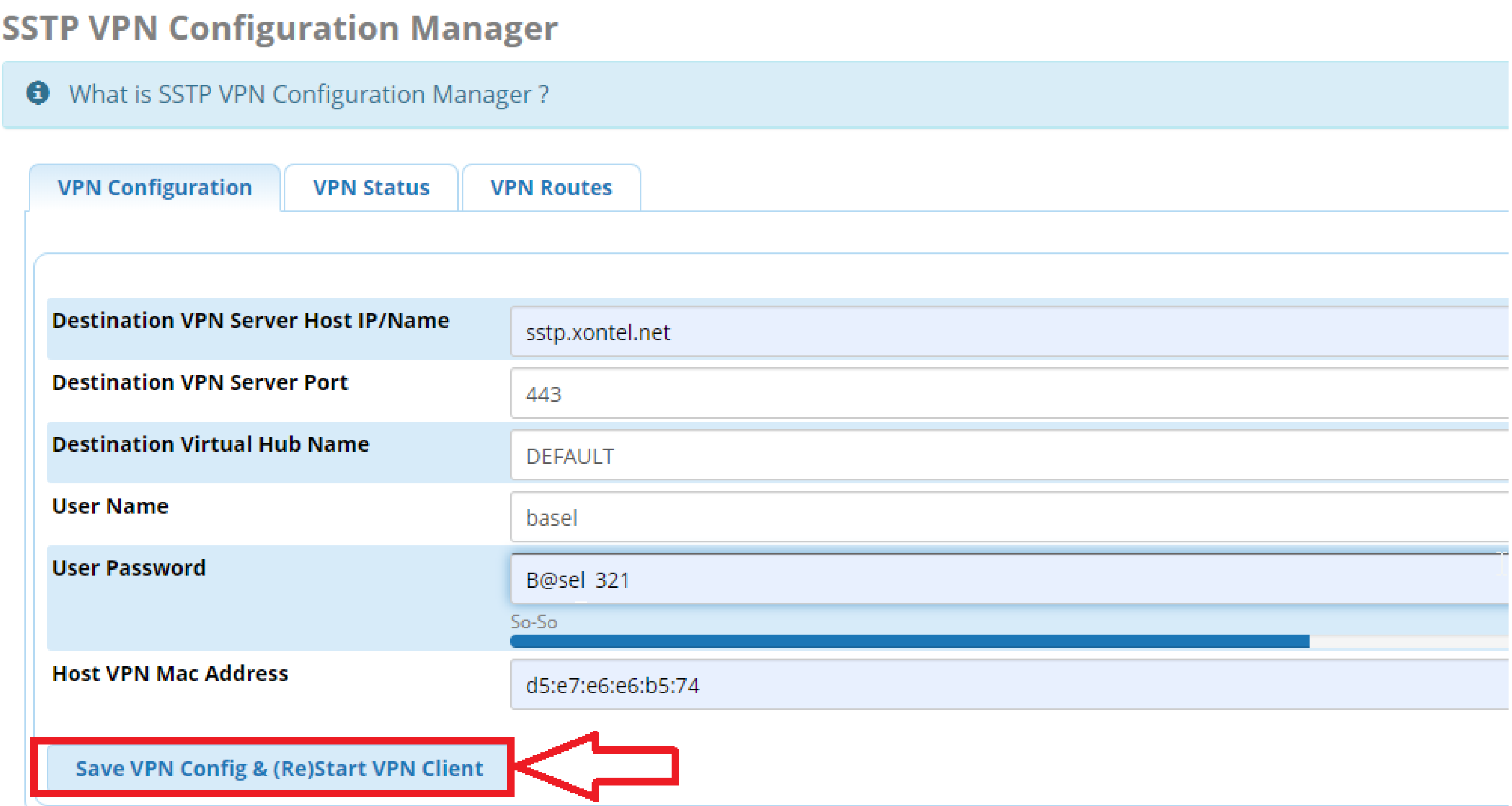
2.7 SSTP VPN Module

From here you can connect XonTel MS PBX with an Open VPN server by uploading Open VPN certificate as shown below

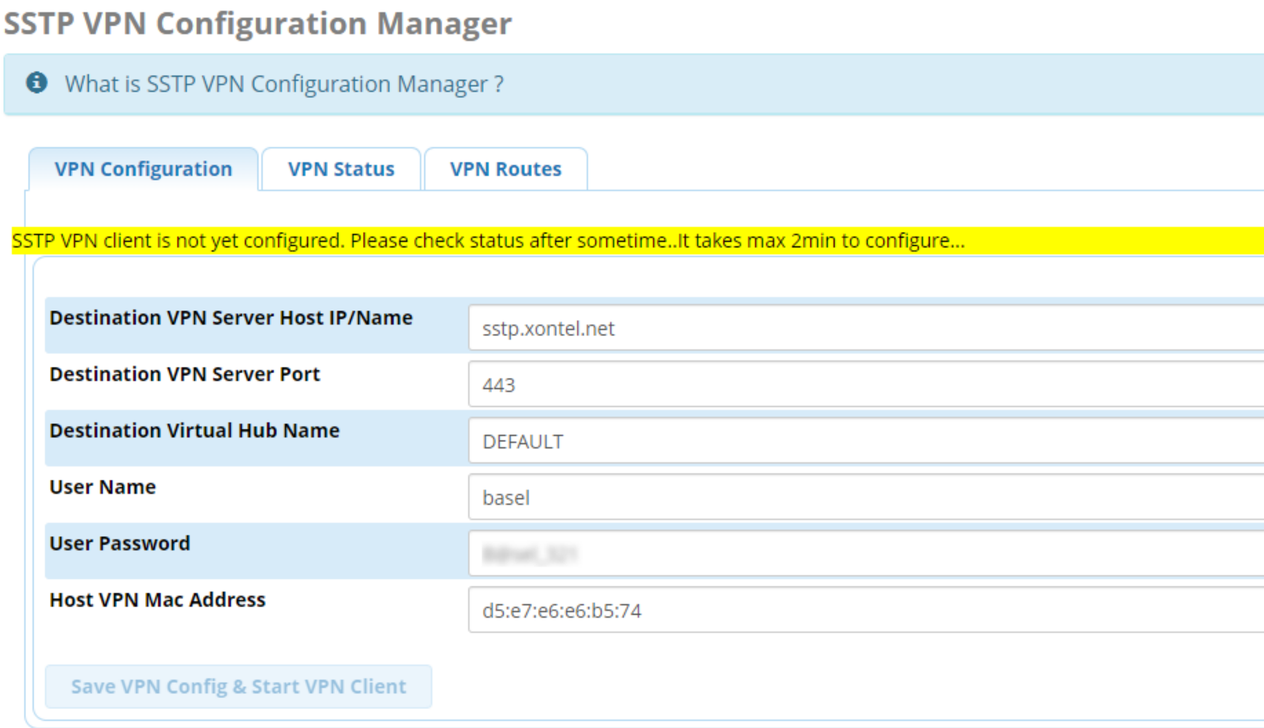
A. Go to " **SSTP VPN Configuration Management** " from " **Admin** " Menu.

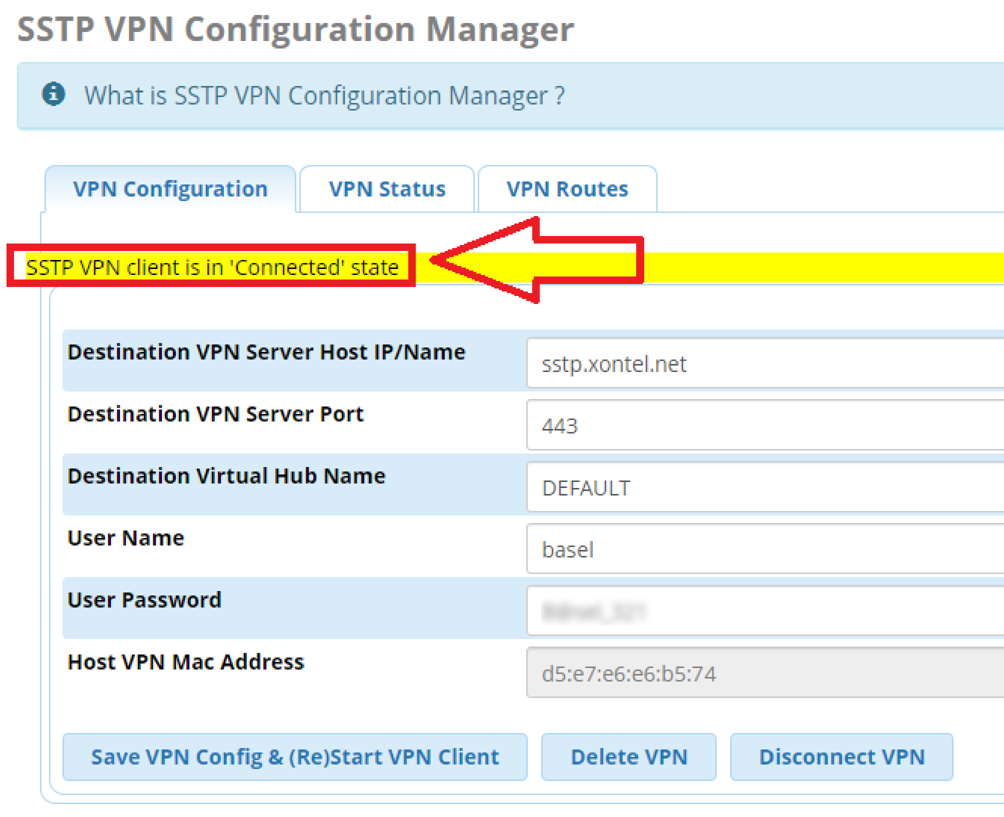


B. Configure your SSTP server, username, password and virtual MAC address of the SSTP connection (if provided) then click “ **Save VPN Config & (Re)Start VPN Client** “ as shown below.

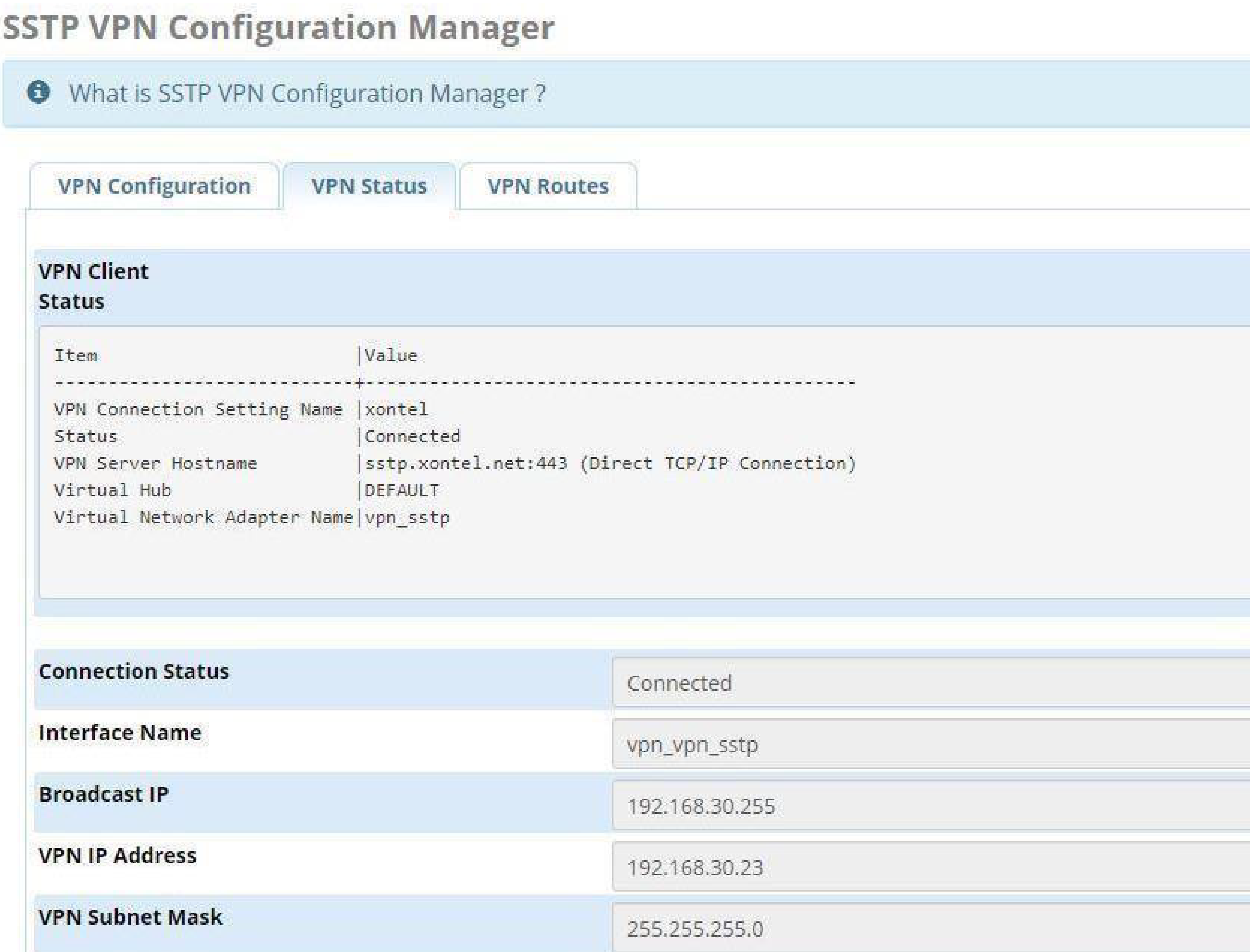


C. Please Wait until you see the message **“ SSTP VPN client is in 'Connected' state “** as shown in the figures below.

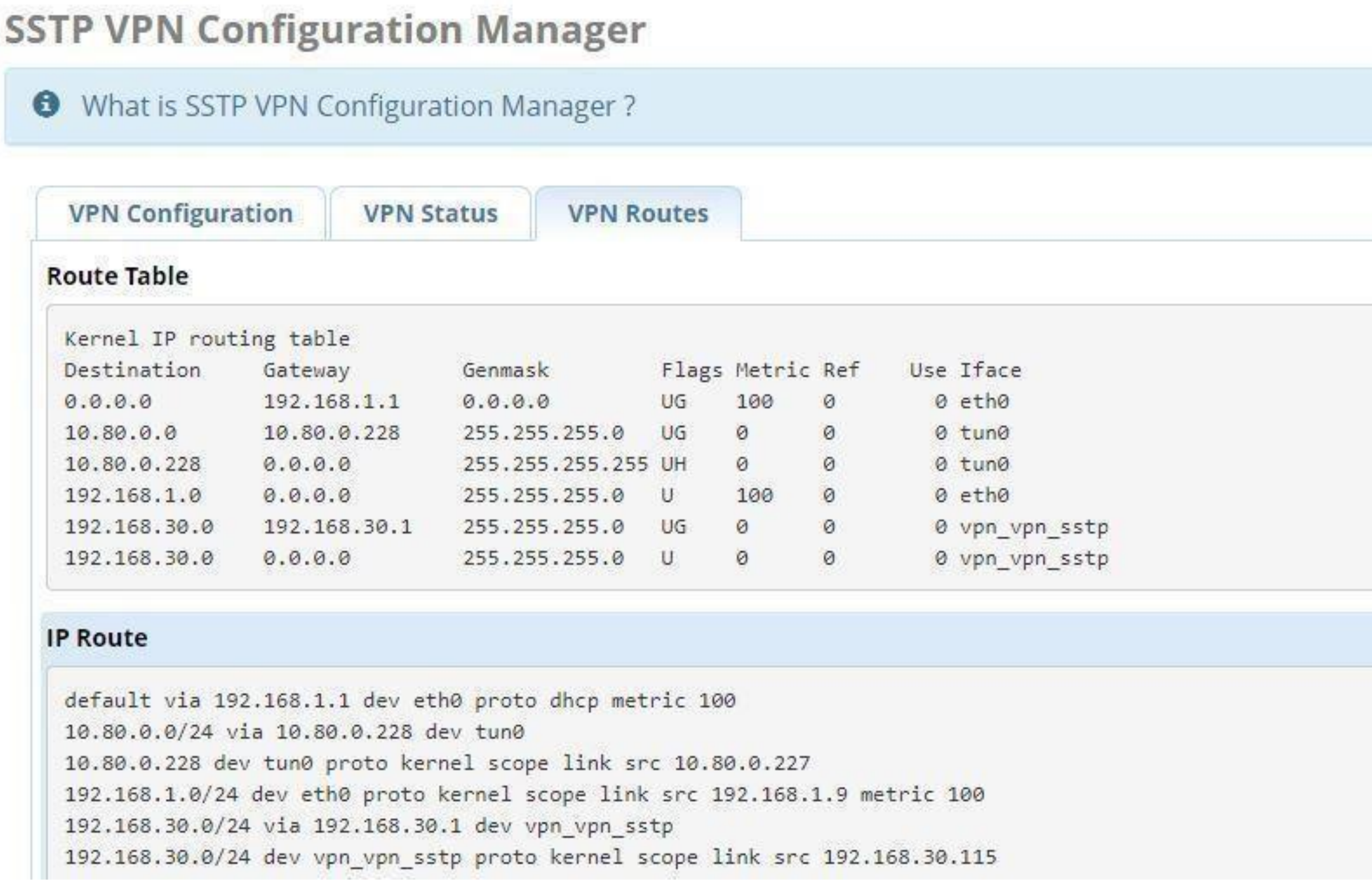




D. Once connection is successfully connected, we can see the status in **VPN Status** tab.



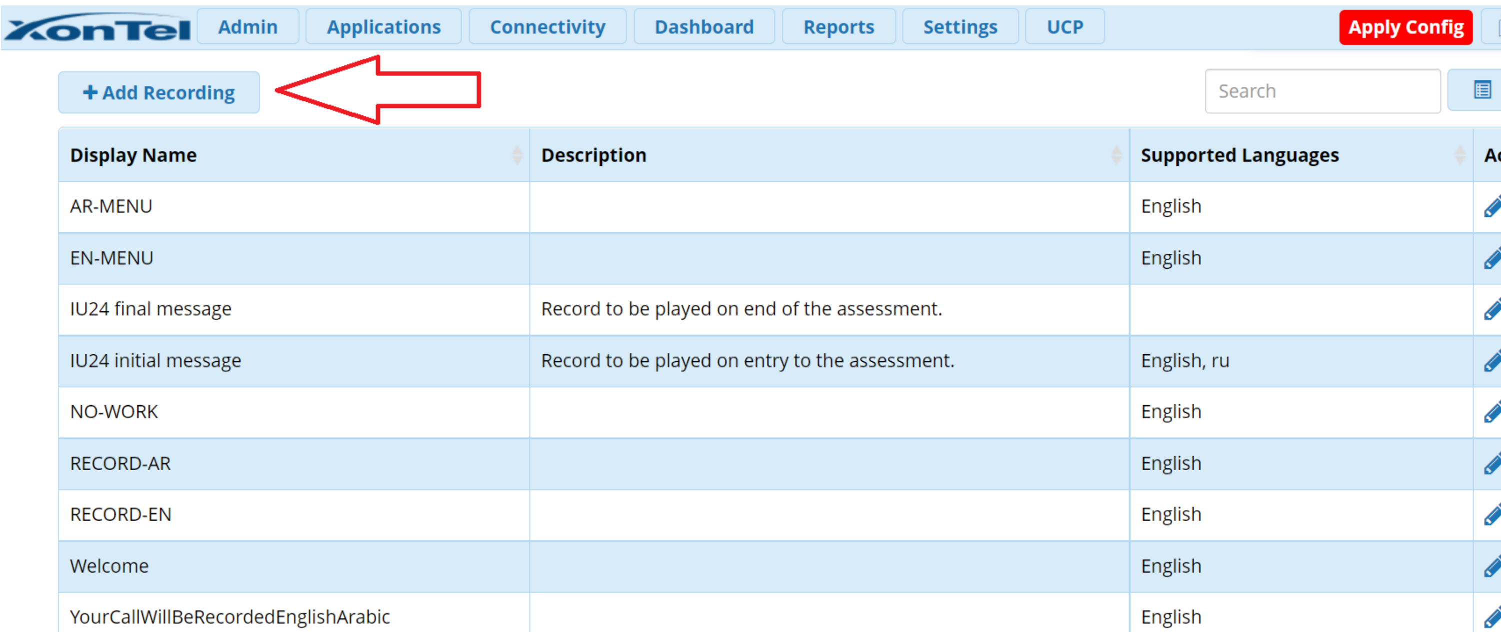
E. **VPN Routes** tab will display the VPN Routes as well as the default route.



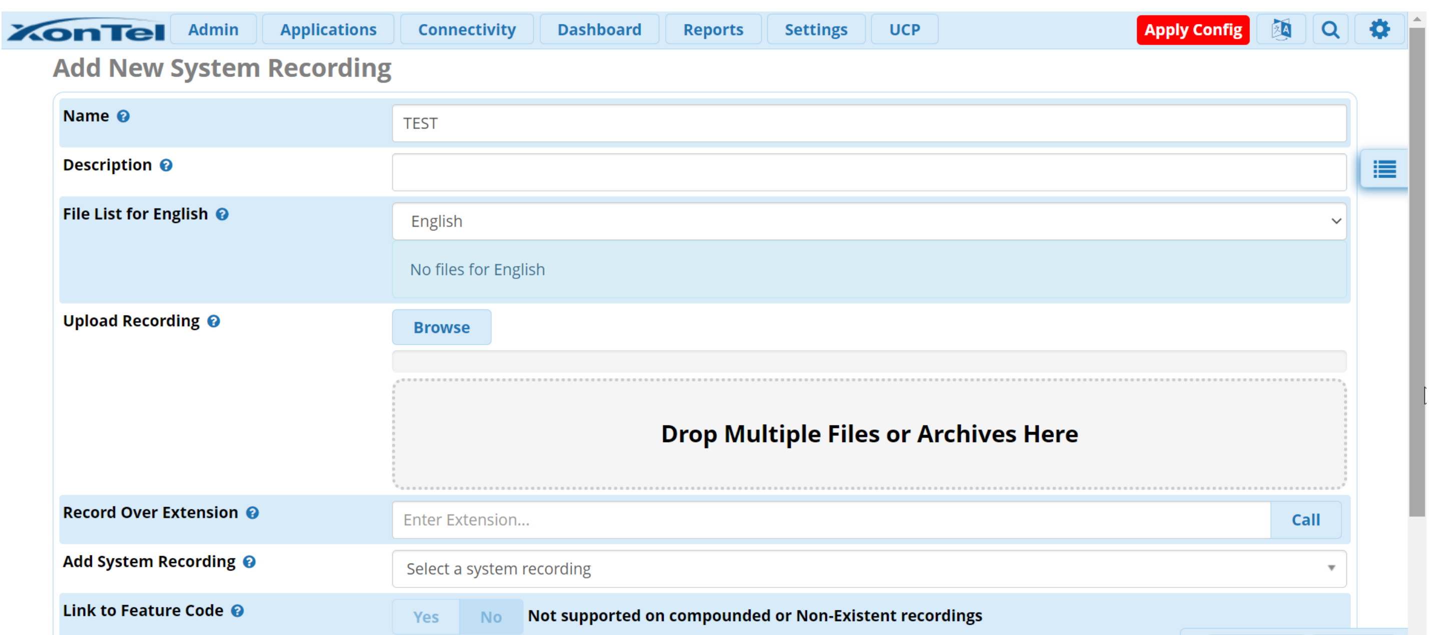
2.8 System Recordings Module

From here you can upload custom prompts to XonTel MS PBX to use it in IVR or in announcements

1. Click **Add Recording** as shown below



2. Upload the prompt from your PC (you can upload multiple prompts in the same system recording)



3. Applications Modules

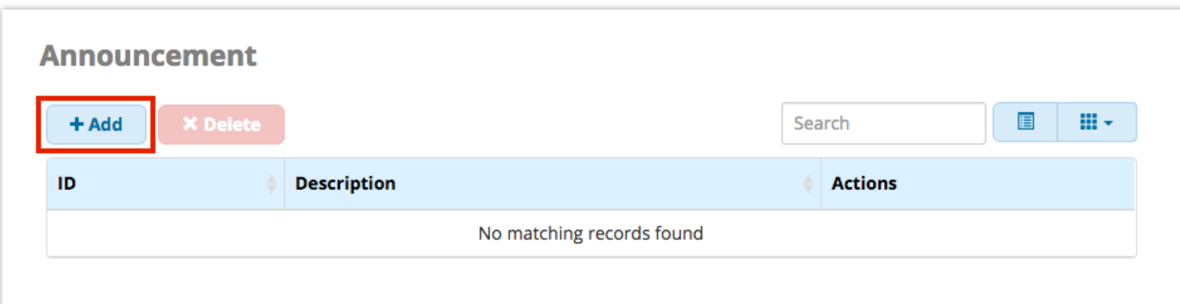
3.1 Announcements

The Announcements module is used to play a recording to callers and then send them to a different destination once the announcement has been played. Do not confuse Announcements with the System Recordings module. The System Recording module is where you create the actual system recordings. The Announcement module just lets you play one of those recordings and continue on with the call flow.

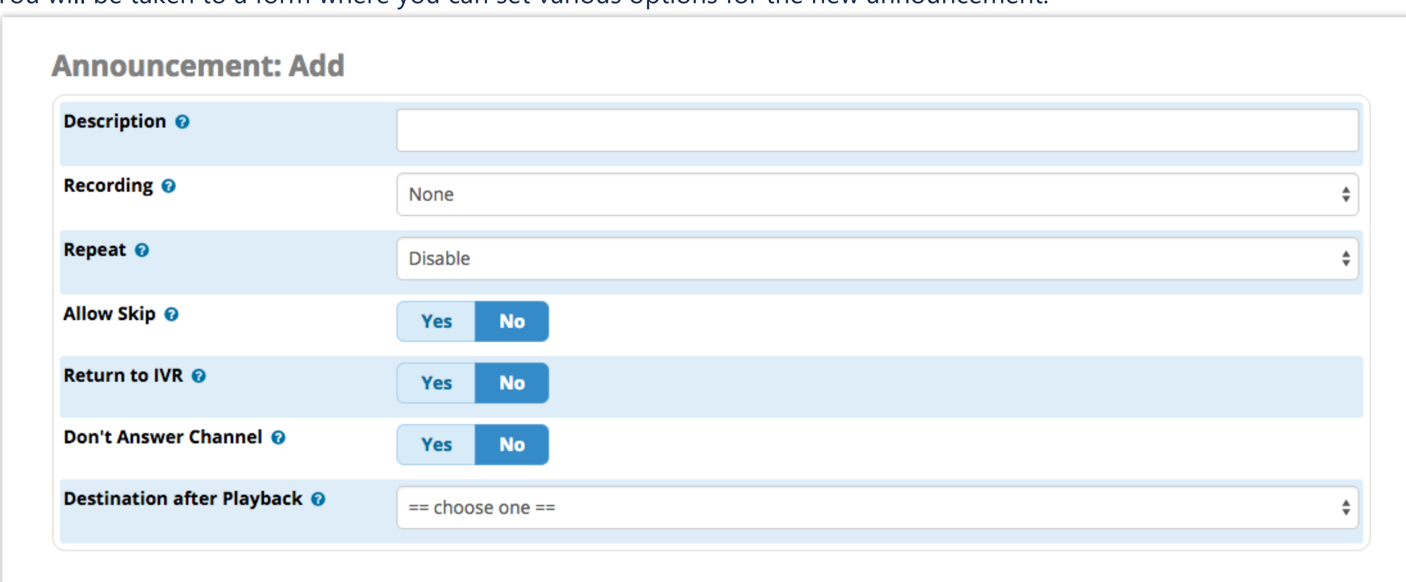
Logging In

- From the top menu click **Applications**
- In the drop down click **Announcements**

- Click the **Add** button.



- You will be taken to a form where you can set various options for the new announcement.



Description

Give the announcement a descriptive name.

Recording

Select the recording to be played. This is the recording that you have created using the System Recording module.

Repeat

You may optionally pick a keypress value from 0-9 or * and # that a caller can press to repeat the announcement. If you use this setting, don't forget to include instructions for the caller in your recording. For example, "To hear our hours again, press pound."

Allow Skip

Yes/No - You can optionally enable the Allow Skip option, which will let the caller press any key on their phone to skip to the end of the recording. They will then go to the destination that is set in this announcement without having to listen to the entire recording.

Return to IVR

Yes/No - If set to **Yes**, a caller who came from an IVR will be sent back to the IVR after the announcement, instead of being sent to the destination set below. This is handy if you have more than one IVR pointing to this announcement, because otherwise you would need to create a separate announcement for each IVR. (A single announcement can only route the caller to one defined destination.) If set to **No**, the caller will only be routed to the destination set below, and will not be sent back to the IVR they came from.

Don't Answer Channel

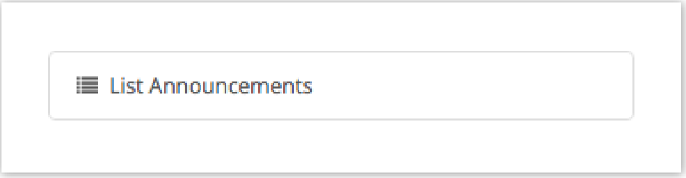
Yes/No - The normal and recommend setting is **No**, which means the behavior is to answer the call and play this message. If you would rather play this message as early media to the caller, you can set this to **Yes**. We do not recommend setting this option, as many phone carriers do not support early media for sending audio messages.

Destination after Playback

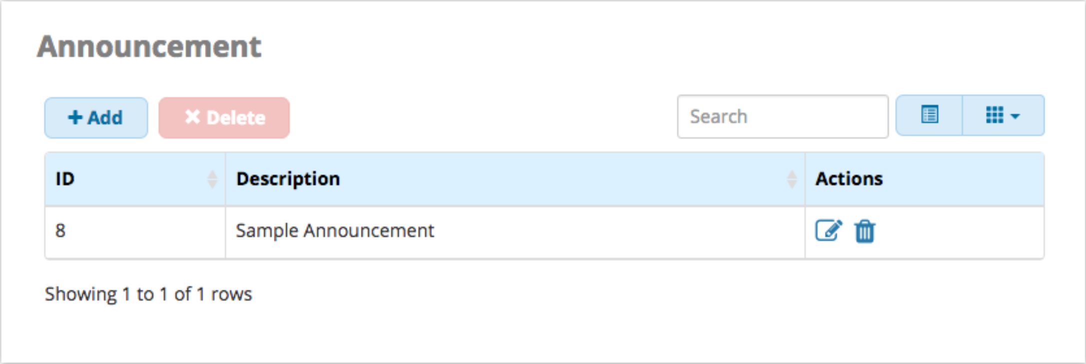
Here you define where to route the caller after they have listened to the message. Remember, this option is ignored if you have set Return to IVR to Yes *and* the caller came from an IVR.

Saving the Announcement

- Click the **Submit** button.
- Click the **Apply Config** button.
- Click **List Announcements** to return to the announcement list.



- Your new announcement should show up in the list.



Editing an Announcement

In the announcement list, click the edit button for the announcement. This will bring up the same form as when creating the announcement.

Make your changes, then click the **Submit** button followed by the **Apply Config** button.

Deleting an Announcement

In the announcement list, click the trash can icon , then click **OK** to confirm the deletion, and click the **Apply Config** button.

Alternatively, when viewing an announcement, click the **Delete** button, click **OK** to confirm the deletion, and click the **Apply Config** button.

3.2 Callback

A callback will hang up on the caller and then call them back, directing them to the selected destination. This means your system will be the originator of the new call, and the other person will be the receiver, instead of the other way around. The callback feature is most commonly used to help callers reduce their mobile and/or international calling fees.

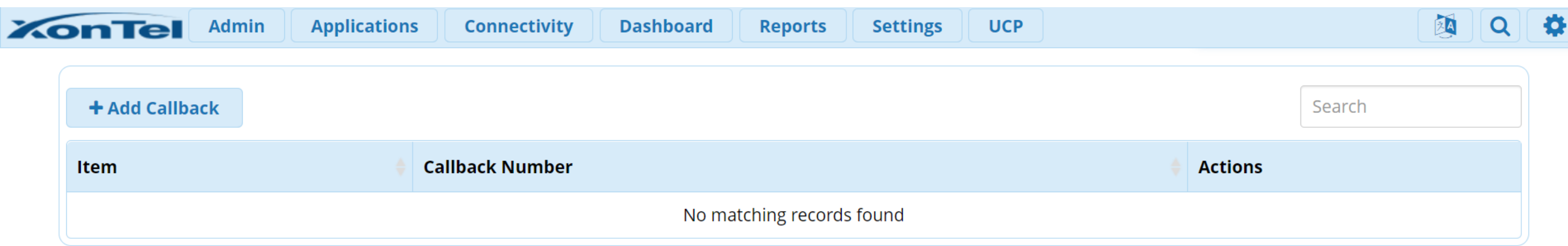
Generally, you would point to a callback from an IVR or directly from an inbound route, but you can point to the callback from any module of your PBX.

Logging in

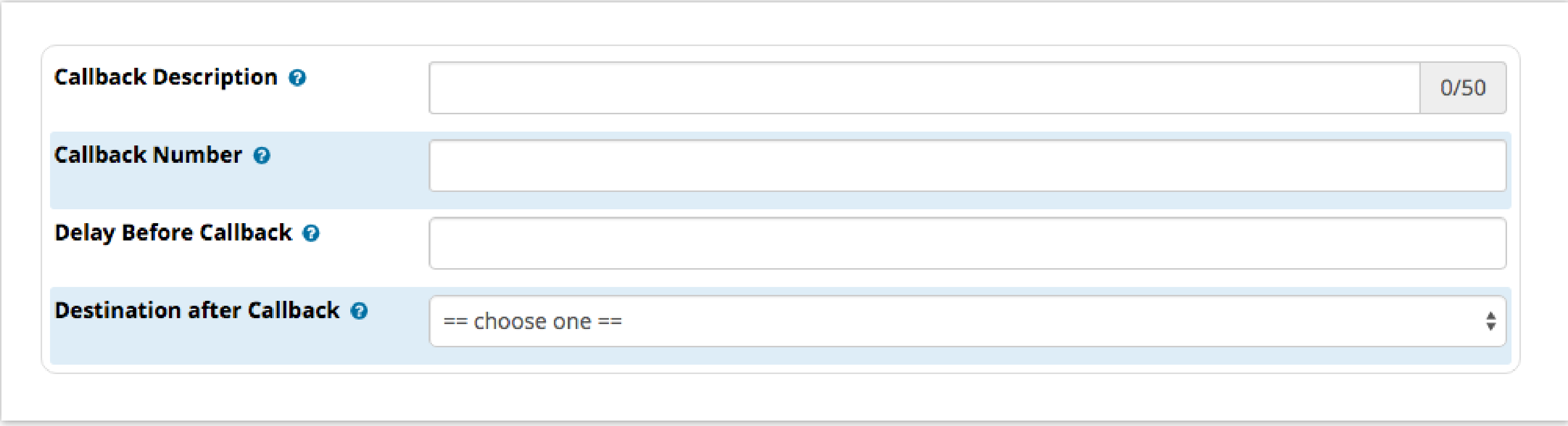
- From the top menu click **Applications**
- In the drop down click **Callback**

Creating a Callback

- Click the **Add Callback** button.



- Fill in the description, number, delay, and destination fields as described below.



Callback Description

Give the callback a descriptive name.

Callback Number

(Usually left blank.) You can optionally hard code the callback number that you will call back. If you leave this blank, the system will call the Caller ID that was received.

Delay Before Callback

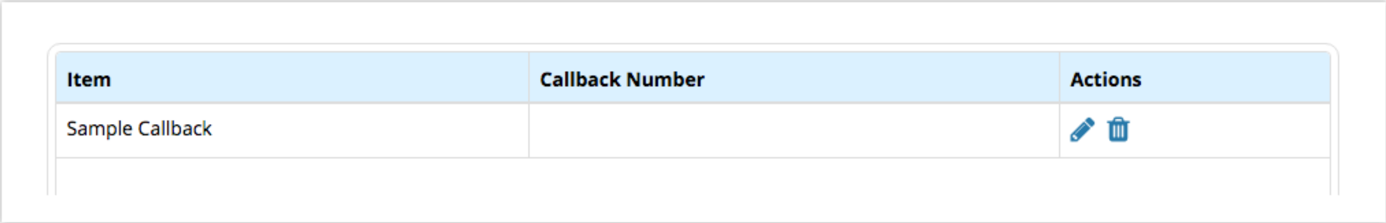
You can set a delay, in seconds, for how long to wait before calling back the caller. Leaving it blank will default to "No Delay."

Destination after Callback

Where to send the caller after the system calls them back. The call proceeds as if the caller had just called in and reached this destination.

Save

- Click the **Submit** button when done.
- Click the **Apply Config** button to apply your changes.
- Callbacks will be displayed in a table.



Item	Callback Number	Actions
Sample Callback		

- **To view/edit:** Click the pencil icon for the callback.
- **To delete:** Click the trash icon for the callback, click **OK** to confirm the deletion, and click the **Apply Config** button.

3.3 Conferences

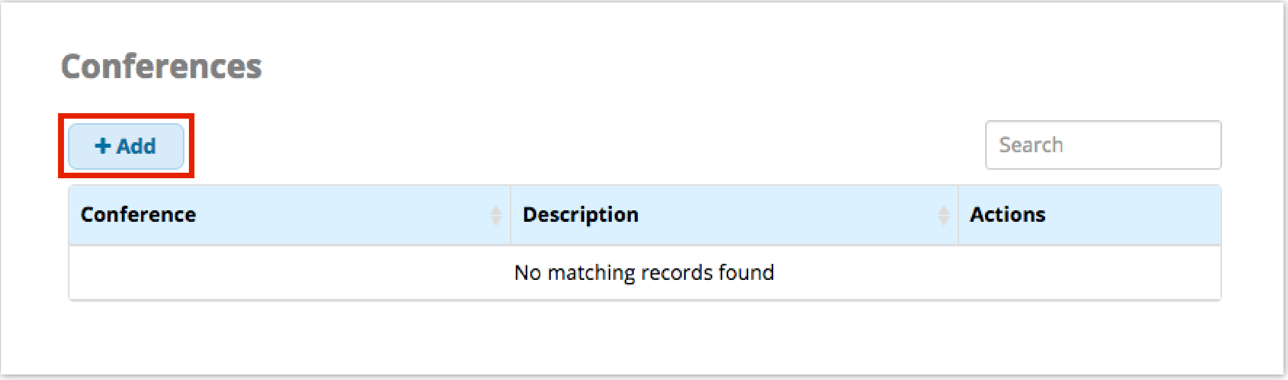
The Conferences Module is used to create a single extension number that your users can dial so that they can talk to each other in a conference call. It also creates a destination to which you can send calls so that they can participate in the conference call. For example, you could create a Conference that will allow your local phones to dial 800, and then enter into a conference call.

Logging In

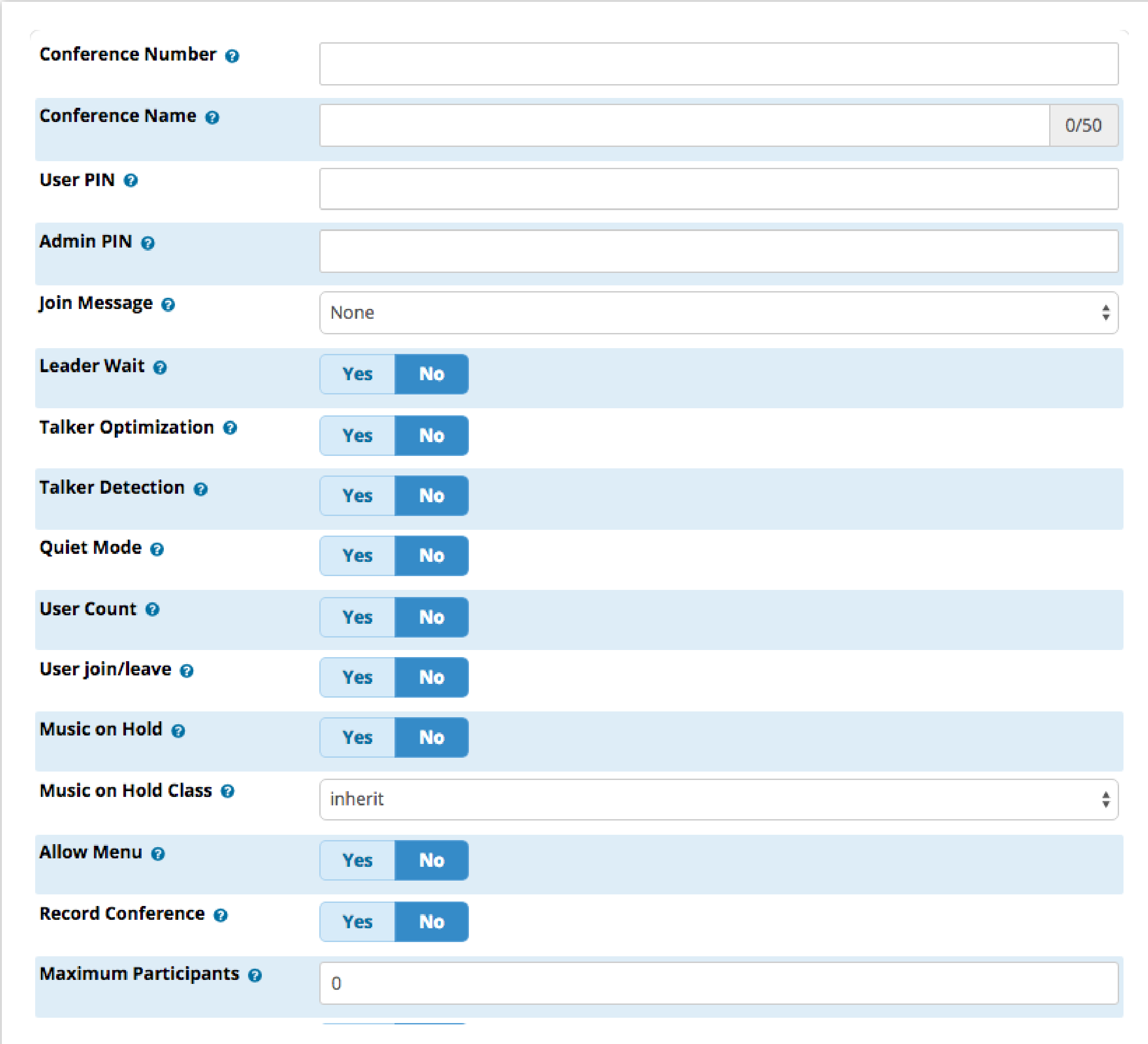
- From the top menu click **Applications**
- In the drop down click **Conferences**

Creating a Conference Room

- Click the **Add** button.



- Fill in the information on the form, as described below.



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

Optional - You can require callers to enter a password before they can enter this conference. If either PIN is entered, the user will be prompted to enter a PIN. The user PIN should be different from the admin PIN.

Admin PIN

Optional unless the "leader wait" option is in use - Enter a PIN number for the admin user. When a user enters this PIN, he/she will be identified as the conference leader.

Join Message

Message to be played to the caller before joining the conference. Default = none. The drop-down menu allows you to select a recording that has been created in the System Recordings module.

Leader Wait

Yes/No - Whether to wait until the conference leader (admin user) arrives before starting the conference.

Talker Optimization

Yes/No - Whether to use talker optimization. With talker optimization, Asterisk treats talkers who are not speaking as being muted. This means that no encoding is done on transmission, and that received audio that is not registered as talking is omitted, preventing buildup in background noise.

Talker Detection

Yes/No - Whether to use talker detection. With talker detection, Asterisk will send events on the Manager Interface identifying the channel that is talking. The talker will also be identified on the output of the meetme list CLI command.

Quiet Mode

Yes/No - Whether to use quiet mode. If quiet mode is enabled, enter/leave sounds will not be played.

User Count

Yes/No - Whether to announce the user count to a user who joins the conference.

User Join/Leave

Yes/No - Whether to announce user join/leave. If this option is enabled, all users will be asked to say their name before they join the conference, and their name will be then announced when they join the conference.

Music on Hold

Yes/No - Whether to enable Music On Hold when the conference has a single caller.

Music on Hold Class

Select the music (or Commercial) played to the caller while they wait for the conference to start. Choose "inherit" if you want the MoH class to be what is currently selected, such as by the inbound route. This music is defined in the Music on Hold module.

Allow Menu

Yes/No - Whether to present a menu (user or admin) when '*****' is received ('send' to menu)

Record Conference

Yes/No - Whether to record the conference call.

Maximum Participants

Enter the maximum number of users allowed to join this conference. Enter "0" for unlimited.

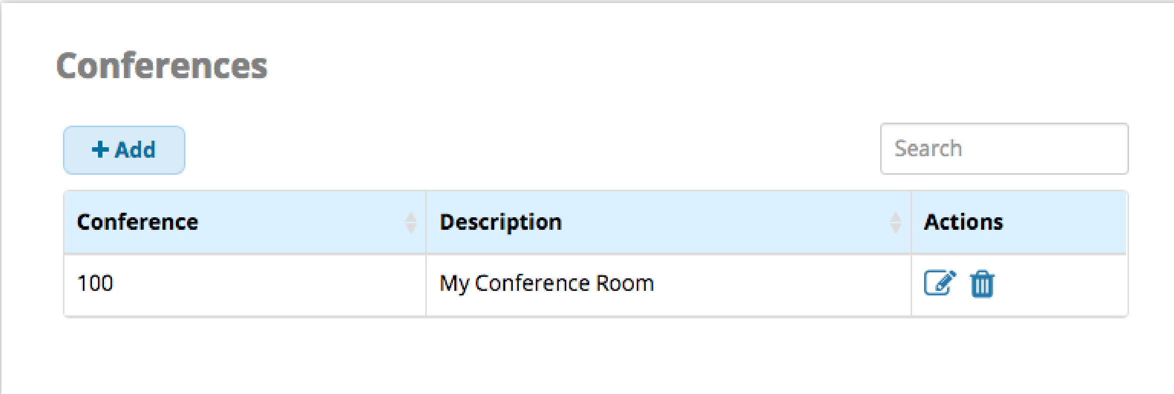
Mute on Join

Yes/No - Whether to mute everyone when they initially join the conference. Please note that if you do not have "Leader Wait" set to "Yes," you will need to have "Allow Menu" set to "Yes" to be able to un-mute yourself.

Save

- Click the **Submit** button.
- Click the **Apply Config** button.

Editing, and Deleting Conferences



- To **view or edit** a conference, click the pencil icon .
- To **delete** a conference, click the trash icon .

3.4 DISA

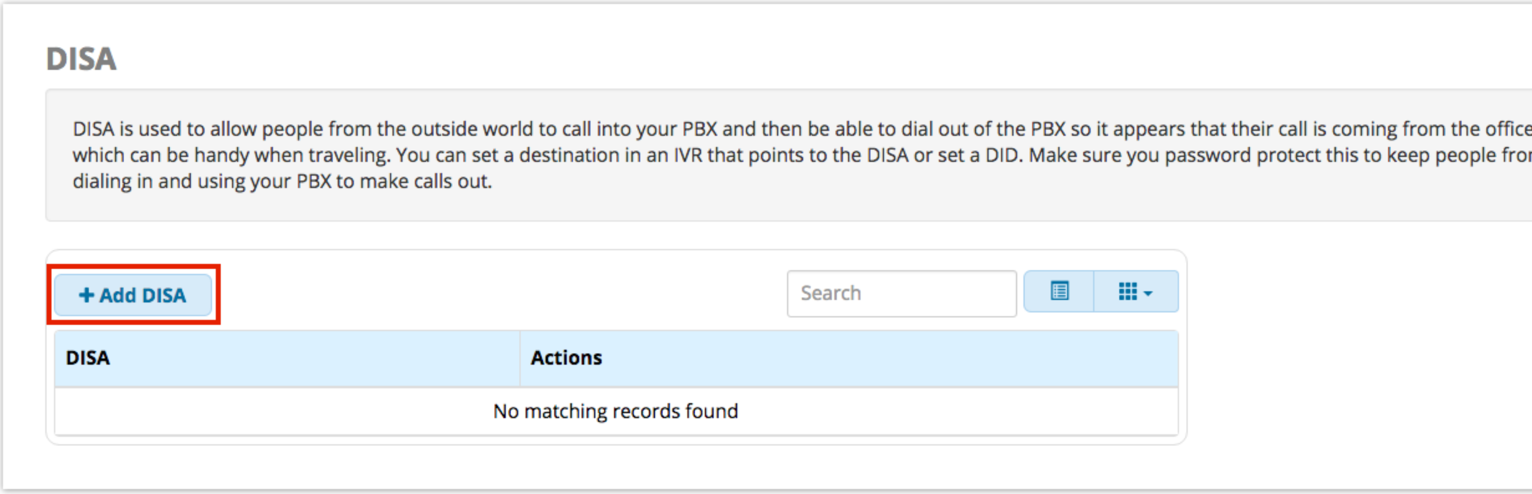
DISA is used to allow people from the outside world to call into your PBX and then be able to dial out of the PBX so it appears that their call is coming from the office, which can be handy when traveling. You can set a destination in an IVR that points to the DISA or set a DID to point to the DISA. Make sure your password-protect this to keep unauthorized people from dialing in and using your PBX to make calls.

Logging In

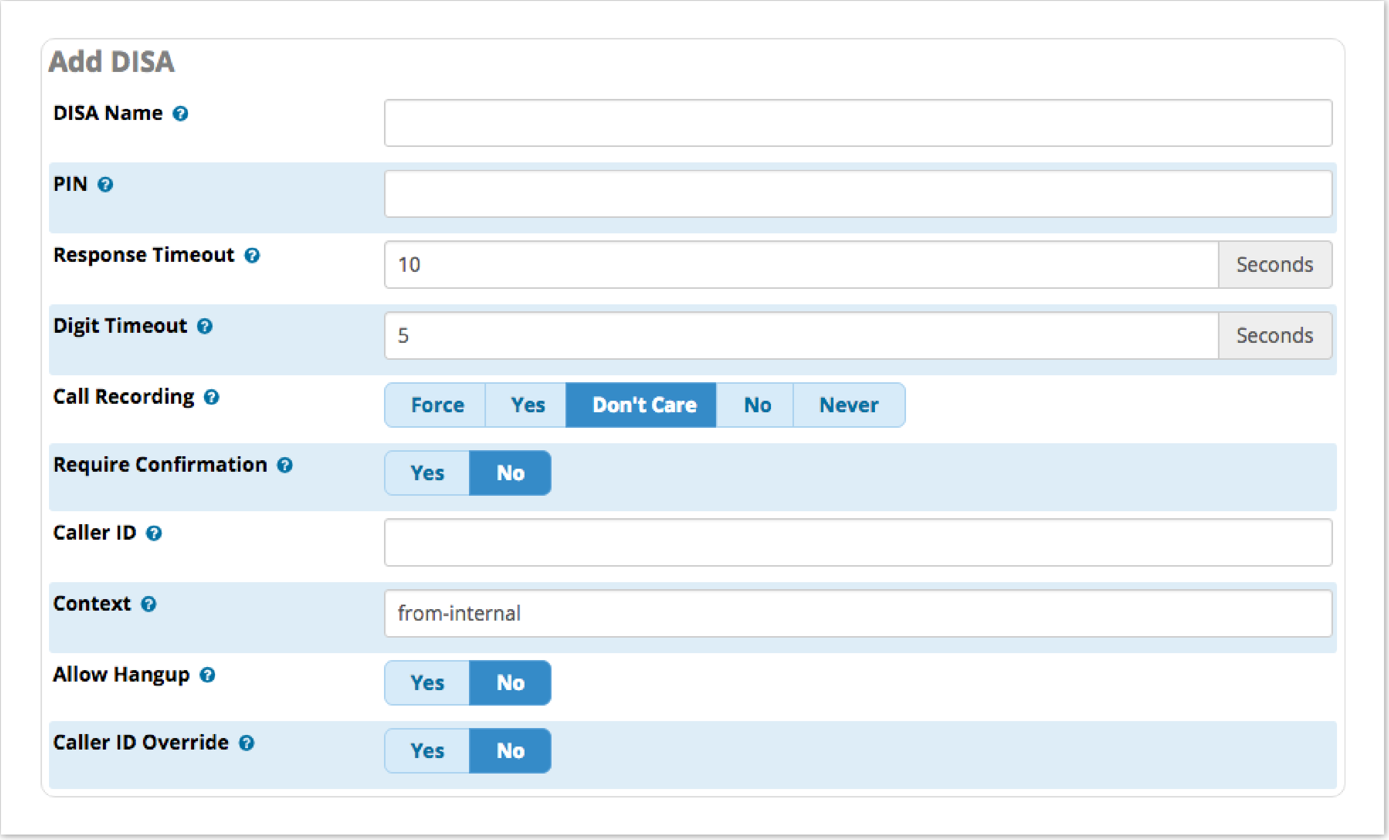
- From the top menu click **Applications**
- From the drop down click **DISA**

Adding a DISA

- Click the **Add DISA** button.



- Fill out the information on the form, as described below.



DISA Name

Give the DISA a brief name to help you identify it.

PIN

The user will be prompted for this number. If you wish to use multiple PINs, separate them by commas.

Response Timeout

The maximum amount of time it will wait before hanging up if the user has dialed an incomplete or invalid number. Default is 10 seconds.

Digit Timeout

The maximum amount of time permitted between digits. Default is 5 seconds.

Call Recording

Force/Yes/Don't Care/No/Never: Whether to record calls in the DISA. See [Call Recording walk through] to learn about the options.

Require Confirmation

Yes/No: Whether to require confirmation before prompting for a password. Used when your PSTN connection appears to answer the call immediately.

Caller ID

This setting is optional. When using this DISA, the user's caller ID will be set to this. Format is "User Name" <5551234>

Context

This should be touched by experts only. Sets the context that calls will originate from. Leave this as “from-internal” unless you know what you are doing.

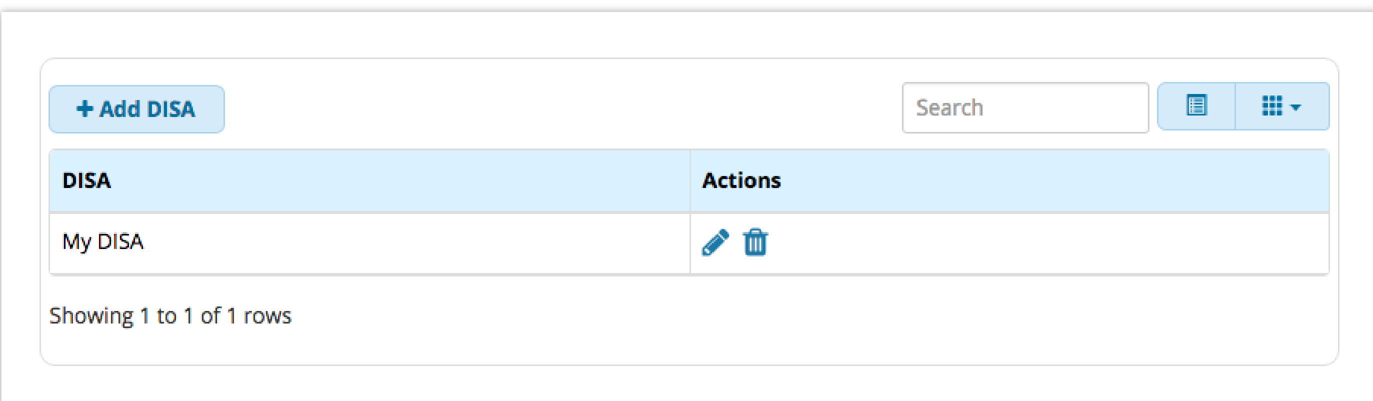
Allow Hangup

Yes/No: Whether pressing the hangup feature code (**) will disconnect the call and present a dial tone for a new call.

Save

- Click the **Submit** button.
- Click the **Apply Config** button.

Editing, or Deleting a DISA

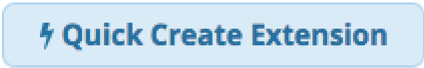


- **To view/edit:** Click the pencil icon .
- **To delete:** Click the trash icon .

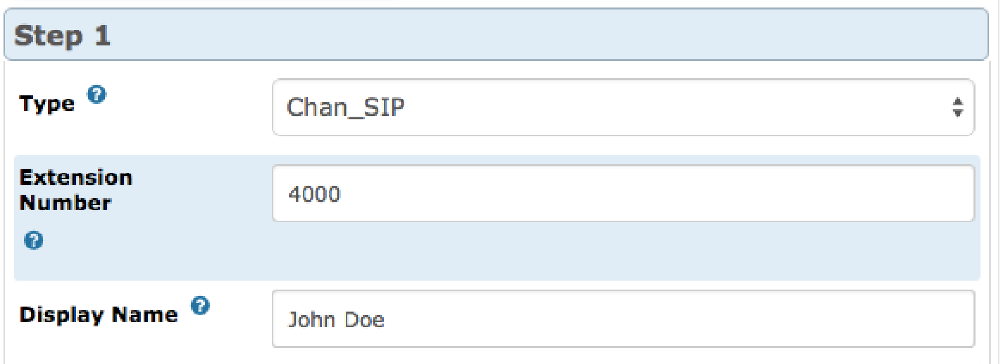
3.5 Extensions

Creating Extensions

Go to **Applications** -> **Extensions**
For the fastest, easiest setup, click the **Quick Create Extension** button.



Select the desired type of new extension, enter the extension number, and enter the display name. *In our example we are creating a new Chan_SIP extension.*



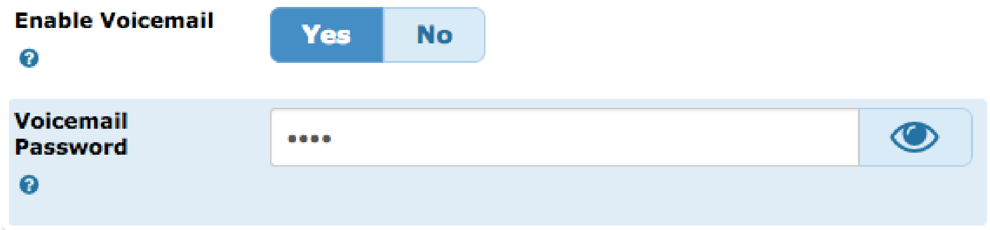
Optional: Enter an e-mail address for the user. This e-mail will be used for services such as voicemail, User Control Panel, and fax.

Click **Next** to go to step 2.

Optional: If you would like to enable voicemail now, do the following:

Click the **Yes** button next to "Enabled." This will make the password field available.

Enter an initial password (digits only). We recommend initially setting this to the extension number, because the first time the user dials *98, they will be prompted to set up their voicemail box and change this password. Users can also change their passwords later by dialing *97 and changing voicemail settings.



Click the **Finish** button.

Click the **X** button ( ) in the upper right-hand corner of the window to close the Quick Create window.
Reload the extensions list page to see your newly created extension in the list.
Repeat the process for each extension you would like to add.
To manage the extensions you created, go to Applications > Extensions -> List Extensions. Here you can see a table showing extensions and whether various settings are enabled. You can click the pencil icon ( ) to edit, or the trash can icon ( ) to delete.

For example: pjsip extension

Extensions Module - PJSIP Extension

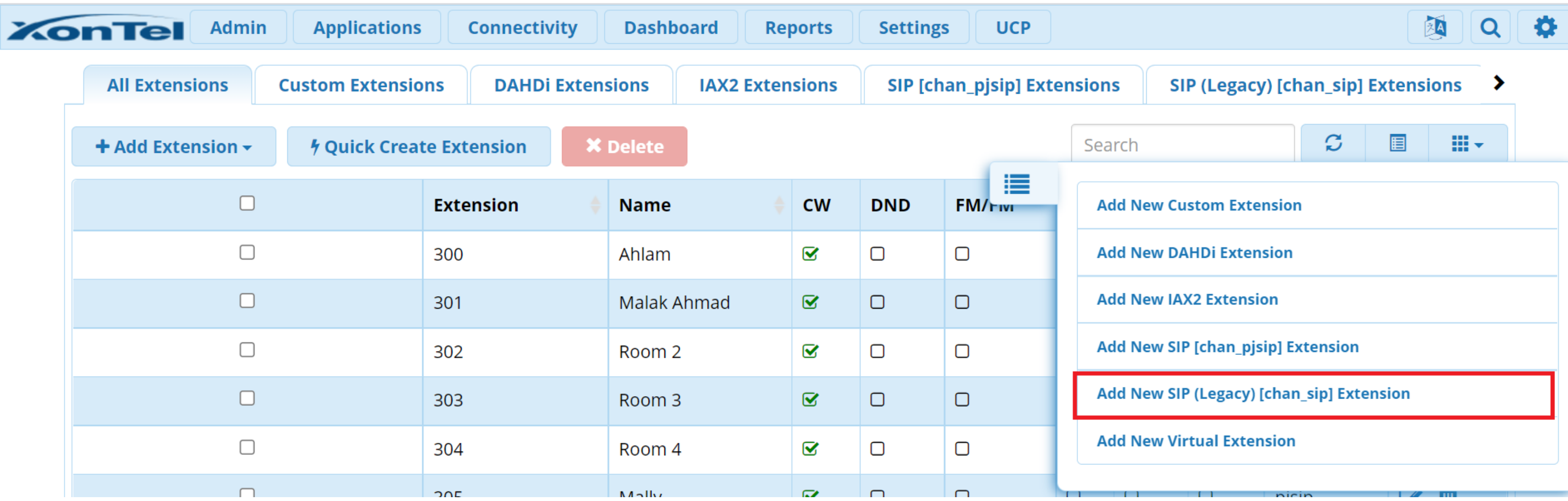
Extensions Module - SIP Extension

Logging in

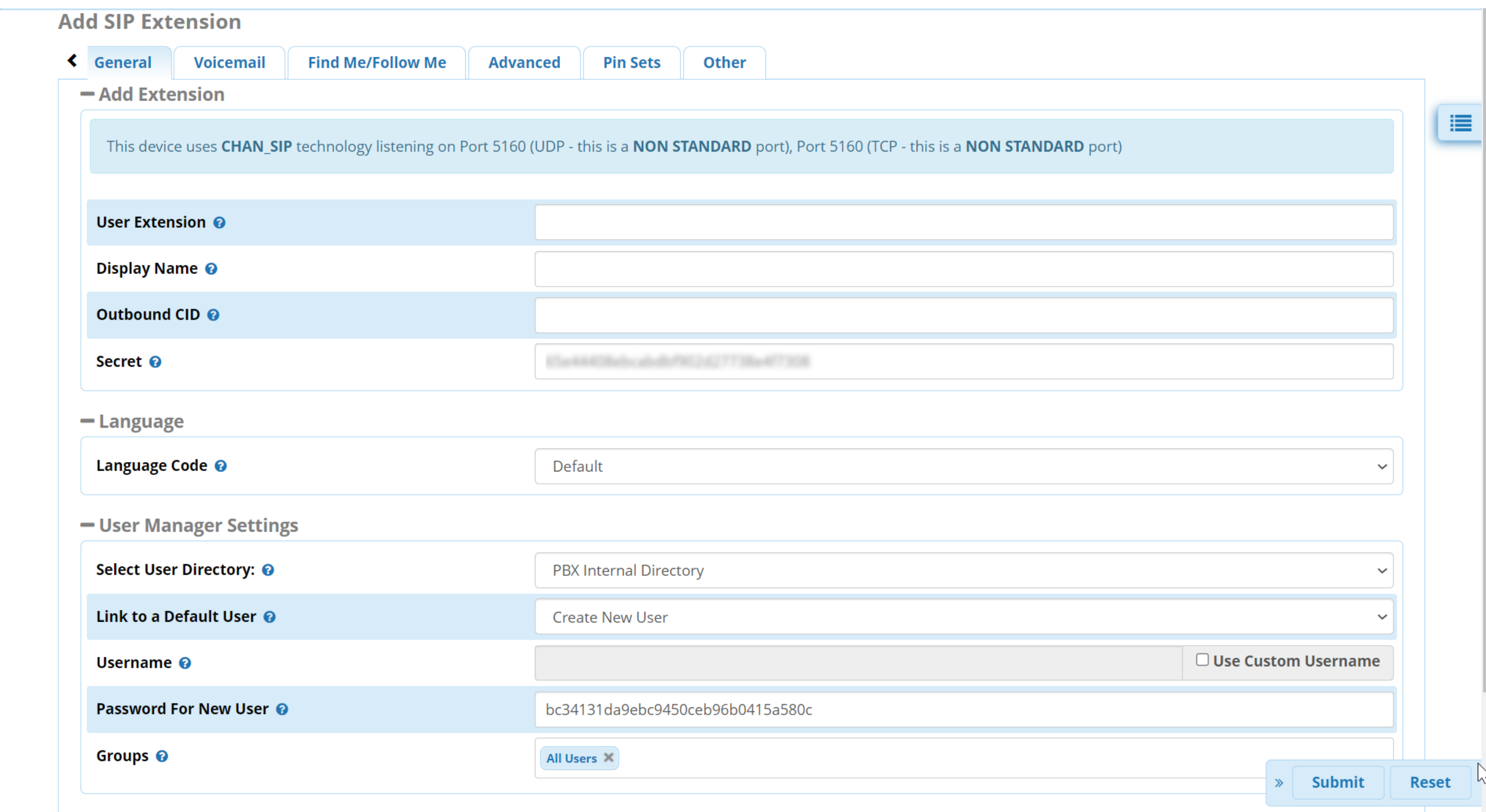
From the top menu click **Applications**
From the drop down click **Extensions**

Adding a SIP Extension

From the Extensions landing page click on the **Add SIP (Legacy) [chan_sip] Extension** button.



General



Add Extension

User Extension

This will be the extension number associated with this user and cannot be changed once saved. We recommend using 3- or 4- digit extension numbers.

Display Name

This is the name associated with this extension and can be edited any time. This will become the Caller ID Name. Only enter the name, NOT the number.

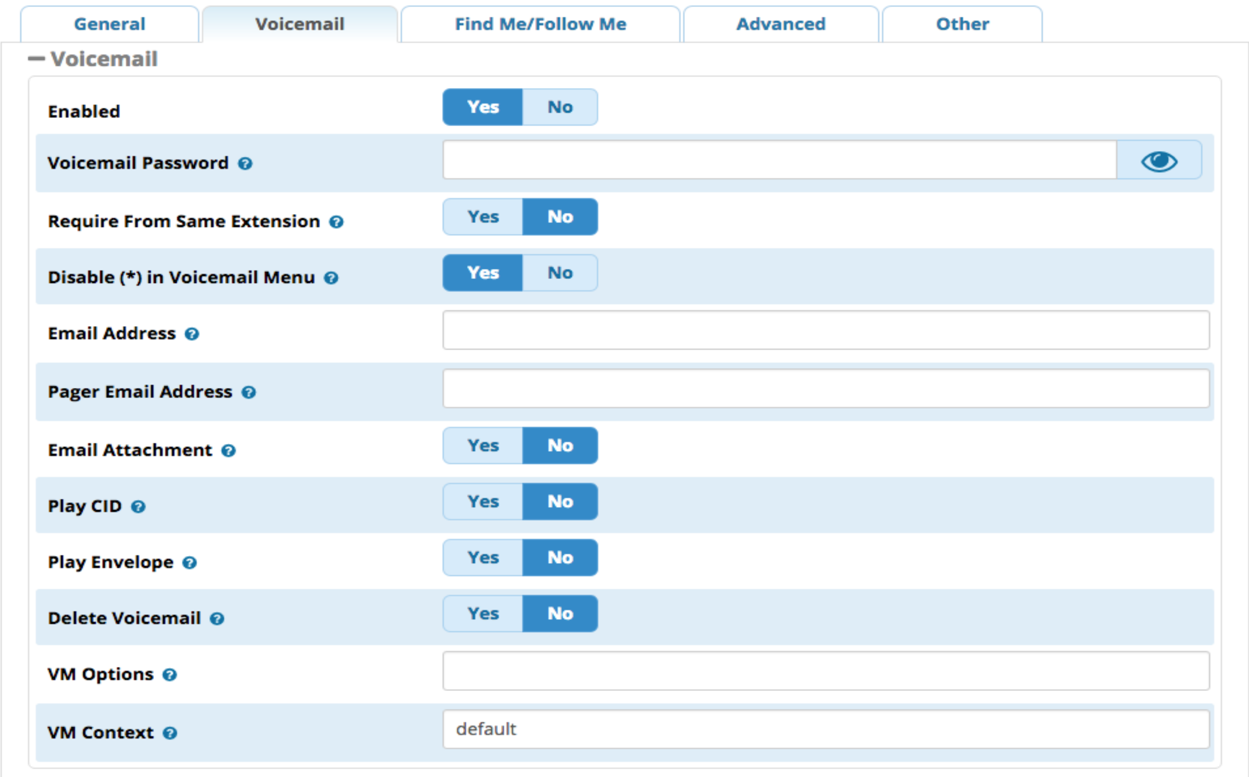
Outbound CID

Overrides the CallerID when dialing out a trunk. Any setting here will override the common outbound CallerID set in the Trunks module. Format: **"caller name"** <#######>

Leave this field blank to disable the outbound CallerID feature for this user. If you leave it blank, the system will use the route or trunk Caller ID, if set.

Secret

Password (secret) configured for the device. Should be alphanumeric with at least 2 letters and numbers to keep secure. A secret is auto-generated but you may edit it. A color-coded bar will display the strength of the secret, ranging from "really weak" to "



Voicemail

Enabled

Yes/No: Whether to enable voicemail for the user.

Voicemail Password

Enter the password (numbers only) the user will use to access the voicemail system. If left blank, it will default to the extension number. The user can change the password after logging into the voicemail system (*98) with a phone.

Require From Same Extension

Yes/No: Whether to require the user to enter their password after they reach the voicemail system from their own extension, by dialing *97. This option does not apply to *98 calls, which will always prompt for a password. For security purposes, a **Yes** setting is recommended in an environment where other users will have physical access to this extension.

Disable (*) in Voicemail Menu

Yes/No: Whether to disable access to the voicemail menu. Default = Yes. If set to **Yes**, a user will not be able to access the voicemail menu by pressing "*". If you have no plans to access your mailbox remotely, set this to **Yes.** If set to **No**, the user can access voicemail remotely by calling into their extension and pressing "*" to reach the menu.

Email Address

Optional - The e-mail that voicemail notifications will be sent to. Further down the page, you have the option of whether to attach the actual voicemail message to the e-mail.

Pager Email Address

Optional - A pager e-mail address or mobile email address that short voicemail notifications will be sent to.

Email Attachment

Yes/No: Whether to attach the voicemail to the e-mail notification. Requires an email address to be set above.

Play CID

Yes/No: Whether to read back the caller's telephone number prior to playing the voicemail, just after announcing the date and time the message was left.

Play Envelope

Yes/No: Whether the system will play the message envelope information (date/time) before playing the voicemail message. This setting does not affect the operation of the envelope option in the advanced voicemail menu.

Delete Voicemail

Yes/No: Whether to delete the voicemail message from the mailbox after it is e-mailed to the user. If set to **Yes**, this would provide functionality that allows users to receive their voicemail via e-mail alone, rather than needing to retrieve it from the web interface or a telephone.

If **Delete Voicemail = Yes**, then you MUST set an e-mail address for the user above, and also set **Email Attachment = Yes**. Otherwise, the voicemail message would be lost forever, because it would not be e-mailed, and would be deleted from the system.

VM Settings

Optional: Advanced settings. Enter voicemail options, separated by the pipe symbol (|). For example, "review=yes | maxmessage=60" May be left blank.

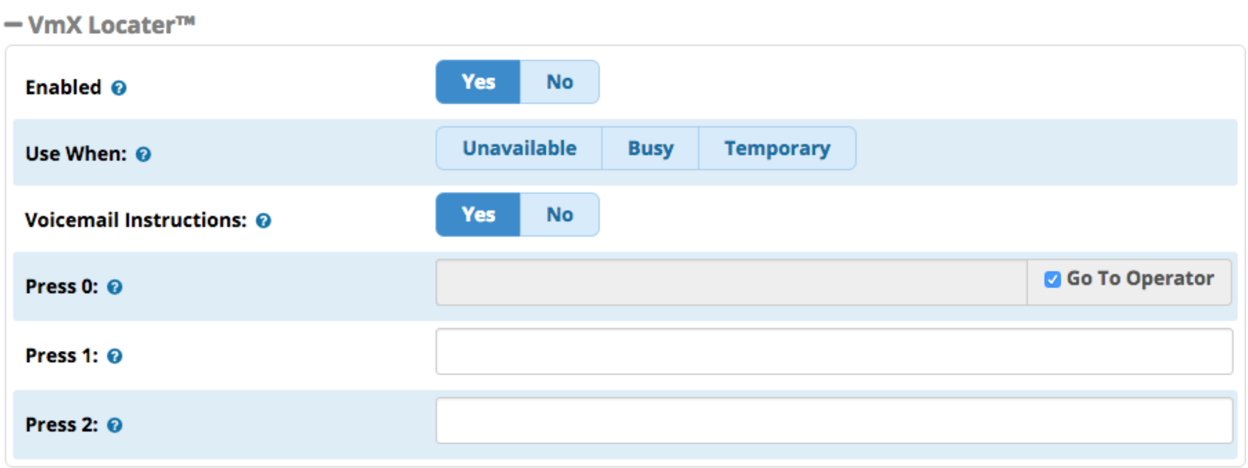
VM Context

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

VMX Locater™

VMX Locater is designed to help a caller reach an operator and/or find you when you are not at your main phone. If enabled, the user will want to consider recording voicemail greetings that instruct a caller on which options to press (0, 1, and/or 2).

Whenever you enter information into the 0, 1, and/or 2 options below, you should run a test to make sure the number is functional, because otherwise the caller might become stranded or receive messages about a number being invalid.



Enabled

Yes/No: Whether to enable the VMX Locater feature. Set to **Yes** if you would like to enable this feature and edit the options below.

Use When

Select one or more of the buttons to enable VMX Locater for these types of greetings: **Unavailable, Busy,** and/or **Temporary.**

Voicemail Instructions

Yes/No: Whether to play instructions after playing your greeting. If set to **No**, only a beep will be played after your personal voicemail greeting.

Press 0

Check the **Go to Operator** box to send the caller to the operator when they press 0. Uncheck the **Go to Operator** box and enter an alternative destination if you want the caller to be sent to a different destination when they press 0. This feature is still accessible to callers even when VMX Locater is disabled for the user.

Press 1

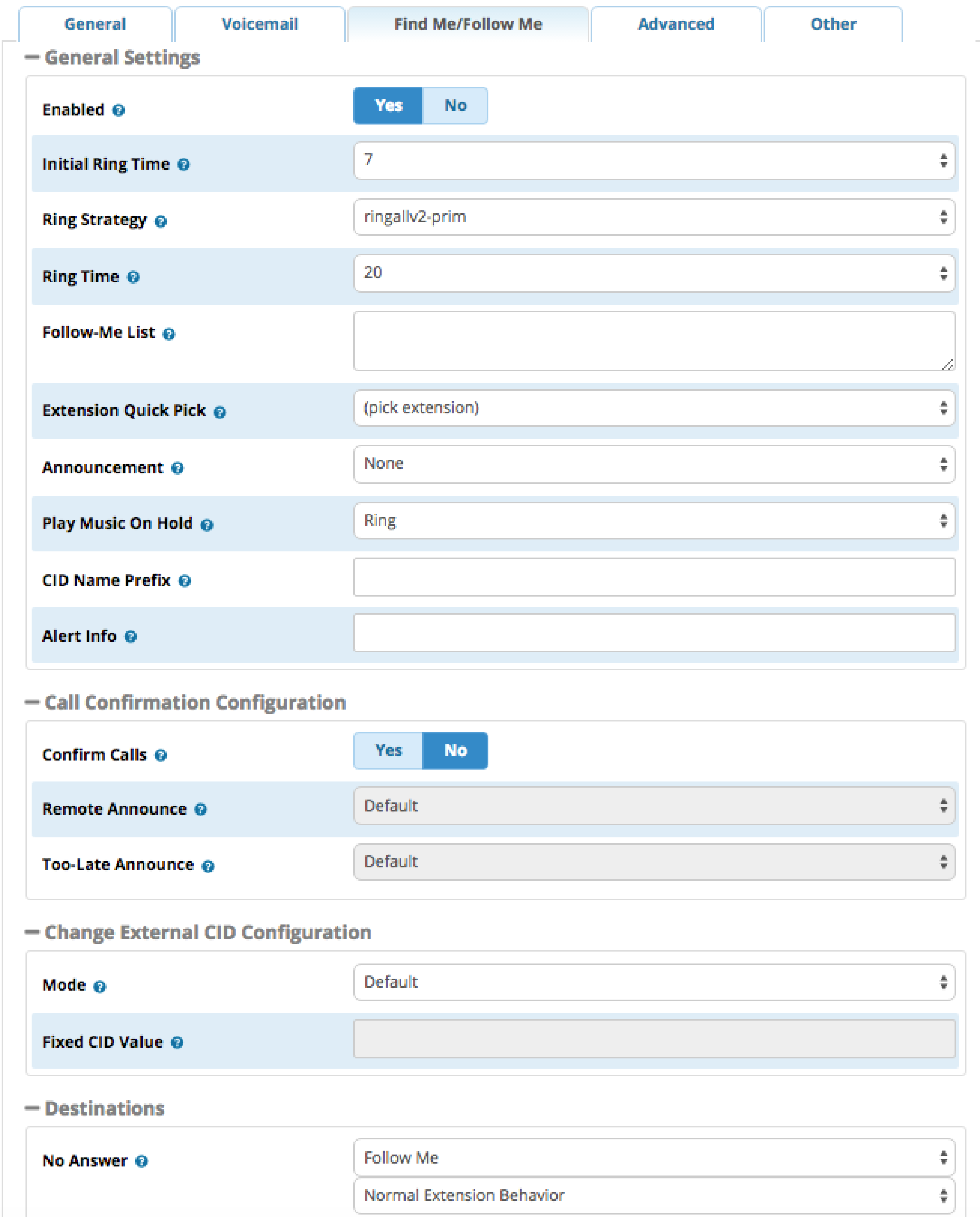
Optional - Enter a destination to send the caller to when they press 1. This can be an internal extension, ring group, queue, or external number such as a cell phone number.

Press 2

Optional - Enter a destination to send the caller to when they press 2. This can be an internal extension, ring group, queue, or external number such as a cell phone number.

Find Me / Follow Me

Click on the **Find Me / Follow Me** tab.
Find Me / Follow Me is enabled by default here so that you may edit the settings. After entering settings, you can disable it if desired.



General

Settings Enabled

Yes/No: Whether to enable Find Me / Follow Me. Must be set to **Yes** (at least temporarily) in order to edit other settings on this page. If you leave it set to **Yes**, Find Me / Follow Me will be active for this extension when you save the extension and apply config. You can set to **No** to disable Find Me / Follow Me for an extension until the user activates it.

Initial Ring Time

Use the drop-down menu to select initial ring time, in seconds. This is the number of seconds to ring the primary extension prior to proceeding to the follow-me list. If "0," the primary extension will not be rung before proceeding to the follow-me list. The extension can also be included in the follow-me list.

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

memoryhunt: ring first extension in the list, then ring the 1st and 2nd extension, then ring 1st 2nd and 3rd extension in the list....
etc.

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

Enter a list of extensions to ring, one per line, or use the Extension Quick Pick menu below. You can include an extension on a remote system, or an external number, by suffixing a number with a pound (#). ex: 2448089# would dial 2448089 on the appropriate trunk.

Extension Quick Pick

This drop-down menu gives you the option to select existing extensions to add to the Follow-Me List above.

Announcement

Select the message to be played to the caller before dialing the find me / follow me list. Default = none.
The drop-down menu shows available system recordings.

Play Music On Hold

If you select a Music on Hold class to play, instead of the default "Ring," the caller will hear that MoH instead of ringing while they are waiting for someone to pick up.

CID Name Prefix

Optional - You can optionally prefix the Caller ID name when ringing extensions in this group. For example, if you prefix with "Sales:", a call from John Doe would display as "Sales:John Doe" on the find me / follow me list extensions that ring.

Alert Info

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

Call Confirmation Configuration

Confirm Calls

Yes/No: Whether to confirm external calls. Call confirmation requires the remote party to press 1 to accept the call. This can help prevent

an unanswered find me / follow me call from reaching an external voicemail box. This feature only works with the ringall or ringall-prim ring strategies.

Remote Announce

Message to be played to the person receiving the call if **Confirm Calls = Yes**. You can use the default message or select one of your System Recordings.

Too-Late Announce

Message to be played to the person receiving the call if **Confirm Calls = Yes** and the call has already been accepted elsewhere.

Change External CID Configuration

Mode

Default: Transmits the caller's CID if allowed by the trunk.

Fixed CID Value: Always transmit the Fixed CID Value below.

Outside Calls Fixed CID Value: Transmit the Fixed CID Value below on calls that come in from outside only. Internal extension-to-extension calls will continue to operate in default mode.

Use Dialed Number: Transmit the number that was dialed as the CID for calls coming from outside. Internal extension-to-extension calls will continue to operate in default mode. There must be a DID on the inbound route for this. This will be BLOCKED on trunks that block foreign CallerID.

Force Dialed Number: Transmit the number that was dialed as the CID for calls coming from outside. Internal extension-to-extension calls will continue to operate in default mode. There must be a DID on the inbound route for this. This WILL be transmitted on trunks that block foreign CallerID.

Fixed CID Value

Fixed value to replace the CID used with some of the modes above. Should be in a format of digits only with an option of E164 format using a leading "+".

Destinations

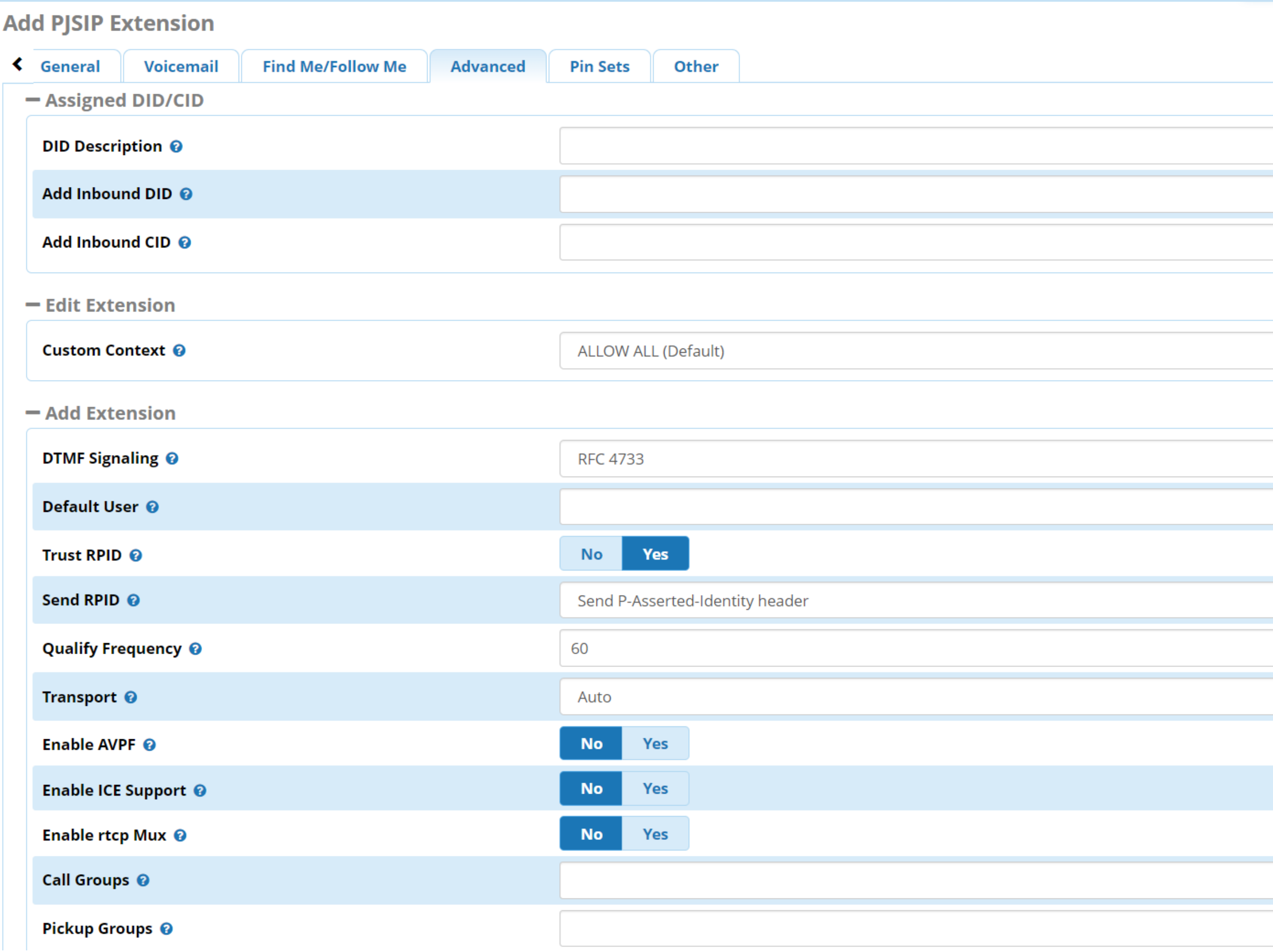
No Answer

Optional destination call is routed to when the call is not answered on an otherwise idle phone. If the phone is in use and the call is simply ignored, then the busy destination will be used.

Remember to set **Enabled = No** at the top of the page after you're done changing settings if you do *not* want find me / follow me to be active. Otherwise, it will be *active* for the extension after you save settings and apply config.

Advanced

Click on the **Advanced** tab.
There are many settings in this tab. See below for explanations of the options.

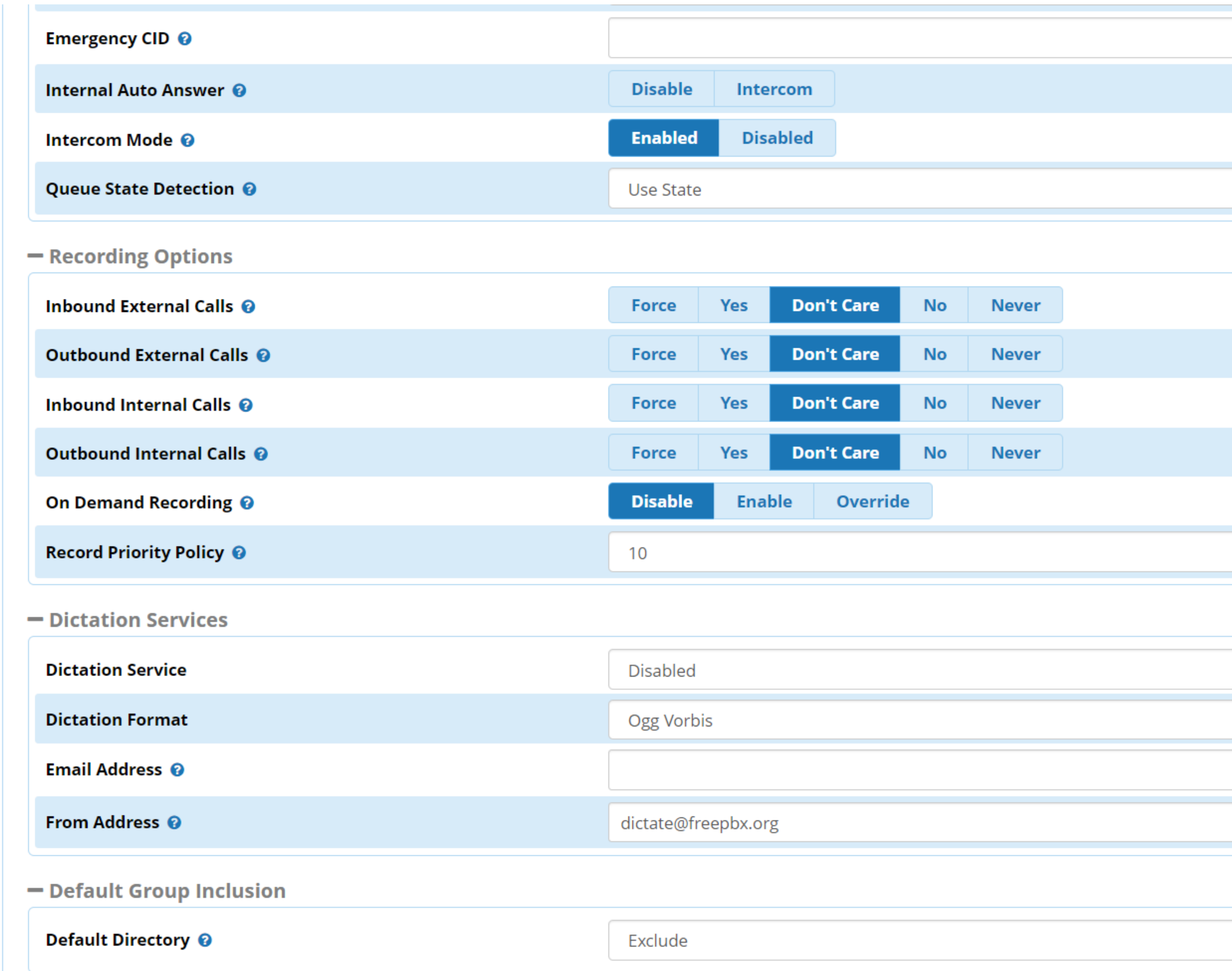


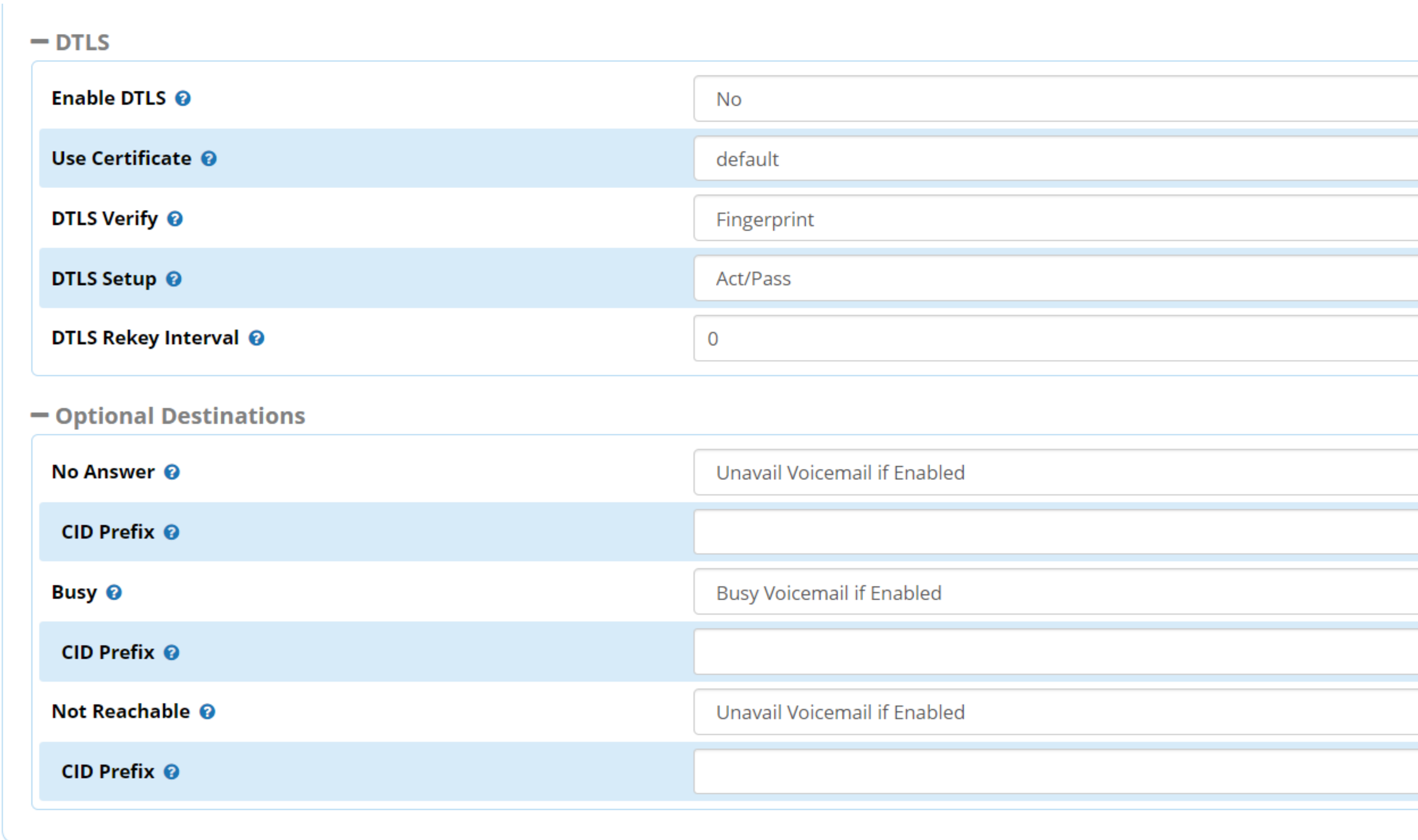
Disallowed Codecs	
Allowed Codecs	
Dial	
Mailbox	
Voicemail Extension	
Account Code	
Max Contacts	1
Media Use Received Transport	Yes **No**
RTP Symmetric	**Yes** No
Rewrite Contact	**Yes** No
Force rport	**Yes** No
MWI Subscription Type	**Auto** Unsolicited Solicited
Aggregate MWI	No **Yes**
Enable RTP bundling	**No** Yes
Media Encryption	None
Session Timers	Yes
Timer Expiration Period	90
Direct Media	No **Yes**
Allow Non-Encrypted Media (Opportunistic SRTP)	**No** Yes
Refer Blind Progress	No **Yes**

Device State Busy at	0
Match (Permit)	
Maximum Expiration	7200
Minimum Expiration	60
RTP Timeout	0
RTP Hold Timeout	0
Outbound Proxy	
Messages Context	
CID Num Alias	
SIP Alias	

— Extension Options

Asterisk Dial Options	HhTtr
Ring Time	Default
Ringer Volume Override	None
Call Forward Ring Time	Default
Outbound Concurrency Limit	3
Call Waiting	**Enable** Disable
Call Waiting Tone	Enable **Disable**
Call Screening	Disable


Emergency CID
Internal Auto Answer
Disable
Intercom
Intercom Mode
Enabled
Disabled
Queue State Detection
Use State
Recording Options
Inbound External Calls
Force
Yes
Don't Care
No
Never
Outbound External Calls
Force
Yes
Don't Care
No
Never
Inbound Internal Calls
Force
Yes
Don't Care
No
Never
Outbound Internal Calls
Force
Yes
Don't Care
No
Never
On Demand Recording
Disable
Enable
Override
Record Priority Policy
10
Dictation Services
Dictation Service
Disabled
Dictation Format
Ogg Vorbis
Email Address
From Address
dictate@freepbx.org
Default Group Inclusion
Default Directory
Exclude



Assigned DID/CID

DID Description

A description for this DID, such as "Fax"

Add Inbound DID

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

Add Inbound CID

Add a CID for more specific DID + CID routing. A DID must be specified in the above **Add Inbound DID** field. In addition to standard dial sequences, you can also enter Private, Blocked, Unknown, Restricted, Anonymous, Withheld, and Unavailable in order to catch these special cases if the Telco transmits them.

Call Camp-On Services

Caller Policy

Asterisk: cc_agent_policy. Used to enable Camp-On for this user and set the technology mode that will be used when engaging the feature. In most cases **Generic Device** should be chosen unless your phones are designed to work with channel-specific capabilities.

Callee Policy

Asterisk: cc_monitor_policy. Used to control whether other phones are allowed to Camp On to this extension. If so, it sets the technology mode used to monitor the availability of the extension. If no specific technology support is available, then it should be set to a **Generic Device**. In this mode, a callback will be initiated to this extension when it changes from an InUse state to NotInUse. If it was busy when first attempted, this will be when the current call has ended. If it simply did not answer, then this will be the next time this phone is used to make or answer a call and then hangs up. It is possible to set this to take advantage of **Native Technology Support** if available and automatically fall back to the **Generic Mode** if not.

Add Extension

DTMF Signaling

The DTMF signaling mode used by this device, usually **RFC** for most phones. [dtmfmode]

Can Reinvite

No/Yes/nonat/update: Re-Invite policy for this device, see Asterisk documentation for details. [canreinvite]

Context

Asterisk context this device will send calls to. Only change this is you know what you are doing. [context]

Host

Host settings for this device, almost always dynamic for endpoints. [host]

Trust RPID

Whether Asterisk should trust the RPID settings from this device. Usually should be yes for CONNECTEDLINE() functionality to work if supported by the endpoint. [trustrpid]

Send RPID

Whether Asterisk should send RPID (or PAI) info to the device. Usually should be enabled to the settings used by your device for CONNECTEDLINE() functionality to work if supported by the endpoint. [sendrpid]

Connection Type

Asterisk connection type, usually friend for endpoints. [type]

NAT Mode

NAT setting, see Asterisk documentation for details. **Yes** usually works for both internal and external devices. Set to **No** if the device will always be internal. [nat]

Yes - (force_rport,comedia): Always ignore info and assume NAT **No - (no):** Use NAT mode only according to RFC3581 (;rport) **Force rport - (force_rport):** Force rport to always be on.

comedia - (comedia): Use rport if the remote side says to use it and perform comedia RTP handling. *Automatic Force Both - (auto_force_rport,auto_comedia): See Below*

Automatic Force rport - (auto_force_rport): Force rport if Asterisk detects that an incoming SIP request crossed a NAT after being sent by the remote endpoint.

Automatic comedia - (auto_comedia): Use comedia if Asterisk detects that an incoming SIP request crossed a NAT after being sent by the remote endpoint.

never - (no): Never attempt NAT mode or RFC3581 support

route - (force_rport): Assume NAT, don't send rport

Port

Endpoint port number to use, usually 5060. Some 2-port devices such as ATA may use 5061 for the second port.

Qualify

Setting to yes (equivalent to 2000 msec) will send an OPTIONS packet to the endpoint periodically (default every minute). Used to monitor the health of the endpoint. If delays are longer then the qualify time, the endpoint will be taken offline and considered unreachable. Can be set to a value which is the msec threshold. Setting to no will turn this off. Can also be helpful to keep NAT pinholes open.

Qualify Frequency

Frequency in seconds to send qualify messages to the endpoint.

Transport

This sets the allowed transport settings for this device and the default (Primary) transport for outgoing. The default transport is only used for outbound messages until a registration takes place. During the peer registration the transport type may change to another supported type if the peer requests so. In most common cases, this does not have to be changed as most devices register in conjunction with the host=dynamic setting. If you are using TCP and/or TLS you need to make sure the general SIP Settings are configured for the system to operate in those modes and for TLS, proper certificates have been generated and configured. If you are using websockets (such as WebRTC) then you must select an option that includes WS.

Enable AVPF

Whether to Enable AVPF. Defaults to no. The WebRTC standard has selected AVPF as the audio video profile to use for media streams. This is not the default profile in use by Asterisk. As a result the following must be enabled to use WebRTC.

Force AVP

Force 'RTP/AVP', 'RTP/AVPF', 'RTP/SAVP', and 'RTP/SAVPF' to be used for media streams when appropriate, even if a DTLS stream is present.

Enable ICE Support

Whether to Enable ICE (Interactive Connectivity Establishment) Support. Defaults to **no**. ICE is a protocol for Network Address Translator (NAT) traversal for UDP-based multimedia sessions established with the offer/answer model. This option is commonly enabled in WebRTC setups.

Enable Encryption

Whether to offer SRTP encrypted media (and only SRTP encrypted media) on outgoing calls to a peer. Calls will fail with HANGUPCAUSE=58 if the peer does not support SRTP. Defaults to **no**.

Call Groups

Callgroup(s) that this device is part of. Can be one or more callgroups, e.g. "1,3-5" would be in groups 1,3,4,5.

Pickup Groups

Pickupgroups(s) that this device can pickup calls from. Can be one or more groups, e.g. '1,3-5' would be in groups 1,3,4,5. Device does not have to be in a group to be able to pick up calls from that group.

Disallowed Codecs

Disallowed codecs. Set this to "all" to remove all codecs defined in the general settings, and then specify specific codecs separated by "&" on the "allow" setting, or just disallow specific codecs separated by "&".

Allowed Codecs

Allow specific codecs, separated by the "&" sign and in priority order. E.g. "ulaw&g729". Codecs allowed in the general settings will also be allowed unless removed with the "disallow" directive above.

Dial

How to dial this device. This should not be changed unless you know what you are doing.

Account Code

Account code for this device.

Mailbox

Mailbox for this device. This should not be changed unless you know what you are doing. This will be automatically set to: <extnum>@device

With an LDAP backend, to enable MWI (Message waiting indication), enter a semi-colon list of mailboxes in the form <extnum>@***default***

e.g. 101@default;102@default ...where the message lamp will light if any of the specified mailboxes has unread mail

Voicemail Extension

Asterisk dialplan extension to reach voicemail for this device. Some devices use this to auto-program the voicemail button on the endpoint. If left blank, the default vmexten setting is automatically configured by the voicemail module. Only change this on devices that have special needs.

Deny

IP Address range to deny access to, in the form of network/netmask.

Permit

IP Address range to allow access to, in the form of network/netmask. This can be a very useful security option when dealing with remote extensions that are at a known location (such as a branch office) or within a known ISP range for some home office situations.

CID Num Alias

The CID Number to use for internal calls, if different from the extension number. This is used to masquerade as a different user. A common example is a team of support people who would like their internal CallerID to display the general support number (a ring group or queue).
There will be no effect on external calls.

SIP Alias

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

Extension Options

Asterisk Dial Options

Cryptic Asterisk Dial Options. Check the **override** box to customize for this extension, or un-check to use system defaults set in Advanced Options. These will not apply to trunk options, which are configured with the trunk.

Ring Time

Number of seconds to ring prior to going to the "no answer" destination. Default will use the global default value set in Advanced Settings. If no voicemail is configured this will be ignored.

Call Forward Ring Time

Number of seconds to ring during a Call Forward, Call Forward Busy or Call Forward Unavailable call prior to continuing to voicemail or specified destination. Setting to **Always** will cause the phone to just continue to ring without going to the "no answer" destination. **Default** will use the current Ring Time. If voicemail is disabled and a destination is not specified, it will be forced into **Always** mode.

Outbound Concurrency Limit

Maximum number of outbound simultaneous calls that an extension can make. This is also very useful as a security protection against a system that has been compromised. It will limit the number of simultaneous calls that can be made on the compromised extension.

Call Waiting

Enable/Disable: Set the initial/current Call Waiting state for this user's extension. If disabled, a second concurrent incoming call would be sent to this extension's "busy" destination.

Internal Auto Answer

Disable/Intercom: When set to **Intercom**, calls to this extension/user from other internal users act as if they were intercom calls meaning they will be auto-answered if the endpoint supports this feature and the system is configured to operate in this mode. All the normal white list and black list settings will be honored if they are set. External calls will still ring as normal, as will certain other circumstances such as blind transfers and when a Follow Me is configured and enabled. If **Disabled**, the phone rings as a normal phone.

Call Screening

Call Screening requires external callers to say their name, which will be played back to the user and allow the user to accept or reject the call. Screening with memory only verifies a caller for their CallerID once. Screening without memory always requires a caller to say their name. Either mode will always announce the caller based on the last introduction saved with that CallerID. If any user on the system uses the memory option, when that user is called, the caller will be required to re-introduce themselves and all users on the system will have that new introduction associated with the caller's CallerID.

Pinless Dialing

Enabling Pinless Dialing will allow this extension to bypass any PIN codes normally required on outbound calls.

Emergency CID

This Caller ID will always be used when dialing out an Outbound Route that is designated as an "Emergency" route (i.e. when dialing 911). The Emergency CID overrides all other CallerID settings.

Queue State Detection

If this extension is part of a queue, then the queue will attempt to use the user's extension state or device state information when determining if this queue member should be called. In some uncommon situations, such as a Follow-Me with no physical device, or some virtual extension scenarios, the state information will indicate that this member is not available even when it is available. Setting this to **Igno re State** will make the queue ignore all state information, thus always trying to contact this member. Certain side effects can occur when this route is taken due to the nature of how queues handle local channels. For example, subsequent transfers will continue to show the member as busy until the original call is terminated. In most cases, this should be set to **Use State**.

Recording Options

Inbound External Calls

Force/Yes/Don't Care/No/Never: Recording of inbound calls from external sources.

Outbound External Calls

Force/Yes/Don't Care/No/Never: Recording of outbound calls to external sources.

Inbound Internal Calls

Force/Yes/Don't Care/No/Never: Recording of calls received from other extensions on the system.

Outbound Internal Calls

Recording of calls made to other extensions on the system.

On Demand Recording

Disable/Enable/Override: Enable or disable the ability to do on demand (one-touch) recording. The overall calling policy rules still apply, and if calls are already being recorded by "Force" or "Never," the cannot be paused unless "Override" is selected.

Record Priority Policy

This is the call recording policy priority relative to other extensions when there is a conflict (i.e. one extension wants to record and the other extension does not). The higher of the two priorities determines the policy. If the two priorities are equal, the global policy (caller or callee) determines the policy.

Dictation

Services Dictation Service

Disabled/Enabled: Whether Dictation service is available for this extension.

Dictation Format

Audio format to use for dictation (**Ogg Vorbis**, **GSM**, or **WAV**).

Email Address

The email address that completed dictations are sent to.

From Address

The email address that completed dictations are sent from. Format is "A Persons Name <email@address.com>", without quotes, or just a plain email address.

DTLS

(Datagram Transport Layer Security)

Enable DTLS

No/Yes: Enable or disable DTLS-SRTP support.

DTLS Verify

Verify that provided peer certificate and fingerprint are valid.

Yes: Perform both certificate and fingerprint verification **No:** Perform no certificate or fingerprint verification **Fingerprint:** Perform ONLY fingerprint verification **Certificate:** Perform ONLY certificate verification

DTLS Setup

Whether we are willing to accept connections, connect to the other party, or both. This value will be used in the outgoing SDP when offering and for incoming SDP offers when the remote party sends actpass

Active: we want to connect to the other party **Passive:** we want to accept connections only **Act/Pass**: we will do both

DTLS Rekey Interval

Interval at which to renegotiate the TLS session and rekey the SRTP session. If this is not set or the value provided is 0 rekeying will be disabled.

Optional

Destinations No Answer

Optional destination call is routed to when the call is not answered on an otherwise idle phone. If the phone is in use and the call is simply ignored, then the busy destination will be used instead.

CID Prefix

Optional CID Prefix to add before sending to this no answer destination

Busy

Optional destination the call is routed to when the phone is busy or the call is rejected by the user. This destination is also used on an unanswered call if the phone is in use and the user chooses not to pick up the second call.

CID Prefix

Optional CID Prefix to add before sending to this busy destination.

Not Reachable

Optional destination the call is routed to when the phone is offline, such as a softphone currently off or a phone unplugged.

CID Prefix

Optional CID Prefix to add before sending to this not reachable destination

Other

Endpoint

Brand

Brand of device to be provisioned

MAC

MAC Address of device to be provisioned

Template

Template to use for device

Model

Model of device to be provisioned

Account

Account number to be assigned

Auto Answer

yes/no: Whether to make this extension automatically answer the initial call received from the system when performing an origination within iSymphony. Only works with Aastra, Grandstream, Linksys, Polycom, and Snom phones.

Language

Language Code

This will cause all messages and voice prompts to use the selected language if installed. Languages can be added or removed in the Sound Languages module.

Default Group

Inclusion **Default Page Group**

Exclude/Include: Whether this extension/device will be part of the default page group.

Default VMblast Group

Exclude/Include: Whether this extension/device will be part of the default voicemail blast group. **Include** will be ignored if the user does not have a voicemail box.

Device Options

Parkinglot

Choose a default Parking Lot for this extension.

Saving the Extension

Click the **Submit** button
Click the **Apply Config** button

3.6 IVR

An IVR or "Interactive Voice Response" menu allows callers to interact with your telephone system via their telephone keypads.

The IVR Module is used to set up a menu system that will play an initial recording to callers, allow them to dial an option or an extension number, and route their call to a particular location based upon what they dial.

For example, you could configure an inbound route to send an incoming call to an IVR, so that when people call your number, they would hear a greeting that would thank them for calling and say, "If you know your party's extension number, you may dial it at any time. For sales, press 1. For service, press 2. For our address and fax number, press 3. For our hours of operation, press 4."

How is the IVR Module related to the other Modules?

The IVR module plays messages that you record or upload in the System Recordings module.

The IVR Module works together with any module that can route a phone call, including Inbound Routes, Ring Groups, Queues, and Paging.

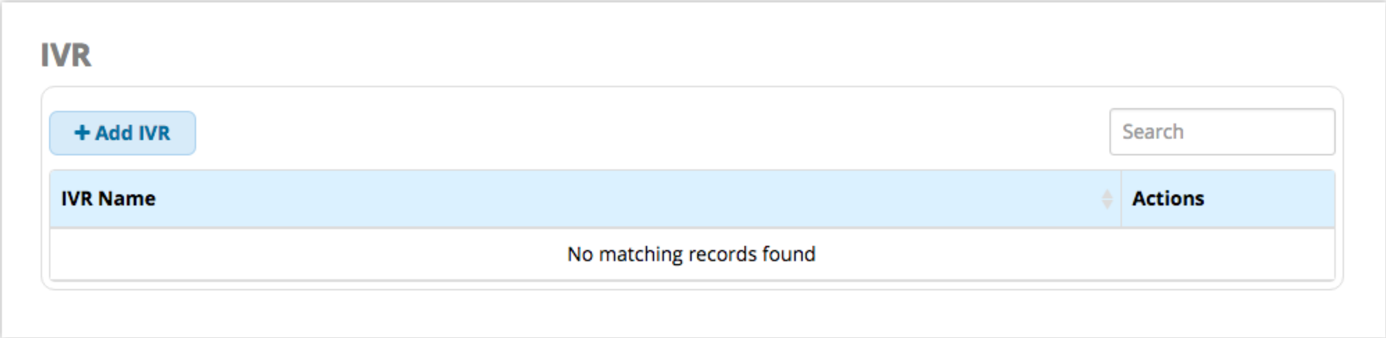
The IVR Module also works together with any module that can act as a destination, because the IVR Module is used to route calls.

Logging In

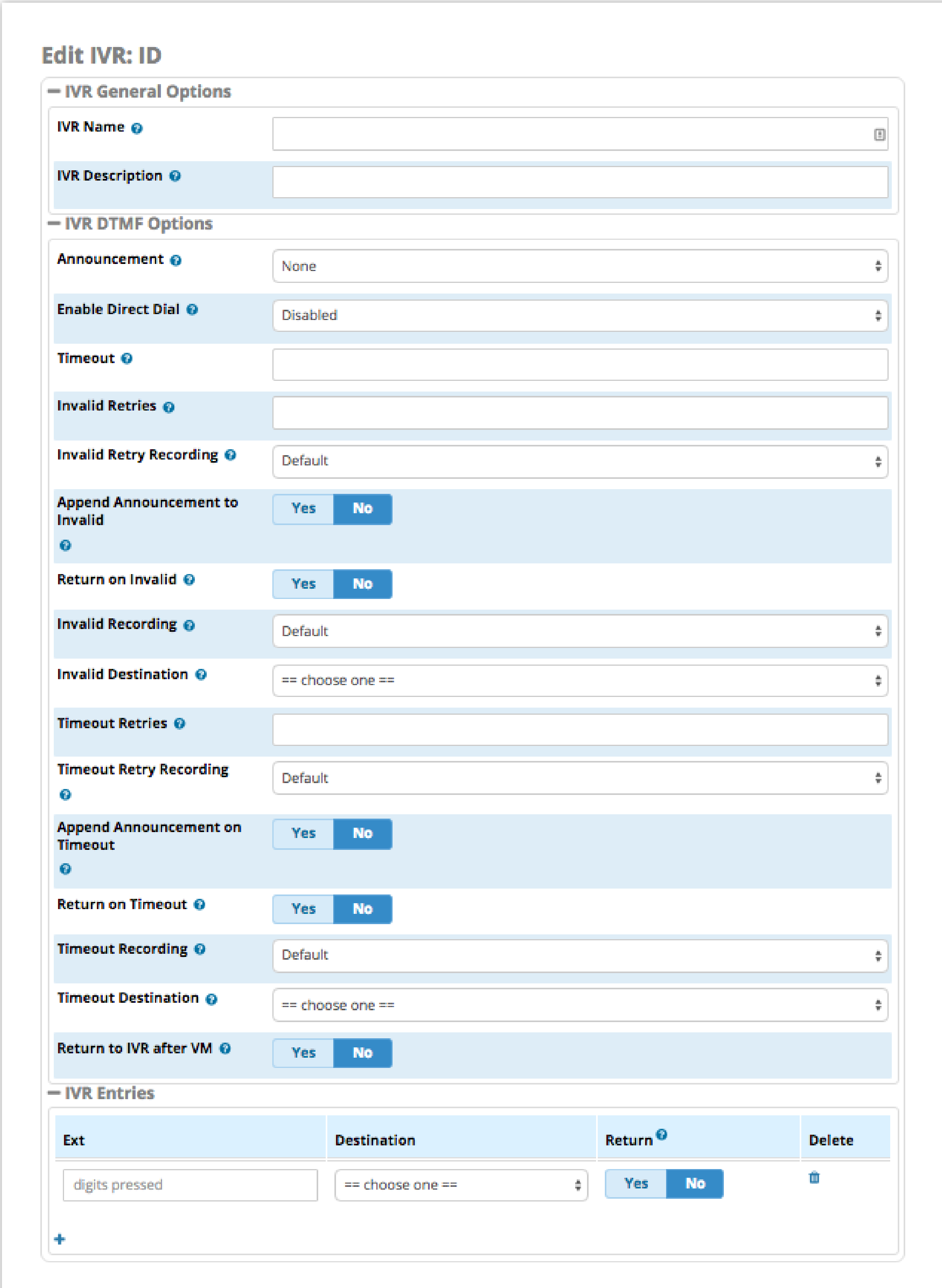
- From the top menu click **Applications**
- From the drop down click **IVR**

Creating a new IVRs

To add an IVR, click the **Add IVR** button.



Fill out the form as described below.



IVR General Options

IVR Name

Enter a name for this IVR.

IVR Description

Optional: Enter a description for the IVR to help you remember what it is for.

IVR DTMF Options

Announcement

Here we choose which recording to be played to the caller when they enter the IVR. This can be any system recording that you have defined in the System Recording module. It will usually give them instructions, such as "press 1 for sales and 9 for support."

Enable Direct Dial

Do you want to allow callers to be able to enter a user's extension number when navigating the IVR to go directly to that user's extension? Your options are:

- **Disabled** - This will not allow any caller to direct dial any extensions on the system. Callers will be restricted to dialing only the IVR entries that you define.
- **Extensions** - This will allow a caller to dial any system extension directly from the IVR, regardless of what entries you define in the IVR.
- **Directory Names** (if a directory exists) - You will get a list of all company directories on your PBX, and you can restrict direct dialing to users who are a part of the company directory. This is a way to restrict which extensions a caller can direct dial from an IVR. To set up a directory, visit the Directory module.

Timeout

Enter the amount of time (in seconds) the system should wait for the caller to enter an option on their phone keypad. If this amount of time passes without the caller entering anything, it will be considered a timeout. After a timeout, the system follows the timeout rules defined below. We recommend setting this to 4 or 5 seconds.

Invalid Retries

How many times a caller is allowed to enter an option without finding a match before we send the caller to the Invalid Destination as defined below. We recommend setting this to 2.

Invalid Retry Recording

The prompt to play to the caller when they enter an invalid entry. This can be any system recording from the System Recordings module.

Append Announcement to Invalid

Yes/No: Controls whether a caller who makes an invalid entry will hear the main IVR announcement again. If set to **yes,** the system will replay the main IVR announcement after playing the invalid retry recording.

Return on Invalid

Yes/No: Controls whether a caller who makes an invalid entry in a "sub-menu" IVR will be returned to the parent IVR. Only applicable if the current IVR was a destination in another ("parent") IVR. If set to **yes**, the caller will return to the parent IVR after an invalid entry. The return path will be to the IVR that was in the call path prior to this IVR, which could lead to strange results if there was another IVR in the call path not immediately before this one.

If set to **no,** the caller will be taken to the "invalid destination" set below after an invalid entry.

Invalid Recording

The recording to play to the caller after they have reached the invalid retry count defined above. This can be any system recording from the System Recordings module.

Invalid Destination

If callers cannot find a match after reaching the number of invalid retries defined above, they will be transferred to the invalid destination you set here. This can be any destination on your PBX.

Timeout Retries

How many times callers are allowed to timeout without pressing any options on their keypad before they are sent to the invalid destination defined above. We recommend setting this to 1.

Timeout Retry Recording

The recording to play to a caller who times out. This can be any system recording from the System Recordings module.

Append Announcement on Timeout

Yes/No: Controls whether a caller who times out will hear the main IVR announcement again. If set to **yes,** the system will replay the main IVR announcement after playing the timeout retry recording.

Return on Timeout

Yes/No: Controls whether a caller who times out in a "sub-menu" IVR will be returned to the parent IVR. Only applicable if the current IVR was a destination in another ("parent") IVR. If set to **yes**, the caller will return to the parent IVR after a timeout. The return path will be to the IVR that was in the call path prior to this IVR, which could lead to strange results if there was another IVR in the call path not immediately before this one.

If set to **no,** the caller will be taken to the "timeout destination" set below after timing out.

Timeout Recording

The recording to play to a caller when they have used the number of timeout retries defined above. This can be any system recording that you defined in the System Recording module.

Timeout Destination

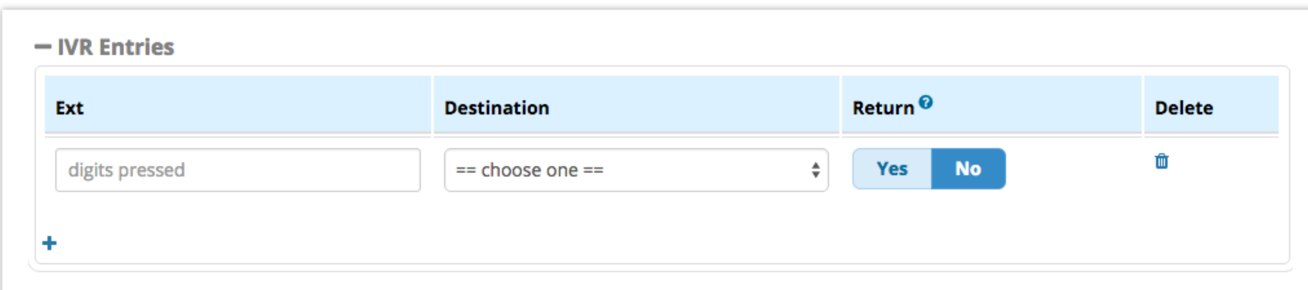
If callers do not make an entry within the maximum number of timeout retries defined above, they will be transferred to the timeout destination. This can be any destination on your PBX.

Return to IVR after VM

Yes/No: Whether to offer callers who end up in a user's voicemail box the option to return to the IVR. If set to **yes**, callers who reach a voicemail box from an IVR will be prompted to leave a voicemail and to press 9 to return to the main menu, which will return them back to this IVR.

IVR Entries

This is where you define options for callers. Press the blue plus sign to add additional entries.



Ext

The digits the caller should press to reach the destination. We recommend using only single-digit entries to keep it simple for your users.

Destination

The destination to route the caller to when they press the digits in the Ext field. This can be any destination on your PBX, such as ring groups, time conditions, queues or anything else.

Return

Yes/No: Whether to send callers back to the parent IVR when they press the digits in the Ext field. For example, this is handy for things such as, "To return to the previous menu, press 9."

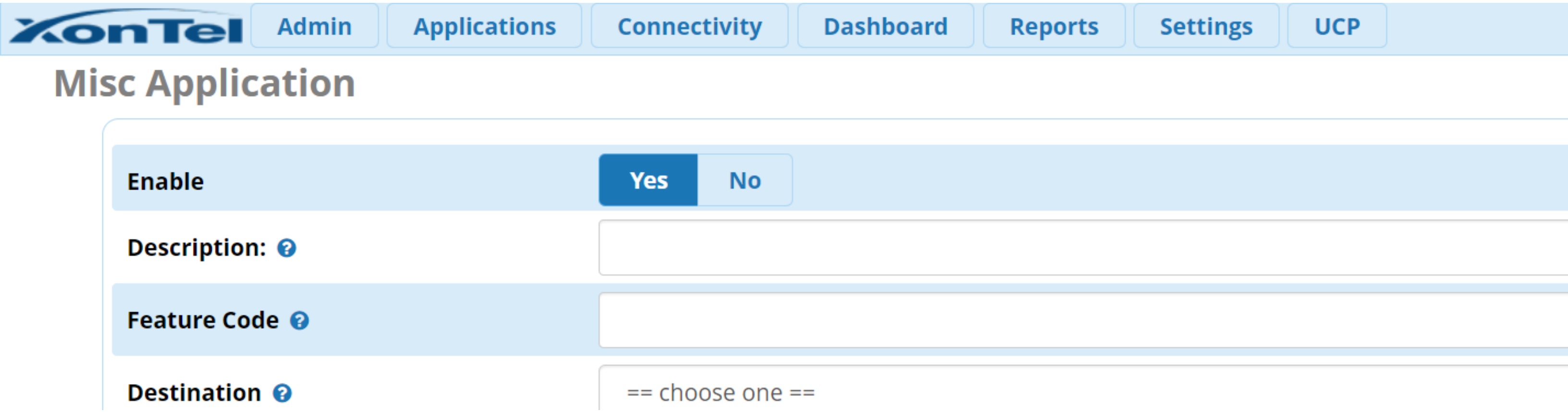
3.7 Misc Applications

A miscellaneous or "misc" application is a custom feature code that you can dial from internal phones to go to various destinations available in the PBX.

Logging in

- From the top menu click **Applications**
- From the drop down click **Misc Applications**

Adding a Misc Application



Enable

Yes/No: Whether to enable this miscellaneous application.

Description

Used to identify this application if it needs to be edited or deleted in the future.

Feature Code

The custom feature code that users will dial to access this application. This can be a star code (example, *7876) or simply a normal extension (example, 7876). This value must be unique and not shared with any user, application, or star code on the PBX. This can also be modified on the feature codes page.

Destination

Where to send callers when they dial the custom feature code.

Save

- Click the **Submit** button
- Click the **Apply Config** button
- Your new misc application will be added to the list at the right side of the screen.

3.8 Misc Destinations

A miscellaneous destination is a custom call target that can be used by another module. Anything that can be dialed from a user's extension can be turned into a misc destination.

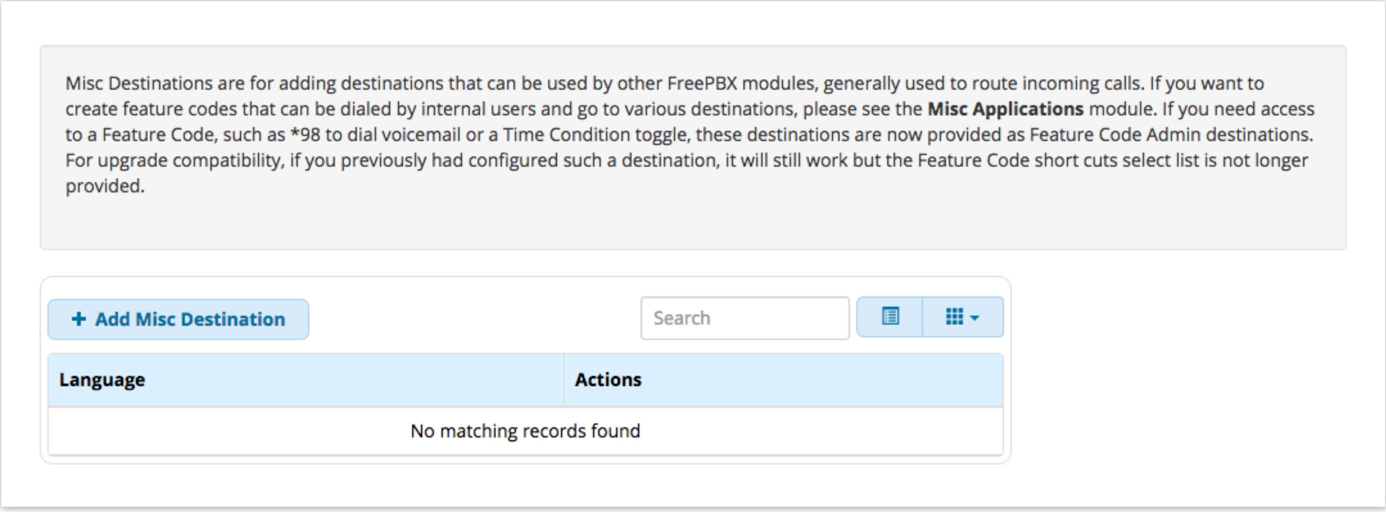
Logging in

On the top menu click **Applications**

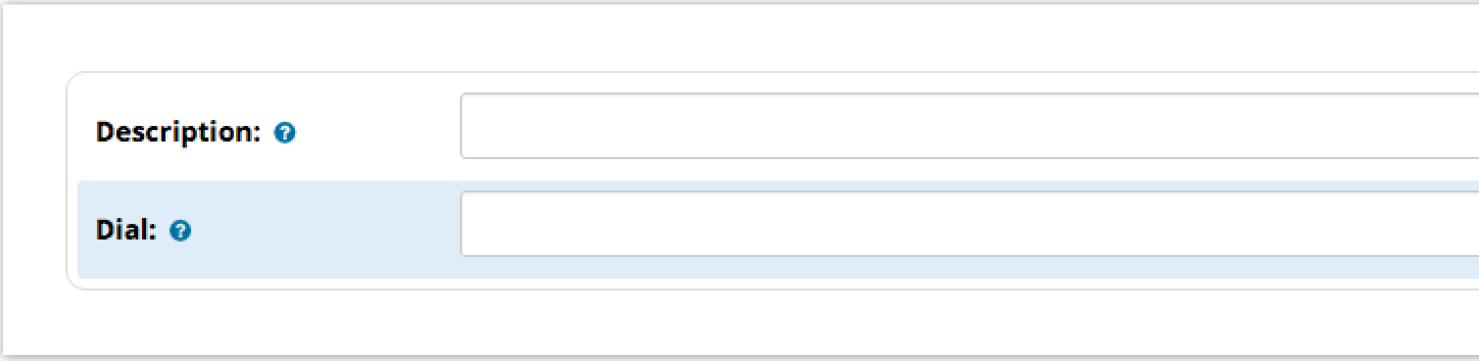
In the drop down click **Misc Destinations**

Adding a Misc Destination

Click the **Add Misc Destination** button.



Fill out the **Description** and **Dial** fields as described below.



Description

Enter a description of the destination to help you identify it. This description will be displayed in other modules that have selectable destinations.

Dial

Enter the extension, telephone number, feature code, or application that the system should dial when a caller is routed to the destination. Anything that can be dialed from a user's extension can be entered into this field.

Save

Click the **Submit** button.

Click the **Apply Config** button.

3.9 Paging and Intercom

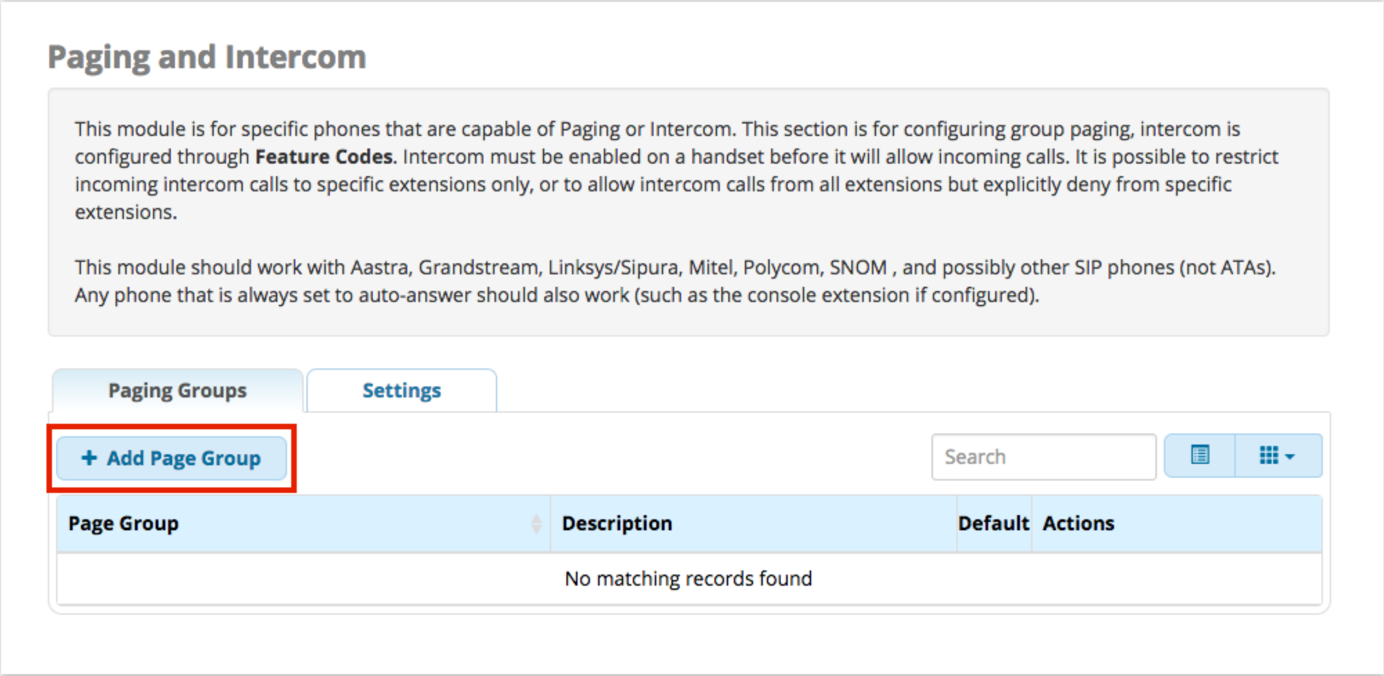
This module is for specific phones that are capable of paging or intercom. In the Paging and Intercom module, you can configure groups of phones that will auto-answer and play the page over their speakers when called from the page group. This module will work with most SIP phones that are supported by the PBX.

Logging In

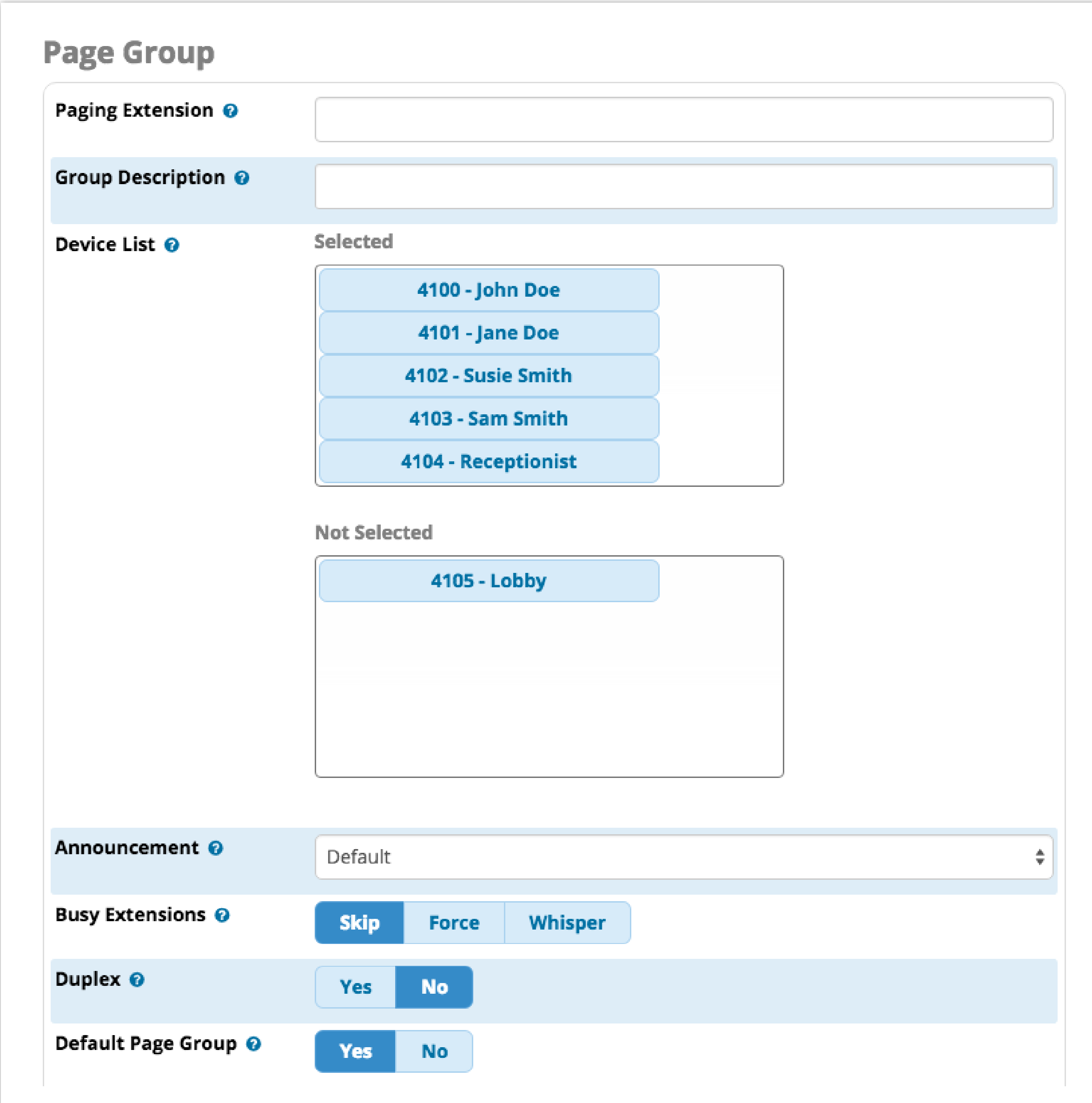
- In the top menu click **Applications**
- In the drop down click **Paging and Intercom**

Paging Groups

Click the **New Page Group** button to add a new page group, or click the pencil icon next to an existing page group to edit.



Standard Module Options



Paging Extension

The extension number for this page group. Users can dial this number to page this group. The number can be any number from 3 to 11 digits as long as it doesn't match an existing extension or feature code.

Group Description

The name of the page group and/or a short description to help you identify it.

Device List

Selected vs. **Not Selected**: Choose which extension(s) to include in the page group by dragging the desired extensions to the **Selected** bin. These will be included in the page group.

Note

We do not recommend having more than 25 phones in a single page group. You can set the maximum number of phones that can be included in any page group by visiting the **Advanced Settings** module and changing the limit under the **Max Paging Participants** section.

Announcement

The announcement to be played to the remote party. Select a **system recording**, **None**, or **Default**. If set to **Default**, it will use the **Auto-Answer Default** global setting found in the **Settings** tab of the Paging and Intercom module. (If that setting is not defined, it will default to a beep).

Busy Extensions

Skip/Force/Whisper: How to handle paging if an extension is busy (such as on a call).

- **Skip**: A busy extension will not receive the page. All other extensions will be paged as usual.
- **Force**: A busy extension will receive the page. The system will not check if the device is in use before paging it. Conversations can be interrupted by the page, depending upon how the device handles the page. In most cases, the phone will ring instead of auto-answering if it is on another call, but some phones will put the caller on hold and play the page. This is not usually a desirable outcome unless you are setting up a page group for emergencies, and you want all extensions to hear the page regardless of whether they are already on a call.
- **Whisper**: The system will attempt to use the ChanSpy capability on SIP channels, resulting in the page being sent to the busy device's earpiece. The page is "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 below, if using Whisper mode.

For Mutlicast, **Skip** | **Force** | **Whisper** doesn't work.
You need to set **Paging Barge** on your phone manually. By Default, Skip is enable (*Paging Barge disabled*), You can set Froce (*Paging Barge*) and play with the prioritry.
If the phone is present into several group (multicast) with different settings , the phone is not able to make the difference.

Duplex

Yes/No: This option controls whether the extension receiving the page is muted by default. If you enable duplex, the extensions that are called in the page group will not be muted, which will allow anyone to talk in the page group. Usually this will be set to **No**.

Paging is typically used for one-way announcements. If Duplex is set to **Yes**, all participants in the page group can talk to each other and hear each other, similar to a conference room.

Any user can dial ***1** to un-mute themselves at any time, regardless of whether Duplex is enabled here.

Default Page Group

Yes/No: Whether to consider this page group a "default" page group. You can create one or more default page groups. This can help you save time when creating extensions in the Extensions module. There, you are given the option of whether to include an extension in the default page group(s), preventing the need to re-visit the Paging & Intercom module to add an extension to the group.

3.10 Parking

The Parking module creates and configures parking lots, sometimes referred to as parking orbits, where calls can be transferred in order to allow another extension to retrieve the calls. This ability is a form of putting a call on hold so that the intended party can retrieve the call from elsewhere.

When a call is parked by transferring that call to the configured parking extension, the call is placed into one of the parking "slots" configured by this module. The parking slot number is announced to the person who parked the call (the "parker"). The slot number can then be dialed from other phones to retrieve the parked call. If the parked call times out and is not retrieved in a timely manner, it can ring back to the parker or be sent to another destination. Parking can be greatly enhanced by programming a phone’s BLF buttons to the configured parking slots or by using parking in in conjunction with visual tools like XactView operator panels or Phone Apps.

Logging In

- In the top menu, click **Applications**
- In the drop down menu, click **Parking**

Editing or Creating Parking Lots

Configuring a parking lot is substantially the same whether using the standard Parking module or using the Parking module with Park Pro installed. Differences are noted in this wiki.

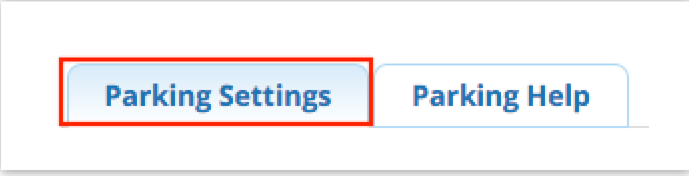
The most important items to configure with parking are:

- Parking Lot Extension
- Parking Lot Starting Position
- Number of Slots
- Parking Timeout
- Destination and Come Back to Origin configuration

Editing the default lot in the standard Parking module

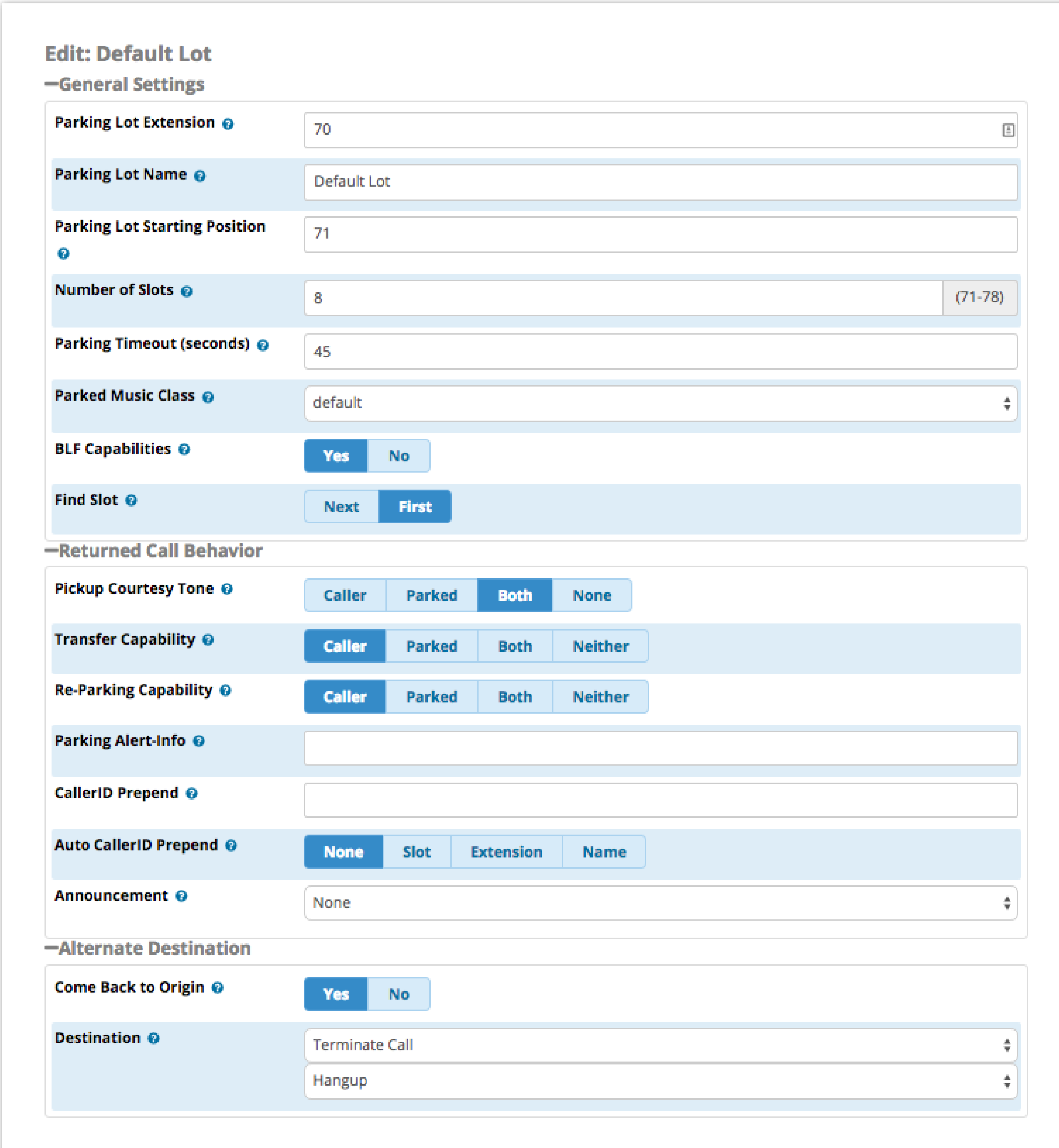
The standard module comes with one "Default" parking lot and does not allow the creation of multiple lots.

You can edit this default lot by going to the **Parking Settings** tab.



Configuration Options

After you've selected a lot to edit, or have created a new lot, you can edit several configuration options. The following screenshot is from the standard Parking module, but we'll explain the Pro options as well.



General Settings

Parking Lot Extension

This is the extension where a call is transferred to in order to send it to the parking lot.

Parking Lot Name

This is a user-friendly name that will show up in the right navigation bar. With Parking Pro, it allows you to identify different parking lots and is used in other parts of the system that may refer to parking lot information, such as the Print Extensions module.

Parking Lot Starting Position

The first slot number for the parking lot. Cannot be the same as the parking lot extension. When used in conjunction with the Number of Slots set below, the system will create a range of extensions for your parking lot, starting with the first slot number.

Number of Slots

The total number of parking slots in this lot. For example, if your extension is 70 and you enter 8 here you would have parking slots 71-78. The slot range will be displayed next to this field.

Parking Timeout (seconds)

The duration of time in seconds that a parked call will remain in the parking lot before timing out. If the call is not picked up within this period, it will automatically be sent to the timeout destination configured in the Alternate Destination section.

Parked Music Class

This is the music class to play to callers who are waiting in the parking lot. If a specific music class has been previously set for the caller prior to being parked, such as if the call came through a Queue that set the music, then this selection will be ignored in favor of the music class that was previously set for the call.

Find Slot

- **Next:** The parking lot will seek the next sequential parking slot relative to the the last parked call instead of seeking the first available slot. This is useful if you have a specific application where you would prefer that calls are parked into the next available slot, such as you want to try and visualize the order in which the calls were parked.
- **First:** Use the first parking slot available. This is the default setting. This might be particularly useful if you have 8 slots available but most phones only have BLF buttons programmed to the first couple of slots. This would maximize the frequency that all calls are parked in the first few slots.

Returned Call Behavior

If a call is not retrieved from the parking lot after the configured timeout duration, then the system will attempt to return the call either directly to the device that parked the call, or to the destination set in the Alternate Destination section. The options configure both capabilities of the returned call, such as whether or not it can be parked again, as well as conditioning of the returned call such as Caller ID pre-pending that may help identify the call as a timed out parked call.

Pickup Courtesy Tone

Caller/Parked/Both/None: Whom to play the courtesy tone to when a parked call is retrieved.

Transfer Capability

Caller/Parked/Both/Neither: Sets who has DTMF-based transfer capability, usually configured as "##," once the call has been picked up. This does not control the transfer capability of a phone's transfer button unless that phone is programmed to send the DTMF code when transferring.

Re-Parking Capability

Caller/Parked/Both/Neither: Sets who can re-park a call after it has timed out.

Parking Alert-Info

Alert-Info to add to the call prior to sending the call back to the originator or alternate destination. Please see our wiki on Alert-Infos for more information on how they work and the options for different phones.

CallerID Prepend

A string to pre-pend to the current Caller ID associated with the parked call prior to sending the call back to the originator or alternate destination. This is often used to identify where a call came from such as PRK to show us it was a Parked Call. If used in conjunction with the Auto CallerID Prepend below, this will be placed first followed by the configured Auto Caller ID.

Auto CallerID Prepend

This will automatically prepend specific identifying information about the parked call after a timeout. The options are:

- **None:** Do not auto populate a CallerID Prepend.
- **Slot:** The parking slot where the parked call was parked prior to the timeout.
- **Extension:** The user extension number who originally parked the call, if parked by a local extension on the PBX
- **Name:** The name associated with the user extension number who originally parked the call, if parked by a local extension on the PBX.

Announcement

A message that will be played to the caller prior to sending the call back to the originator or to the alternate destination. You can select "none" or one of your system recordings.

Alternate Destination

Come Back to Origin

Yes/No: Whether to send a timed-out parked call back to the device that parked the call. If **No**, the timed-out call will be routed straight to the destination set below. If **Yes,** the call will be sent back to the origin, but if that device is not available or does not answer, the destination below will ultimately be used. Therefore, a reasonable destination such as a receptionist, ring group, voicemail, or similar should be set.

Destination

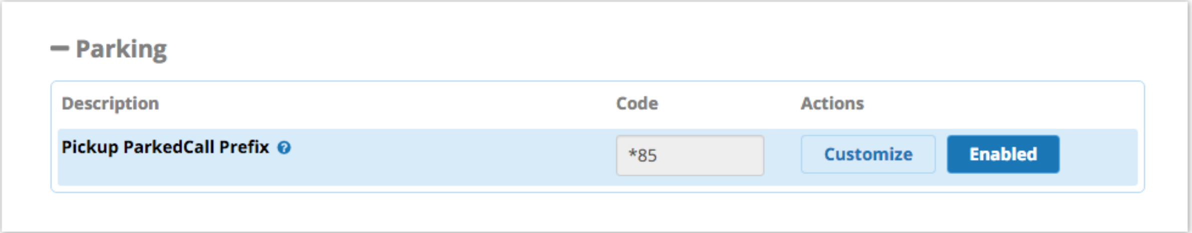
This is the destination where a timed-out parked call will be sent either directly (if Come Back to Origin = No), or when a device is unreachable or not responding. This can be any destination on your PBX.

Save

When finished, click the **Submit** button, then click the **Apply Config** button.

Pick Up Parked Call Feature Code

Parking includes a feature code called **Pickup ParkedCall Prefix.** It is ***85** by default and can be changed in the Feature Codes module. When used in conjunction with a parking lot number, it picks up "the next call" from the specified lot. When used in conjunction with a specific slot number, it picks up the call in that slot.



3.11 Queues

Automatic Call Distribution (ACD) or call queuing provides a way for a PBX to queue incoming calls. A queue is a "stack" or "line" of calls that need to be answered. When a call is directed into the queue, by default, the calls are answered in a first-in, first-out order. Call queues are useful when you have more callers than people available to answer calls. Callers placed into a queue will hear music or advertising until someone is available to answer their call. The Queues module allows you to create and design queues that allow callers to speak with agents as quickly and painlessly as possible.

Queues Consist of:

- **Callers** - Incoming calls placed in the queue
- **Agents** - Members who answer the queue calls (extensions or users that log in as agents)
- **Static** - The agent is always a part of the queue and cannot log out.
- **Dynamic** - The agent can log into or log out of a queue.
- **Ring Strategy** - A strategy for how to handle the queue and divide calls between queue members
- **MoH** - Music or advertisements played for callers while waiting in the queue
- **Announcements** - Played for callers and members

Agent Login

Queue Agent Login Toggle (All Queues)

- ***45** is the default queue login toggle feature code. Agents can log into **all** queues in which they are a dynamic member by dialing ***45**. They can log out by dialing ***45** again. The system will give voice prompts to the caller to indicate status of their queue login.

Queue Agent Login Toggle (Single Queue)

- Dynamic agents can log into or out of a specific queue by dialing ***45xxxx** where **xxxx** is the queue number. The system will give voice prompts to the caller to indicate status of their queue login.

Queue Agent Login Toggle (All Queues with Hint)

- Agents can log into **all** queues in which they are a dynamic member by dialing ***45*yyyy** where yyyy is the user's extension number. They can log out by dialing ***45*yyyy** again. The system will give voice prompts to the caller to indicate status of their queue login and will track a hint indicating login status making it suitable for a phone BLF button.

Queue Agent Login Toggle (Single Queue with Hint)

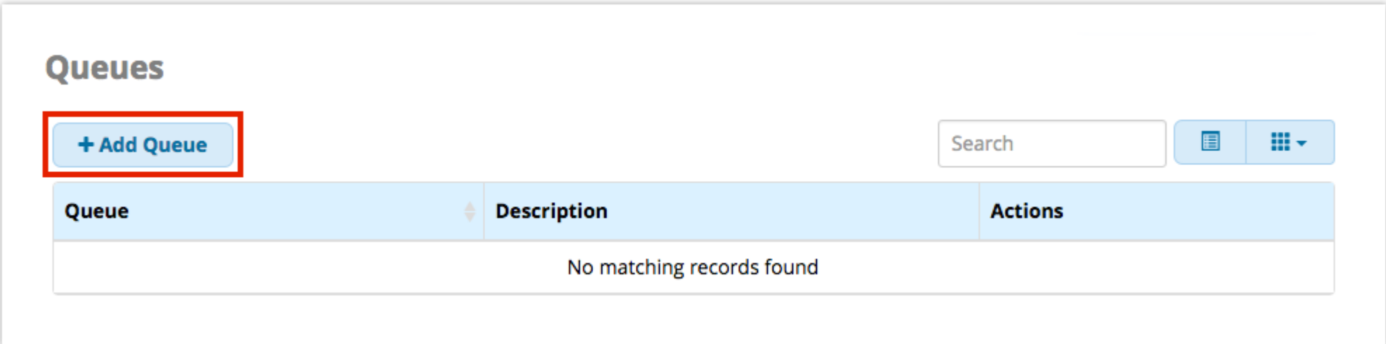
- Dynamic agents can log into or out of a specific queue by dialing ***45yyyy*xxxx** where **xxxx** is the queue number and **yyyy** is the user's extension. The system will give voice prompts to the caller to indicate status of their queue login. This dial string has an associated hint that will track the users login status of the queue, making it suitable for a phone BLF button.

Logging in

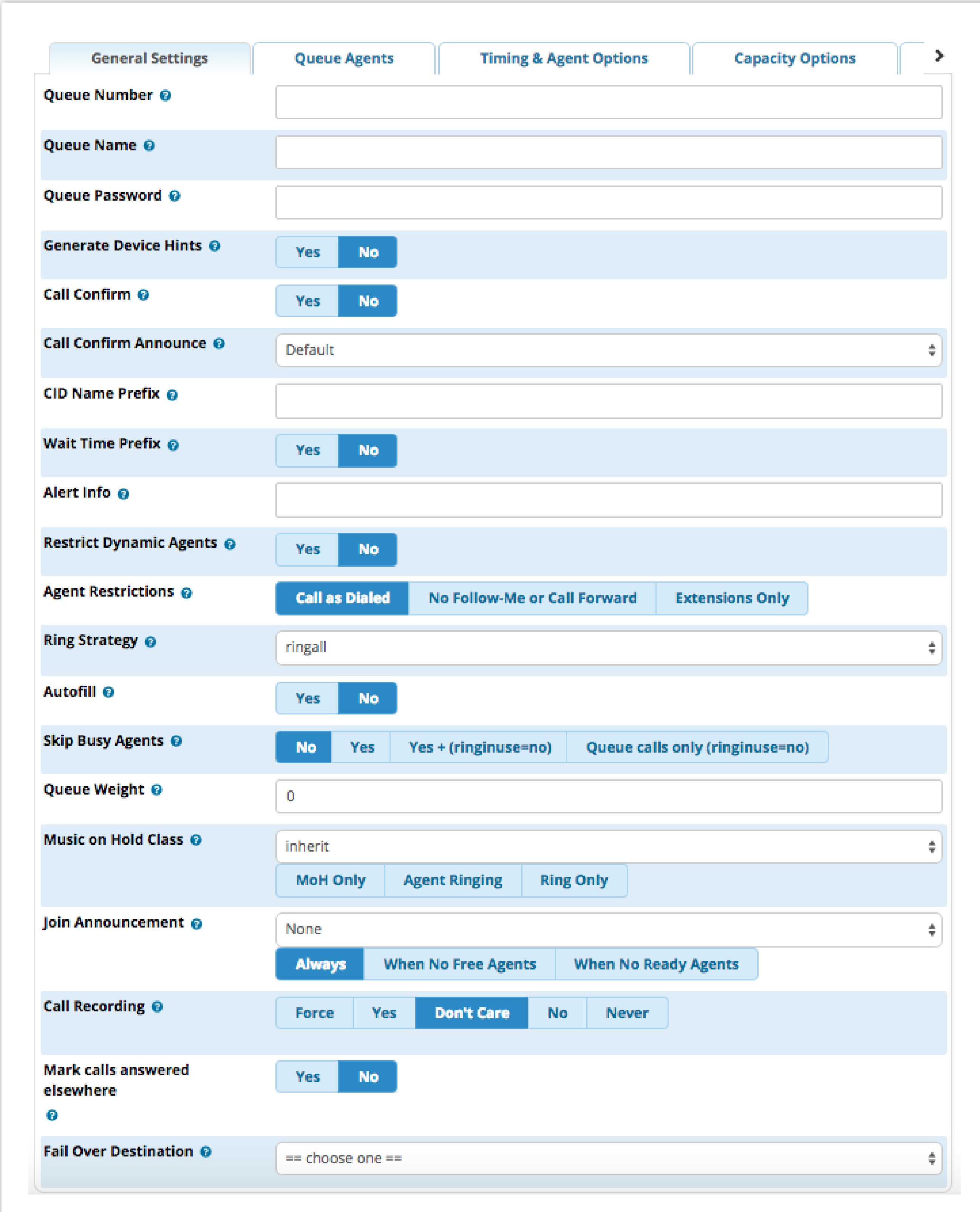
- In the top menu click **Applications**
- In the drop down click **Queues**

Adding a Queue

Click the **Add Queue** button.



General Settings



Queue Number

Use this number to dial into the queue or transfer callers to this number to put them into the queue. This number can also be used in conjunction with feature codes related to agent login/logoff and BLF monitoring.

Queue Name

Give the queue a brief name to help you identify it.

Queue Password

Optional: You can require agents to enter a password before they can login to this queue. The password is only used when logging in with the legacy queue * code. When using the toggle codes, you must use the “Restrict Dynamic Agents” option in conjunction with the dynamic members list to control access.

Generate Device Hints

This option has been removed. It is ***ALWAYS*** enabled

Individual hints and dialplan are generated for each SIP/PJSIP/IAX2 device that could be part of this queue. These hints are used in conjunction with programmable BLF phone buttons to log into and out of a queue and generate BLF status as to the current state. The format of the hints is ***45ddd*qqq** where *45 is the currently defined toggle feature code, ddd is the device number (typically the same as the extension number) and qqq is this queue's number.

Call Confirm

If set to **yes**, queue calls to external phone numbers are forced into call confirmation mode. This includes queue calls to agents who are using external numbers, as well as to internal extensions whose call forwarding or follow-me settings are causing calls to route to external numbers. If call confirmation is enabled, the member must accept 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 that has an alternate message set. This message will override any other message specified. You can add additional recordings in the System Recordings module.

CID Name Prefix

Optional: You can prefix the caller ID 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 the caller has been waiting. It will be rounded to the nearest minute, in the form of “Mnn:” where “nn” is the number of minutes. If the call is subsequently transferred, the wait time will reflect the time since it first entered the queue or reset if the call is transferred to another queue with this feature set.

Alert Info

Optional: This can be used for distinctive ring with SIP devices.

Restrict Dynamic Agents

If set to **Yes,** only the dynamic members listed in the Queue Agents tab will be able to log in. No one else would be allowed to log in as a dynamic member.

Agent Restrictions

- **Called 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.
- **No Follow-Me or Call Forward**: Follow-Me and call forward settings on internal extensions will be ignored. Any other agent will be called as dialed. This behavior is similar to how extensions are dialed in ring groups.
- **Extensions Only:** Same as the no follow-me or call forward mode above, EXCEPT 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 in as a dynamic agent. The call will simply be blocked when the queue tries to call it. For dynamic agents, set the “Agents Regex Filter” in the Advanced Options tab to provide some validation.

Ring Strategy

Some ring strategies (notably 'Linear') may require that Asterisk be restarted before it takes effect.

- **ringall**: Ring all available agents until someone answers (default). If using penalties, all agents will start with a penalty of 0 for the defined ring time as defined in the “Agent Timeout” setting below. Then all agents with a 1,2,3 and so forth.
- **leastrecent**: Ring agent who was least recently called by this queue.
- **fewestcalls**: Ring the agent with the 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.

Autofill

If this is set to **Yes**, and multiple agents are available, the PBX 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 the other callers wait.

Skip Busy Agents

- **No:** The queue will call agents even if they are on an occupied phone.
- **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. In various hunt-style ring strategies, the next agent will be attempted.
- **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.
- **Queue calls only (ringinuse=no): T**he queue configuration flag "ringinuse=no" is set for this queue, 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 with 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 a setting that sets the "ringinuse=no" flag, there is a NEGATIVE side effect. An agent who transfers a queue call will remain unavailable to 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" (priority level). The higher the weight, the higher the priority. (Default = 0) If there are agents common to multiple queues, the queue with the highest priority will deliver its calls first.

Music on Hold Class

The Music on Hold (MoH) played to the caller while they wait in line for an available agent. Select an option from the drop-down menu. Choose "inherit" if you want the MoH class to be what is currently selected, such as by the inbound route.

Below the drop-down menu, select one of these three options:

- **MoH Only:** Play music until the agent answers.
- **Agent Ringing:** Play MoH until an agent's phone is presented with the call, then play ringing. If the agent doesn'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.

Join Announcement

The announcement played to callers before they join 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. You can add additional recordings in the System Recordings module.

- **Always:** Always play the announcement to callers.
- **When No Free Agents:** Play the announcement to callers when no agents are free to answer the call right now. A "free" agent is off the phone but might be completing their "wrap-up" time, so they might not be available to answer the call immediately.
- **When No Ready Agents:** Play the announcement to callers when there are no ready agents. A "ready" agent is someone who is both "free" (off the phone) *and* has completed their wrap-up time (if any is set) after a previous call, so they are theoretically available to answer the call immediately.

Call Recording

Force/Yes/Don't Care/No/Never: Set whether to record incoming calls to this queue.

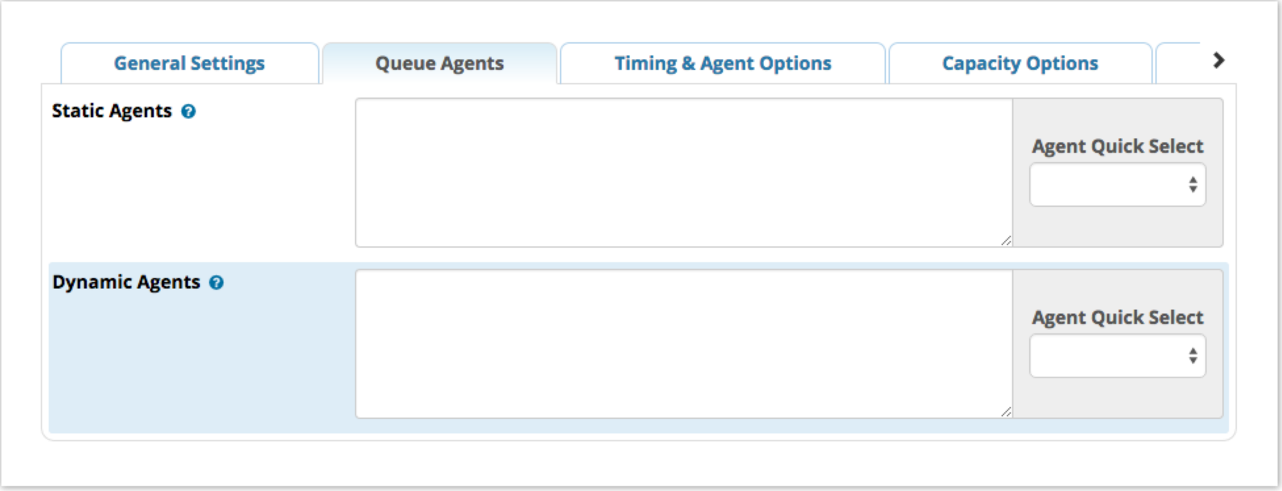
Mark calls answered elsewhere

If set to **Yes**, all calls are marked as "answered elsewhere" when cancelled. The effect is that missed queue calls are *not* shown on the phone as missed calls (if the phone supports it).

Fail Over Destination

Important: Set a failover destination here by choosing a valid destination from the drop-down menus. The caller would be sent to this destination if they exit the queue for reasons such as maximum wait time, queue capacity, or join empty/leave empty settings discussed later in this wiki.

Queue Agents



Static Agents

Static agents are extensions that are assumed to always be in 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 (outbound routing must contain a valid route for external numbers). You can use the **Agent Quick Select** menu to quickly find and add extensions.

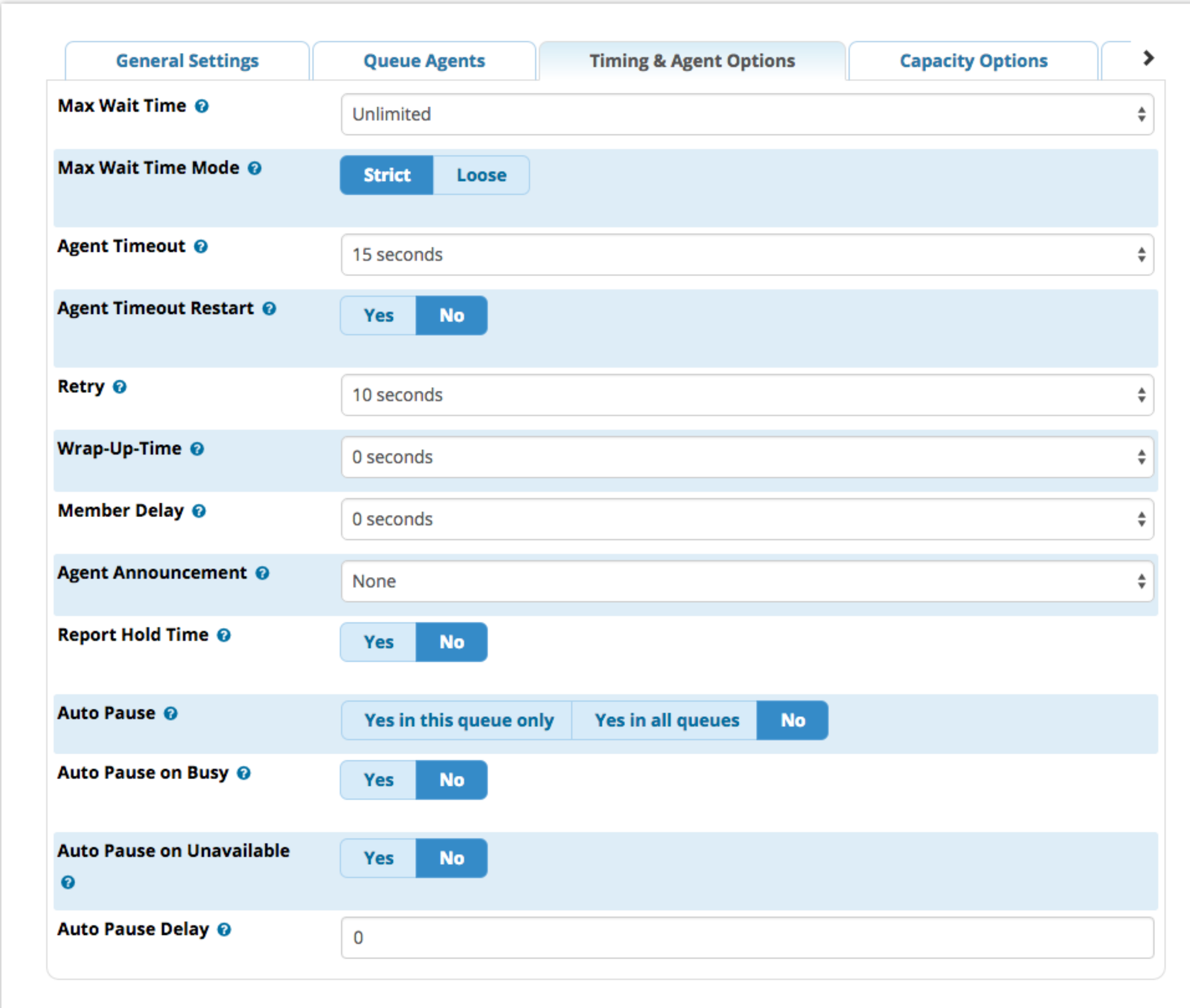
You can put a comma (**,**) after the agent followed by a penalty value to set an agent penalty. It will default to zero.

An advanced mode has been added which allows you to prefix an agent number with S, P, X, Z, D, or A. This will force the agent number to be dialed as an Asterisk device of type SIP, PJSIP, IAX2, ZAP, DAHDI, or Agent, respectively. This mode is for advanced users and can cause known issues in the PBX as you are bypassing the normal dialplan. If your “Agent Restrictions” are not set to “Extension Only,” you will have problems with subsequent transfers to voicemail. Other issues may also exist.

Dynamic Agents

Dynamic agents are extensions or telephone numbers that can log in and out of the queue. Extensions included here will NOT automatically be logged in to the queue.

Timing & Agent Options



Max Wait Time

Defines the maximum number of seconds a caller can wait in a queue before being pulled out. (Default = unlimited) Choose the max wait time from the drop down menu.

Max Wait Time Mode

Set the PBX timeout priority.

- **Strict**: When the “Max Wait Time” of a caller is hit, they will be pulled out of the queue immediately.
- **Loose**: If a queue member is currently ringing with this call, then the PBX will wait until the ringing stops or the call is rejected 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 ring time or individual extension defaults.

Agent Timeout Restart

If set to **Yes**, then the timeout 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 to wait before trying all the phones again. Choosing **No Retry** will exit the queue and go to the failover destination as soon as the first attempted agent times-out. Additional agents will not be attempted.

Wrap-Up-Time

This is how many seconds to wait after a successful call before sending another call to a potentially free agent (default is 0, or no delay). If using Asterisk 1.6+, you can also set the "Honor Wrapup Time" across queues on the Advanced Settings page so that this is honored across queues for members logged on to multiple queues.

Member Delay

If you wish to have a delay before the member is connected to the caller, or before the member hears any announcement message), set this to the number of seconds to delay.

Agent Announcement

Announcement played to the agent prior to bridging in the caller.

Examples: "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 module. Compound recordings composed of 2 or more sound files are not displayed as options since this feature cannot accept such recordings.

Report Hold Time

If set to **Yes**, the caller's hold time will be reported to the agent before the caller is connected to the agent.

Auto Pause

Whether to 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. Options are **Yes in this queue only**, **Yes in all queues**, and **No**.

Auto Pause on Busy

If set to **Yes**, agents will be auto paused immediately (or after the auto pause delay set) if their devices report busy upon a queue call attempt.

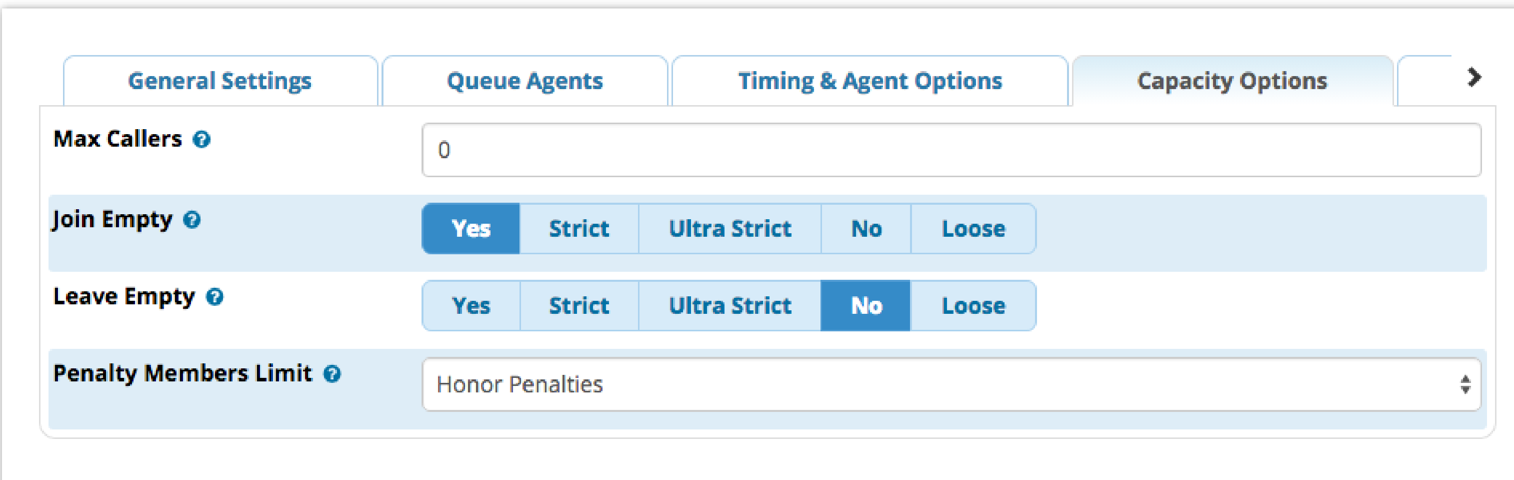
Auto Pause on Unavailable

If set to **Yes**, agents will be auto paused immediately (or after the auto pause delay set) if their devices report congestion upon a queue call attempt.

Auto Pause Delay

This setting will delay the auto pause of an agent by a certain number of seconds after taking 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, the agent will not be auto paused if they don't answer the call. If the agent is presented with a call 120 seconds or later after answering the last call, and they do not answer, they will then be auto paused. If they have taken no calls, this will have no effect.

Capacity Options



Max Callers

Define the maximum number of callers who can be waiting in the queue at the same time (0 for unlimited). If not set to 0, and the maximum capacity is reached, additional calls would be sent to the failover destination set in the General Settings tab.

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 the caller will not enter the queue. If agents are in use or ringing someone else, the caller will still be admitted.
- **Ultra Strict**: Same as “Strict” plus a queue member must be able to answer the phone NOW. Simply speaking, available agents who 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 PBX dialplan).
- **Loose**: Same as “NO,” except callers will be admitted if there are paused agents who could become available.

Leave Empty

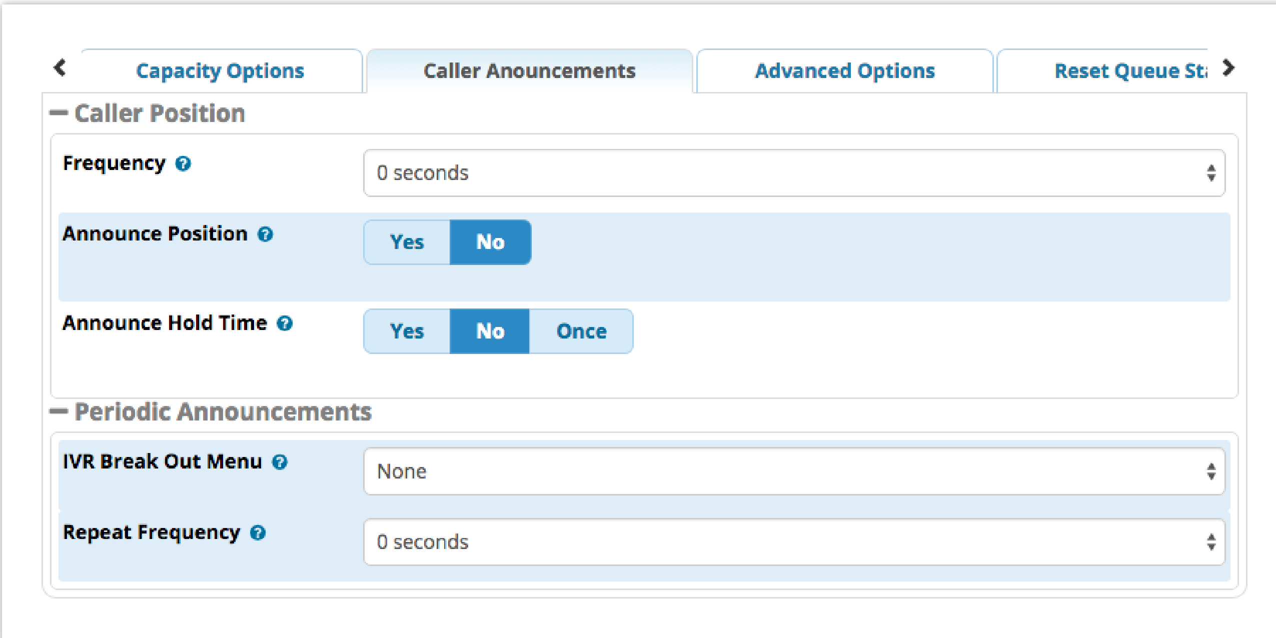
Determines if callers should be excited prematurely from the queue in situations where it appears no one is currently available to take the call. The options include:

- **Yes**: Callers will exit if all agents are paused, show an invalid state for their device or have penalty values less than “QUE_MAX PENALTY” (not currently set in PBX 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, the 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 there 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

Asterisk: penaltymemberslimit. 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.

Caller Announcements



Caller Position

Frequency

Define how often to announce queue position and estimated holdtime (0 to disable announcements).

Announce Position

If set to **Yes**, the system will announce the caller's position in the queue to the caller. For example, "You are number three..."

Announce Hold Time

Yes/No/Once: Whether the system will include the estimated hold time in position announcements. Hold time will not be announced if less than one minute.

Periodic Announcements

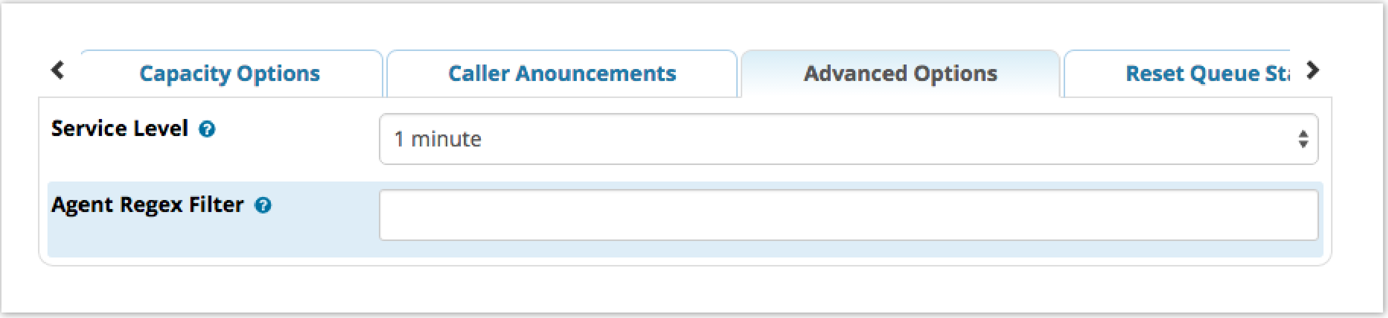
IVR Break Out Menu

You can optionally present an existing IVR as a "Break Out" menu. This IVR must only contain single-digit "dialed options." The recording set for the IVR will be played at intervals specified in "Repeat Frequency" below. The announcement for the selected IVR must be system recording comprised of a single sound file. IVRs that have announcements composed of compount recording files will not be presented as selections for this field.

Repeat Frequency

How often to announce a voice menu to the caller (0 to disable announcements).

Advanced Options



Service Level

Used for service level statistics (calls answered within service level time frame).

Agent Regex Filter

WARNING: Make sure you understand what you are doing, or otherwise leave this blank!

Provides an optional regex expression that will be applied against the agent callback number. If the callback number does not pass the regex filter, it will be treated as invalid. This can be used to restrict agents to extensions within a range, to prevent callbacks from including keys like *, or for any other use that may be appropriate.

Examples:

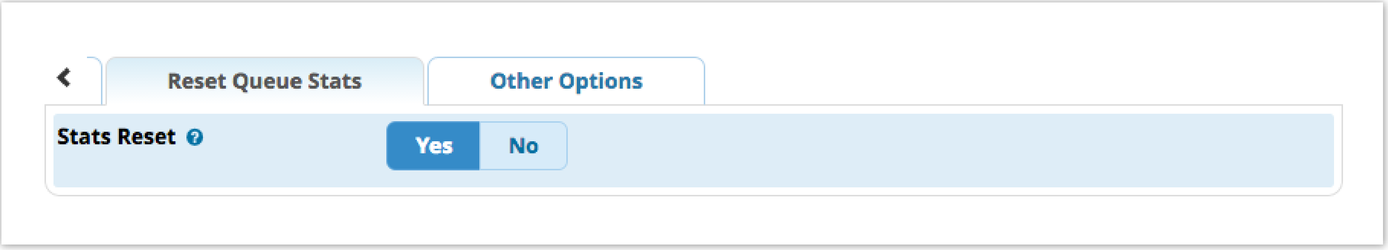
^([2-4][0-9]{3})$

This would restrict agents to extensions 2000-4999.

^([0-9]+)$

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

Reset Queue Stats



Stats Reset

Whether to enable queue statistics reset.

Save

When finished, click the **Submit** button, then click the **Apply Config** button.

Editing and Deleting Queues

You may edit/delete queues by clicking the desired queue in the right side navigation menu.

3.12 Queue Priorities

Queues by default will sort callers with a first-in, first-out order. The Queue Priority module allows you weight some callers differently from others. By giving certain callers a higher priority, they are allowed to bypass all of the other callers with a lower priority to receive faster service. The default setting is for all callers to have a priority of zero. Callers with a higher number will be placed in front of priority zero callers. Queue priorities are often used when providing service level agreements (SLAs).

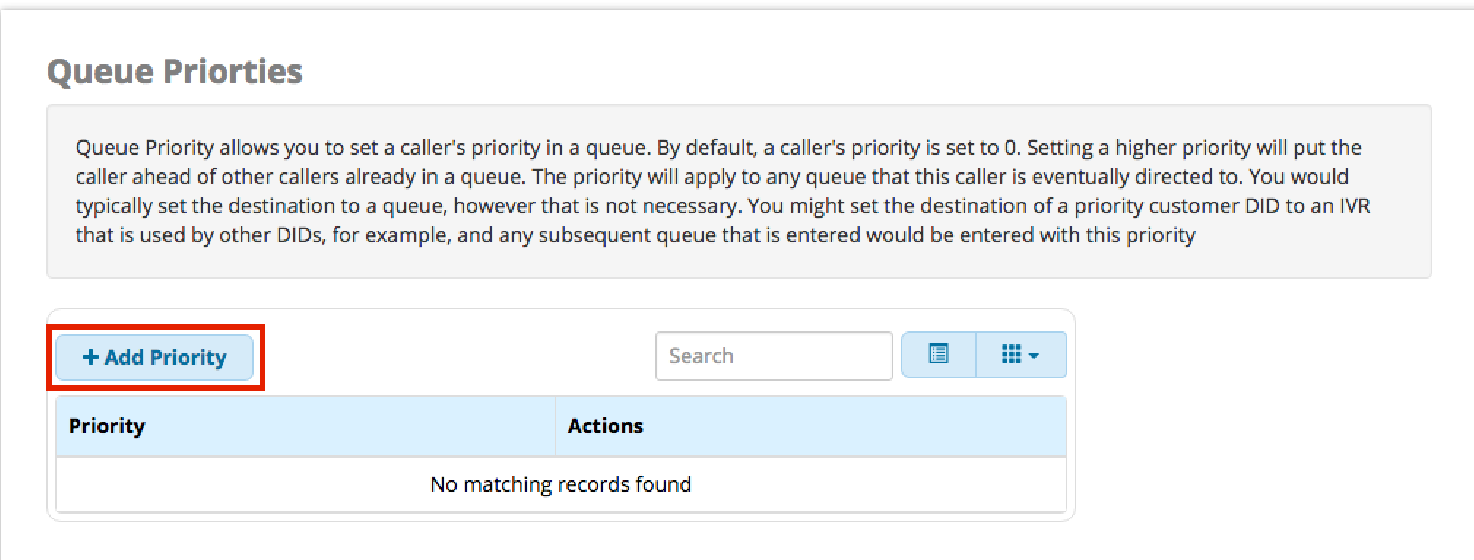
After you have created a queue priority, you can set it as a destination in any other module. A call that flows through the queue priority instance will retain its priority setting even if the next destination is not a queue. The priority would be retained when the caller reaches a queue later in the call flow.

Logging in

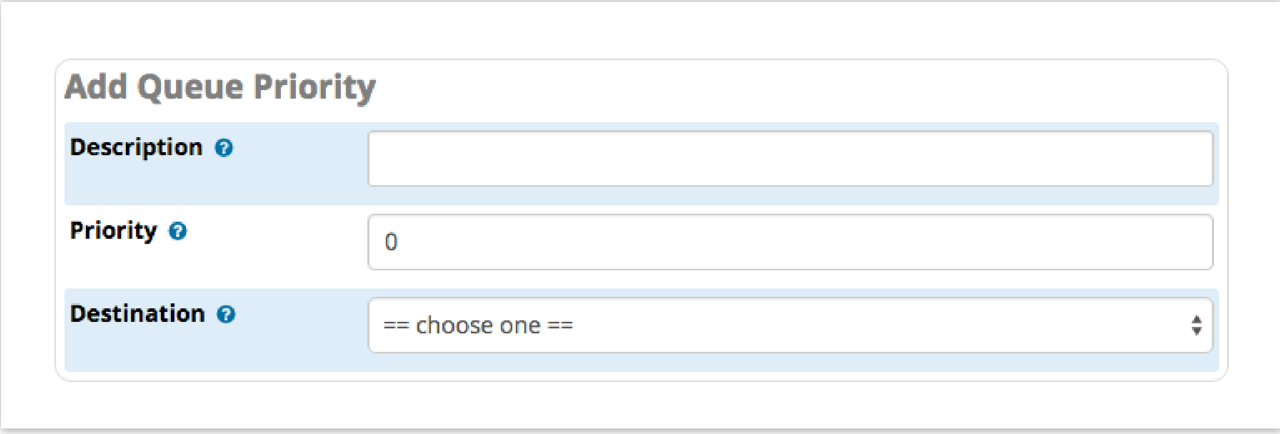
- From the top menu click **Applications**
- In the drop down click **Queue Priority**

Adding a Queue Priority

Click the **Add Priority** button.



Fill out the form as described below.



Description

Create a descriptive name for the queue priority you are setting. We suggest you use a description that easily identifies the queue priority. ie. "VIP Customers"

Priority

Caller priorities can be set from 0-20. The default setting for all calls is zero. The higher the number, the higher the priority assigned.

Destination

This section is for selecting the call target that the caller will be sent to with their new priority. From this point forward, the caller will have a priority weight as set in the priority field above when they enter any queue. Please note that the destination does not have to be a queue. It could be an IVR or any other destination on your PBX. Once assigned a priority, the system will recognize the priority in any queue the call eventually enters.

Save

- Click the **Submit** button**.**
- Click the **Apply Config** button.

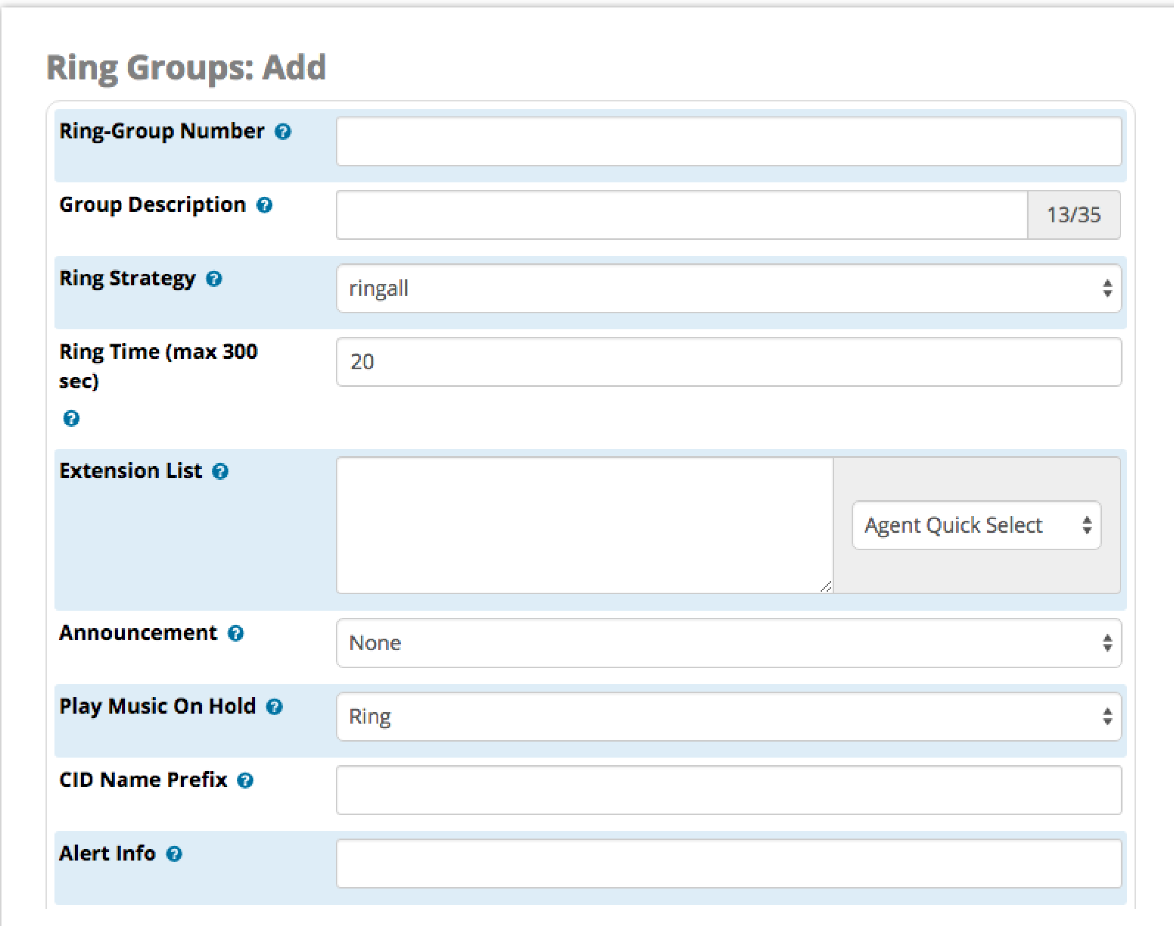
3.13 Ring Group

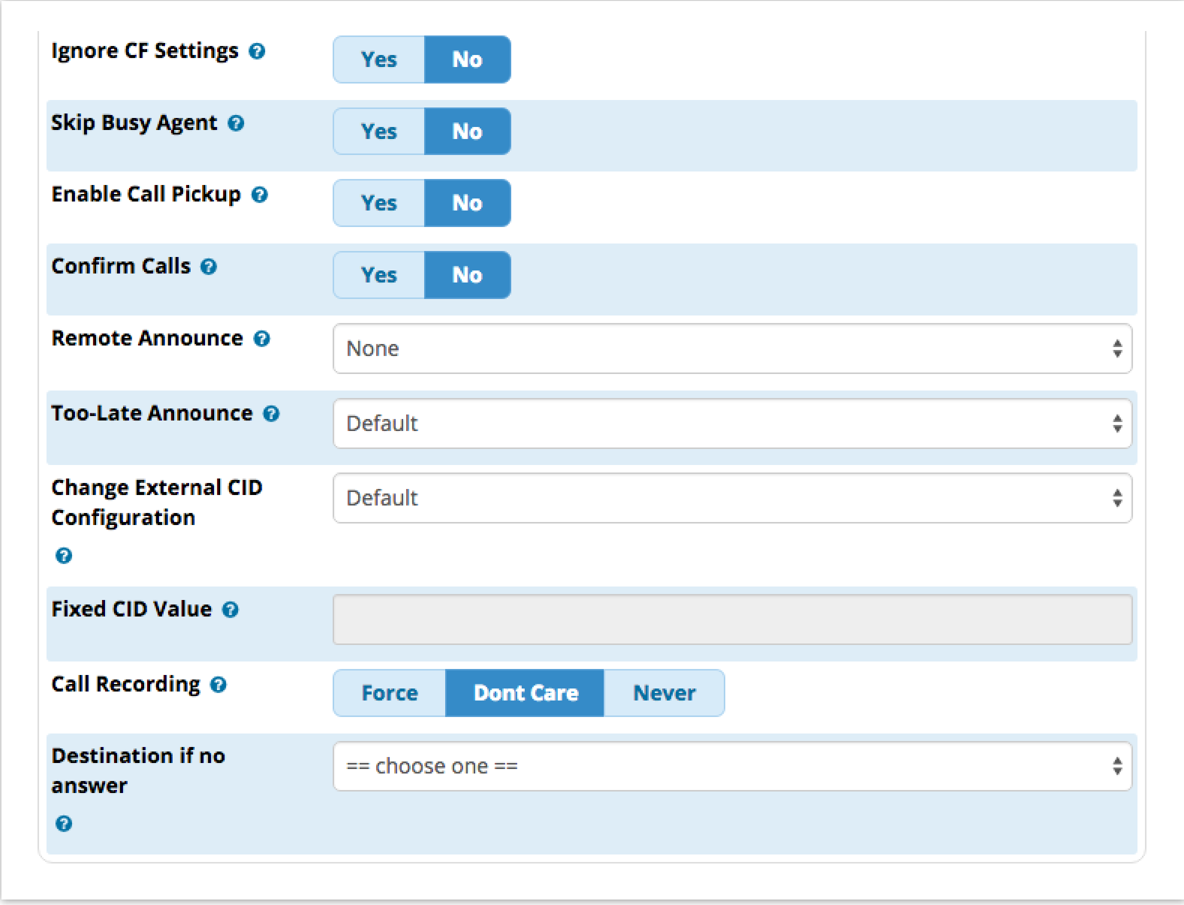
The Ring Groups module provides a method to ring several extensions with a variety of ring strategies. It allows for several useful features such as announcements, CID name prefix, call confirmation, and others. Ring groups can include local extensions and DIDs (which become outbound calls from the system).

Logging In

- From the top menu click **Applications**
- From the drop down click **Ring Groups**

Adding a Ring Group





Ring Group Number

The number to dial to reach this ring group. After the ring group is created, this number cannot be changed.

Group Description

A descriptive title for the ring group to help you identify it.

Ring Strategy

Choose from the following methods in which the extension list can be dialed:

ringall

This will ring all available channels simultaneously until someone answers. This is the default.

hunt

This will take turns ringing each available extension one at a time.

memoryhunt

This will ring the first extension in the list. After that, it will ring the 1st and 2nd together, then the 1st, 2nd and 3rd extensions together, and so on.

***-prim**

This suffix changes the behavior of the other ring strategies. When -prim is selected, the first extension listed becomes the "primary" extension.

- If the primary extension is occupied or in Do Not Disturb (DND) mode, none of the extensions will be rung. Caller is sent directly to the "no answer" destination set at the bottom of the module.
- If the primary is set to Call Forward Unconditional, the primary extension (its call forward number) will not be rung, but the other extensions will be rung.

firstunavailable

This will only ring the first available channel.

firstnotonphone

This will only ring the first channel that is not off-hook, ignoring call waiting.

random

Calls ring extensions without a predefined priority in a random order. This helps spread ring group calls evenly among the group, which can simulate queue behavior when a queue cannot be used.

Ring Time (max 300 sec)

The time, in seconds, that the phones will be rung. For hunt-style strategies, this is the ring time for each iteration. (i.e. Ring first extension for 60 sec., then first and second together for another 60 sec., etc.)

Extension List

List extensions to ring, one per line, or use the “Agent Quick Select” for quick insertion. You can include an extension on a remote system or dial an external number by suffixing the number with a “#.” For example, “3609319999#” would route out whichever trunk is set for 10-digit outbound dialing, and “2000#” would route out whichever trunk is cross-connected to the remote system (via SIP/IAX2) and matches “2XXX” for outbound dialing. See the Outbound Routes user guide for more information.

Extensions that are not suffixed with a “#” will not ring a user's Follow Me. To dial Follow Me, queues, or numbers that are not extensions, use “#” at the end.

Announcement

Message to be played to the caller prior to calling the ring group. Default is "none." You can select a system recording from the drop-down list. To create additional system recordings, visit the System Recordings module.

Play Music On Hold

The default setting is to play ringing to the caller. Alternately, a Music on Hold (MoH) class can be set to play instead of ringing.

CID Name Prefix

(Optional) You can prefix the caller ID name when ringing extensions on the system. For example, a prefix of "Sales:" would make a caller ID name of "John Doe" appear as "Sales: John Doe."

Alert Info

(Optional) ALERT_INFO for devices that support distinctive ringing. For example, "<Bellcore-dr4>." (For supported phones)

Ignore CF Settings

Yes/No: When set to **Yes**, agents who attempt to call forward will be ignored. This applies to call forward all/unconditional (CF), call forward unavailable (CFU), and call forward busy (CFB). Extensions entered with "#" at the end, such as an extension's Follow Me, may not honor this setting.

Skip Busy Agent

Yes/No: When set to **Yes**, agents who are on an occupied phone will be skipped as if the line were returning as busy. This means that call waiting or multi-line phones will not be presented with the call. In the various hunt-style ring strategies, the next agent will be attempted.

Enable Call Pickup

Yes/No: When set to **Yes**, calls to the ring group can be picked up with the directed call pickup feature using the group number. When set to **No**, individual extensions that are part of the group can still be picked up by doing a directed call pickup to the ringing extension. This works whether or not Call Pickup is enabled here.

Confirm Calls

Yes/No: Set this to **Yes** if you're calling external numbers that need confirmation. For example, a mobile phone may go to voicemail, and that will pick up the call. When **Confirm Calls = Yes**, the remote side must press "1" on their phone before the call is put through. Note that 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 above. Default is "none." You can select a system recording from the drop-down menu. To create additional system recordings, visit the System Recordings module.

Too-Late Announce

Message to be played to the person receiving the call, if the call is accepted by someone else before they press "1." You can select a system recording from the drop-down menu. To create additional system recordings, visit the System Recordings module.

Change External CID Configuration

Select from the following modes.

Default

This transmits the caller's CID if allowed by the trunk.

Fixed CID Value

This always transmits the "Fixed CID Value" entered below.

Outside Calls Fixed CID

This will transmit the "Fixed CID Value" value only on calls that come from the outside. Internal extension-to-extension calls will still operate in default mode.

Use Dialed Number

This will transmit the number that was dialed as the CID for calls coming from the outside. Internal extension-to-extension calls will still operate in default mode. There must be a DID on the inbound route for this. This will be blocked on trunks that block foreign caller ID.

Force Dialed Number

This will transmit the number that was dialed as the CID for calls coming from the outside. Internal extension-to-extension calls will still operate in default mode. There must be a DID on the inbound route for this. This will be transmitted on trunks that block foreign caller ID.

Fixed CID Value

When needed, enter your “Fixed CID Value” here. Enter digits only, except the “+” prefix can be used with the E164 format.

Call Recording

Force/Don't Care/Never: 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 (**Don't Care**). If recording is denied, then one-touch on-demand recording will be blocked, unless the user has the "Override" call recording privilege.

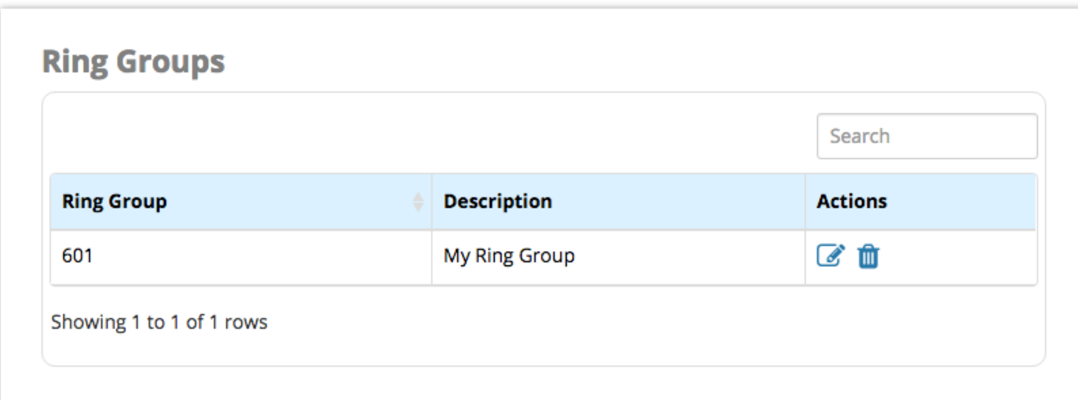
Destination if no answer

Choose where to send the call after the ring time has been exceeded or after a -prim mode prevents ringing the group. Most often, this is set to an extension or a general voicemail box.

Save

When finished click the **Submit** button, then click the **Apply Config** button.

Editing / Deleting a Ring Group



From the module home screen:

- **To Edit:** Click the edit button , make changes, click the **Submit** button, then click the **Apply Config** button.
- **To Delete:** Click the trash can button , click the **OK** button in the alert window to confirm deletion, then click **Apply Config**.

3.14 Set CallerID

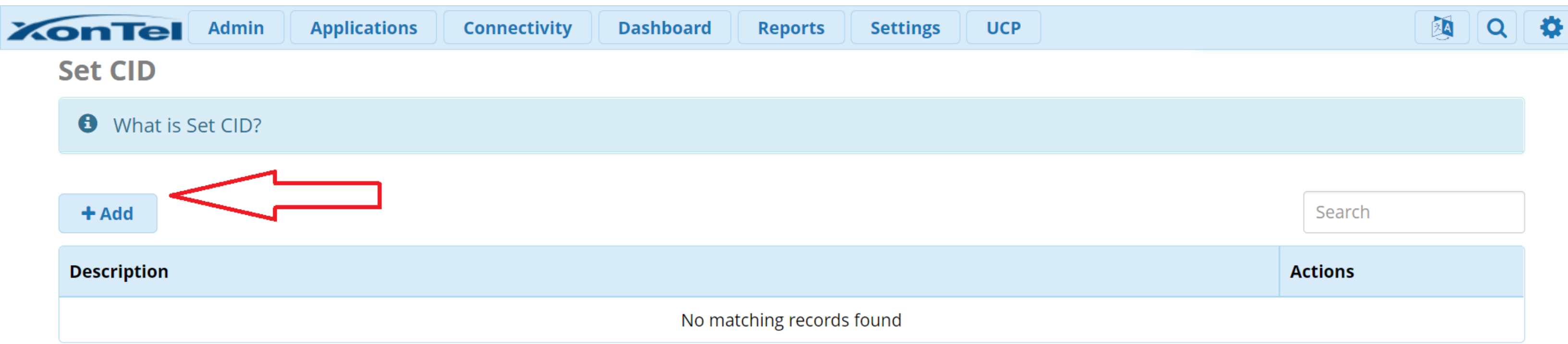
The Set CallerID module is a simple and effective way to manipulate caller ID (CID) within the call flow to help identify who is calling, use the proper greeting for a caller, give priority, or even handle calls from multiple companies. The module allows you to change the caller ID of a call and then continue on to the desired destination.

Logging In

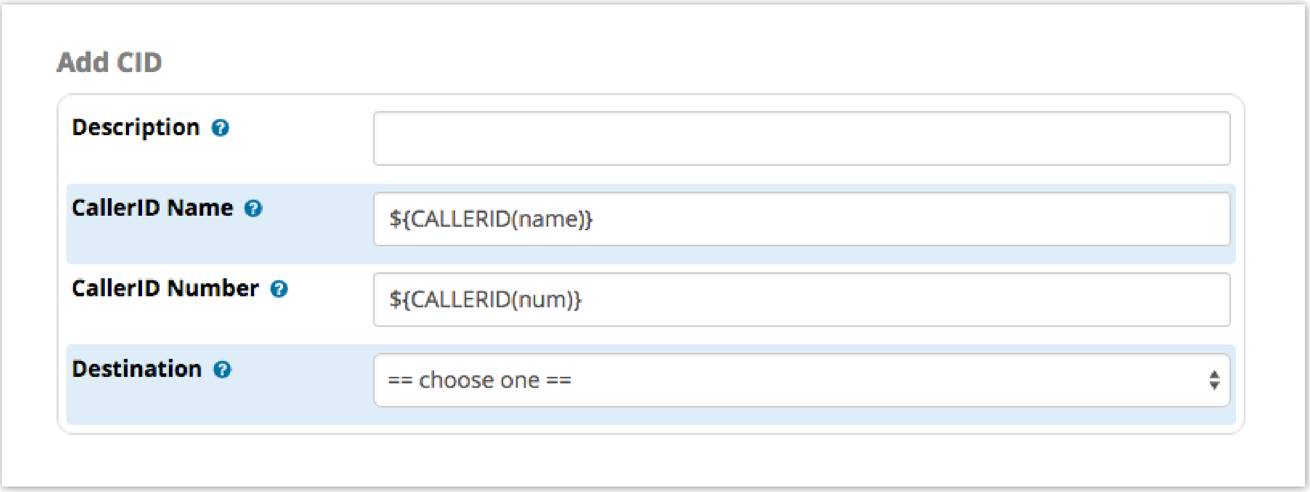
- From the top menu click **Applications**
- In the drop down click **Set CallerID**

Creating a Set CallerID Instance

Click the **Add** button.



Fill out the four elements of the form:



Description

Enter a descriptive name for this CallerID Instance to help you identify its purpose. Example: “Sales CID”

CallerID Name

The caller ID name will be changed to this. If you are appending to the current caller ID name, don’t forget to include the appropriate variables. If you leave this box blank, the caller ID name will be blank. Default caller ID name variable: ${CALLERID(name)} See the "Working With Variables" section below.

CallerID Number

The caller ID number will be changed to this. If you are appending to the current caller ID number, don’t forget to include the appropriate variables. If you leave this box blank, the caller ID number will be blank. Default caller ID number variable: ${CALLERID(num)} See the "Working With Variables" section below.

Destination

Choose the target destination to continue the call. The call will flow to this destination with the new CallerID Name and Number set.

Save

Click the **Submit** button and then click the **Apply Config** button to save the changes.

Working With Variables

Note: This uses Asterisk variables which can be modified in the same manner as if you were writing a dial plan.

Modifiers:

Example	**Description**
${VARIABLE:**n**}	Skip **n** characters
${VARIABLE:**-n**}	Only grab last **n** characters
${VARIABLE:**s**:**n**}	Starting at character **s** grab **n** characters

Examples:

Description	**Variable**	**Input**	**Output**
Strip + From a phone number	${CALLERID(num)**:1**}	+4805551212	4805551212
Add a 1 to the phone number	**1**${CALLERID(num)}	4805551212	14805551212
Replace caller name with account code (assuming account code is set to 12345)	**${CDR(accountcode)}**	John Smith	12345

3.15 Time Groups

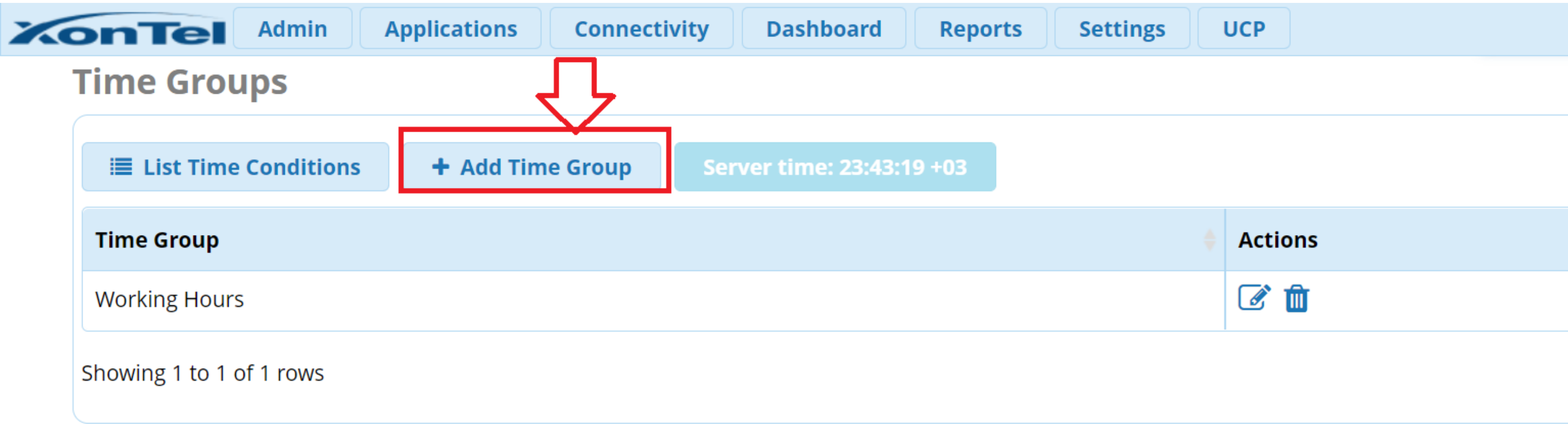
A Time Group is a list of times against which incoming or outgoing calls are checked. The rules specify a time range, by the time, day of the week, day of the month, and month of the year. Each time group can have an unlimited number of rules defined. Time groups typically are associated with time conditions, which control the destination of a call based on the time. A time group can also be assigned to an outbound route in order to limit the use of that route to the times defined in the time group.

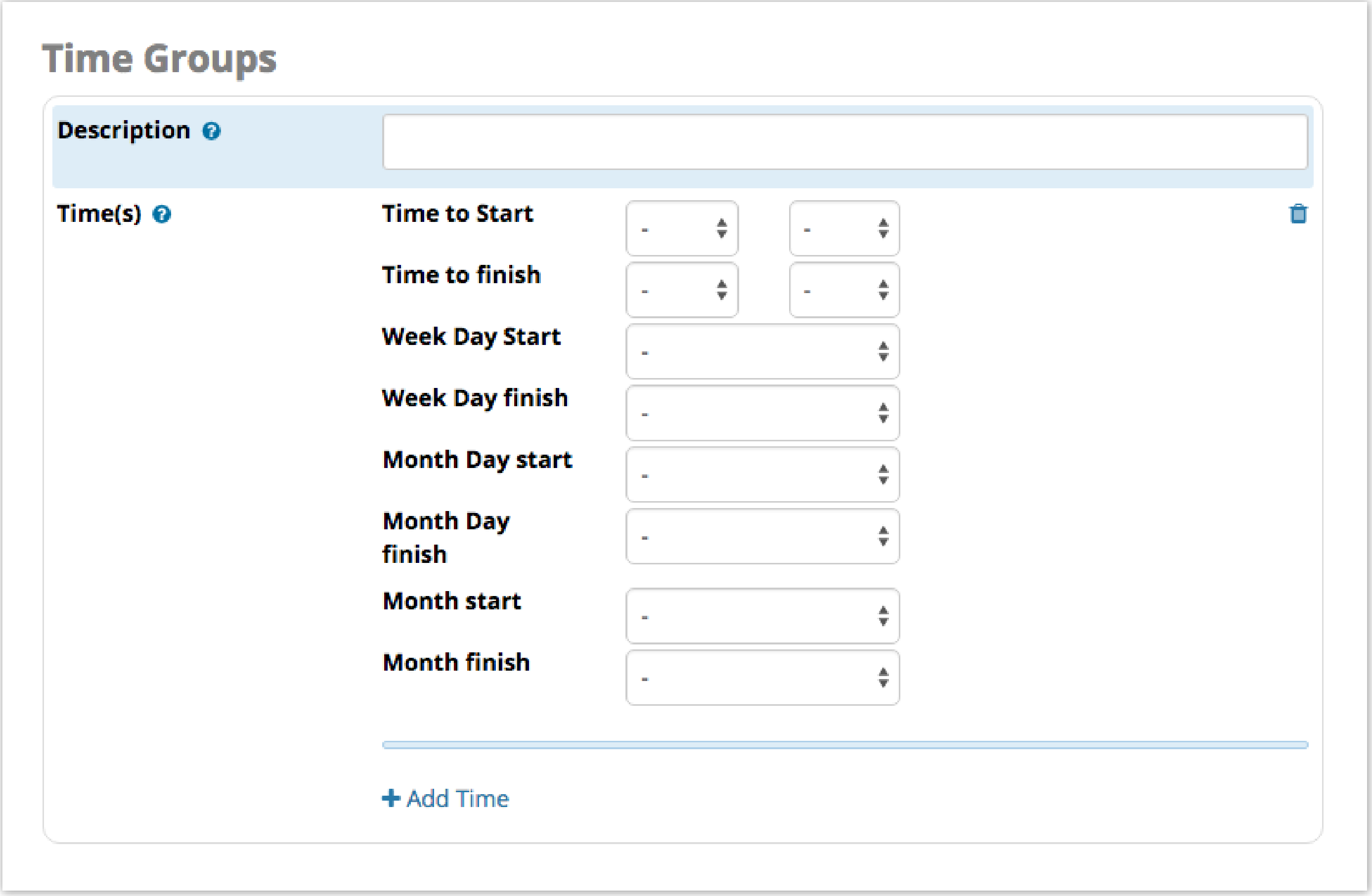
Logging In

- On the top menu click **Applications**
- In the Drop down click **Time Groups**

Adding a Time Group

Click the **Add Time Group** button.





Description

Enter a description to identify this time group. For example, “Closed Hours” works better to something generic like “Time Group 1.”

Time(s)

This is where you will define a time range. By default, there is one range available. You can define multiple ranges in the same time group by clicking the **Add Time** button.

The Time Group will evaluate to "True" during the times/days/months you define.

Available parameters are:

- **Time to start**
- **Time to finish**
- **Week Day start**
- **Week Day finish**
- **Month Day start**
- **Month Day finish**
- **Month start**
- **Month finish**

Tip: Unset Parameters

Unset (blank) week day, month day, and month parameters will default to "all." For example, setting a start time of 09:00 and an end time of 17:00, and nothing else (no day, month, etc.), will make the condition true from 9AM to 5PM every day of the week, every day of the month, every month of the year.

If ***times*** are unset (blank) *and* there is also a week day, month day, and/or month range set, the day/date range will be considered an *exclusion*. You are essentially telling the system, "I want *no time* during this day/date range to be considered a match." You can use this technique to exclude certain days/dates from a broader time period. See examples below.

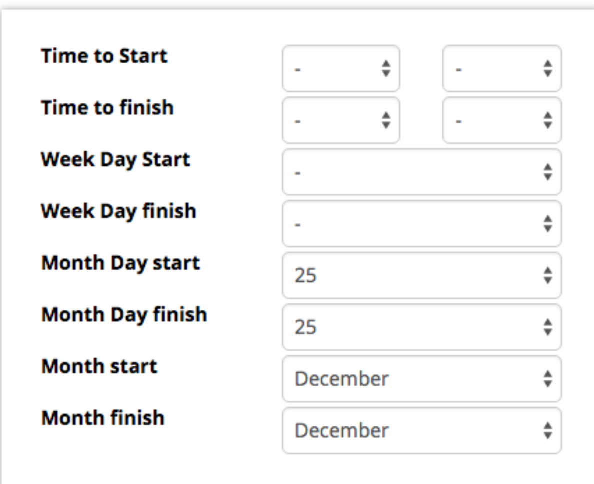
Excluding Time Periods Such as Holidays

After you have defined your "normal" or default time period(s) by adding one or more time ranges as described earlier, you can then add entries to *exclude* certain dates.

To define an excluded period, leave the **time to start** and **time to finish** *BLANK*. (Make no selection for times). Then, select a week day, month day, and/or month range in which you want this exclusion to apply.

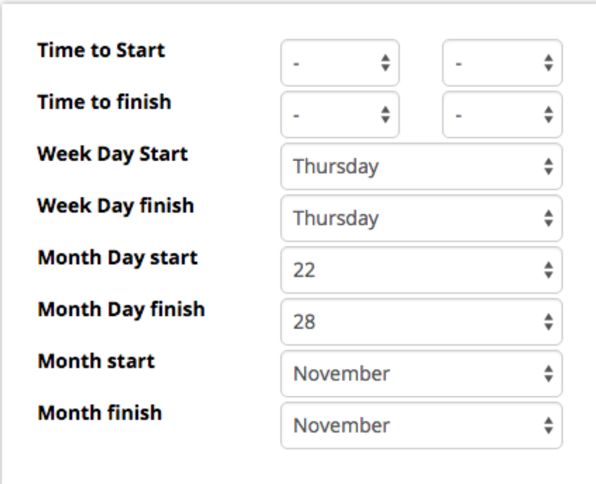
Date-Specific Holidays:

Holidays such as Christmas, which always fall on the same calendar date, can be set by choosing the day of the month (for both start & finish) and the month (for both start & finish).

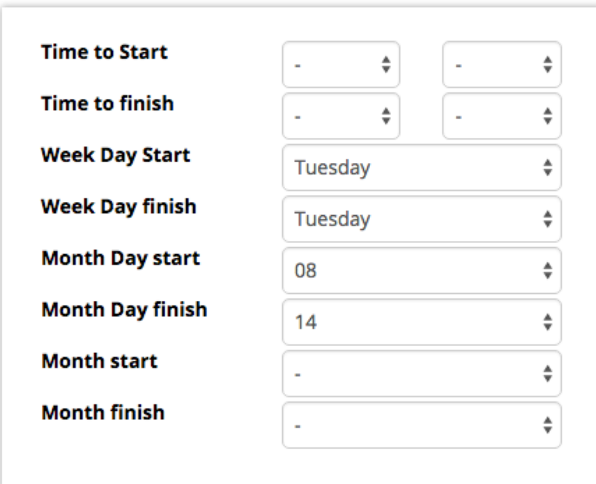


Floating Holidays:

Floating holidays - those that do not always fall on the same calendar date - require a bit more logic. You want the system to look for a specific day of the week within a possible date range. For example, Thanksgiving in the U.S. is the 4th Thursday of November. The possible dates are 11/22 through 11/28. To set an exclusion for Thanksgiving, you'd ask the system to look for a Thursday within that date range in November:



This type of logic can be applied to any "floating" date, such as "the second Tuesday of each month," for example:



Save

Click the **Submit** button, then click the **Apply Config** button.

3.16 Time Conditions

The Time Conditions module defines a set of rules based on time groups. A time condition has two call destinations, one if the time of the call matches the time group assigned, and another if there is no match. Time conditions are often used to control how the PBX routes calls during business hours vs. outside business hours.

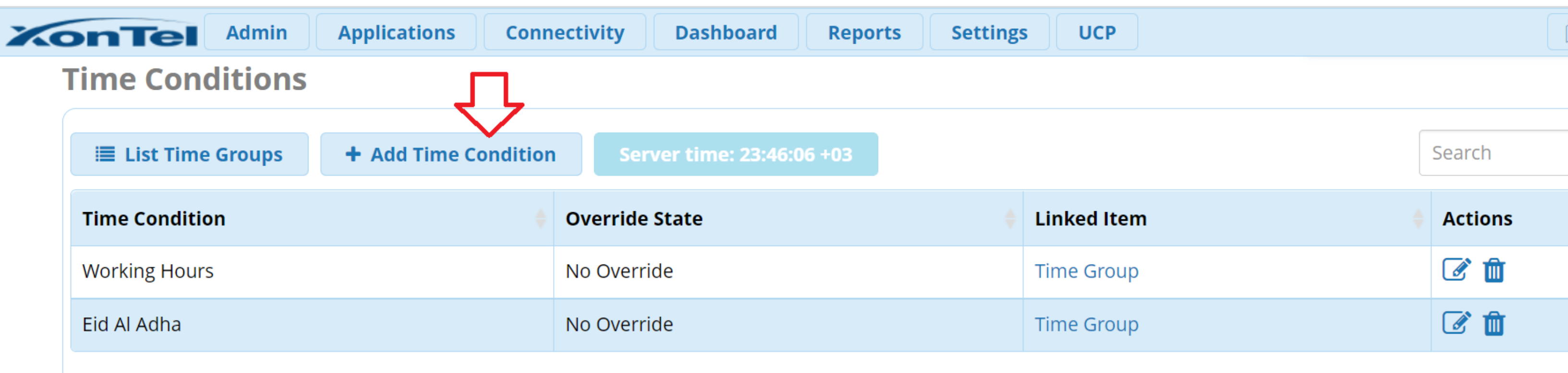
Time groups should be set up prior to setting up time conditions. Visit the Time Groups module under the Applications menu. Because the Time Conditions module depends on time groups, there are two easy ways to reach the Time Groups module from within the Time Conditions module. One is to click the **List Time Groups** button at the right side of the screen, and the other is to click the clock button next to an existing time condition.

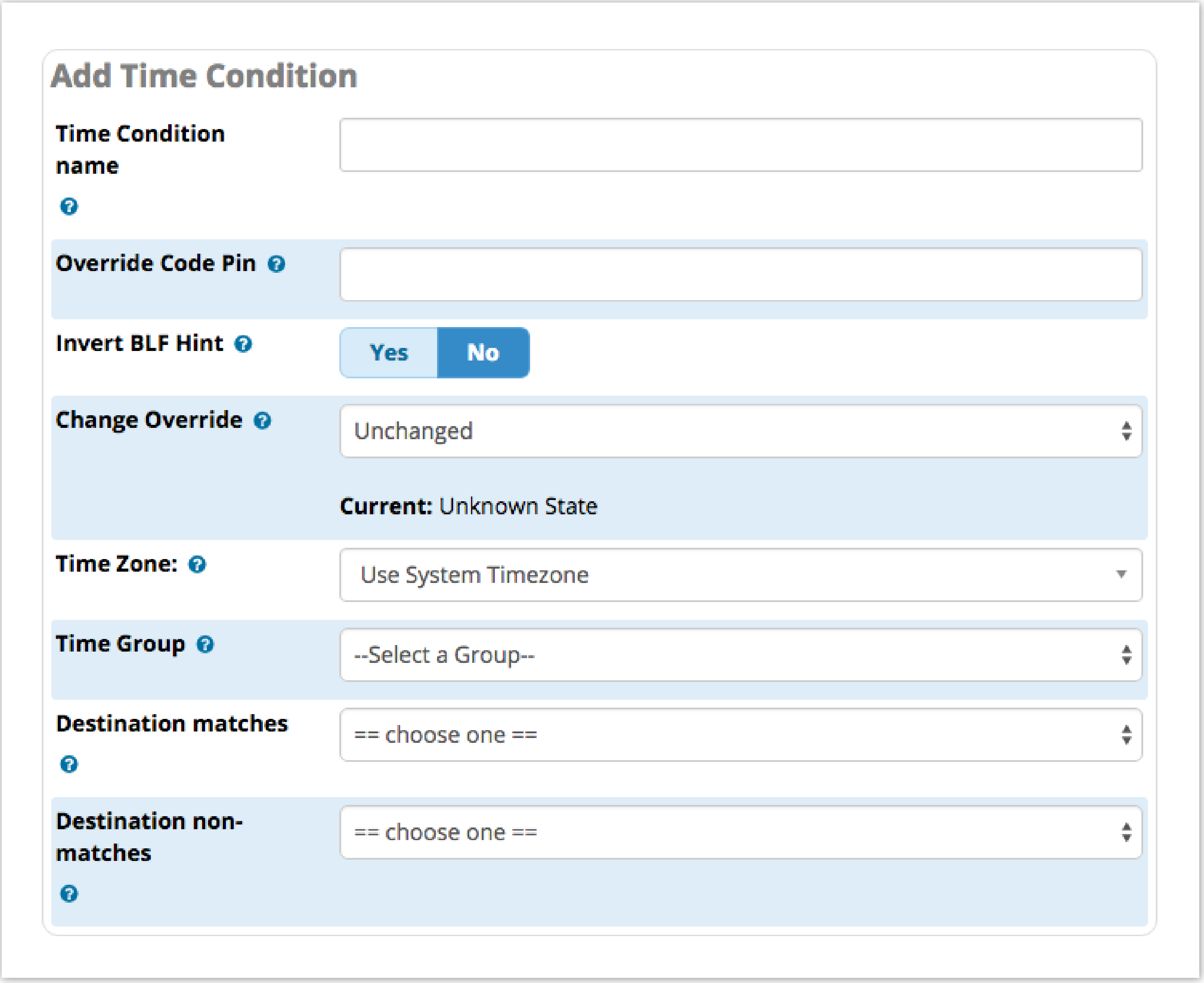
Logging in

- From the top menu click **Applications**
- From the drop down click **Time Conditions**

Creating a Time Condition

Click the **Add Time Condition** button to add a new time group.





Time Condition Name

Enter a description to identify this time group. For example, “Closed Hours” works better than something generic like “Time Condition 1.”

Override Code Pin

(Optional) If a PIN is entered here, users will be prompted to enter the PIN after dialing the override feature code. A PIN can help prevent unauthorized changes. If no PIN is entered here, users will be able to override the time condition by dialing the feature code.

Invert BLF Hint

Yes/No: Whether to invert the behavior of the busy lamp field (BLF) for this time condition. Depending upon the way the time condition is set up, and depending upon the BLF light color behavior some phones (red vs. green), the default setting can be confusing to some users. Therefore, you have the option to change the BLF behavior.

By default (**Invert BLF Hint = No**), the BLF hint is "INUSE" when the time condition is NOT matched, and "NOT_INUSE" if the time condition is matched.

If **Invert BLF Hint = Yes**, the behavior will be the reverse of what is described above. If set this way, the BLF hint will be "INUSE" if the time condition is matched, and "NOT_INUSE" if the time condition is NOT matched.

Change Override

The current override status of the time condition is displayed here below the drop-down menu. If this is a new time condition that has not yet been saved, the state will be "Unknown." Otherwise, you will see the current state.

The drop-down menu gives you the opportunity to change the override status:

- **Unchanged**: The override state will not be changed.
- **Reset Override:** Removes any override that is set.
- **Temporary Matched / Unmatched:** Creates a temporary override that will send calls to the matched or unmatched destination (whichever is selected) until the current time span has elapsed. After that, the behavior will return to normal. A temporary override can be set and removed by a feature code, here in the GUI, or by other applications such as an XML-based phone option.
- **Permanent Matched / Unmatched:** Creates a "permanent" override that will send calls to the matched or unmatched destination (whichever is selected) until the override is removed. This override will not automatically be reset after a time span has elapsed. A permanent override cannot be set via a feature code, but it can be *removed* by a feature code. A permanent override can *only* be set here in the GUI or by other applications such as an XML-based phone option.

Time Group

The time group this time condition will be checked against. A time group defines the times that are considered a "match." You can create new time groups in the Time Group module.

Destination matches

This destination will be used as the call target when the current time matches the time group selected above.

Destination non-matches

This destination will be used as the call target when the current time does not match the time group selected above.

3.17 Voicemail Blasting

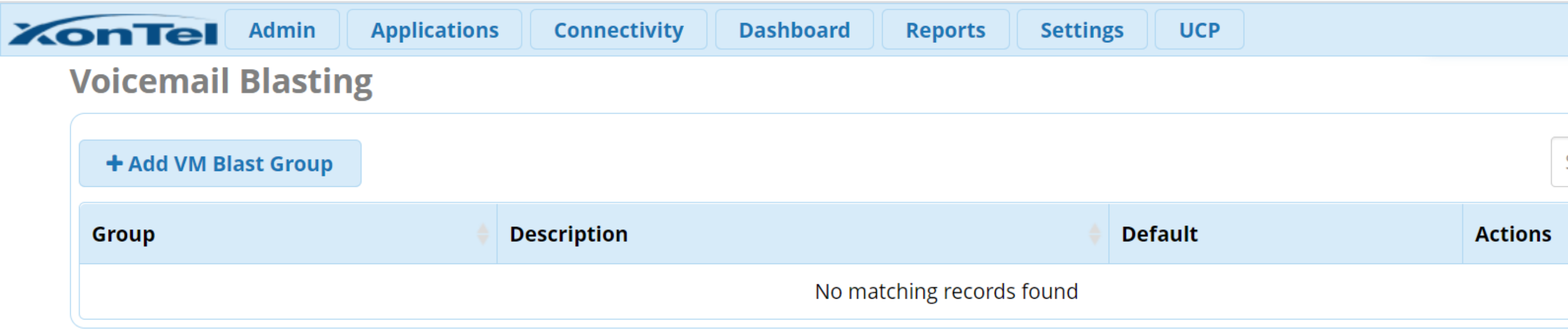
The Voicemail Blasting module is used to assign a voicemail blast (VMBlast) number to a group of users. A user can dial this number to leave a voicemail message for the group. All members of the group will receive the message in their voicemail boxes.

Logging in

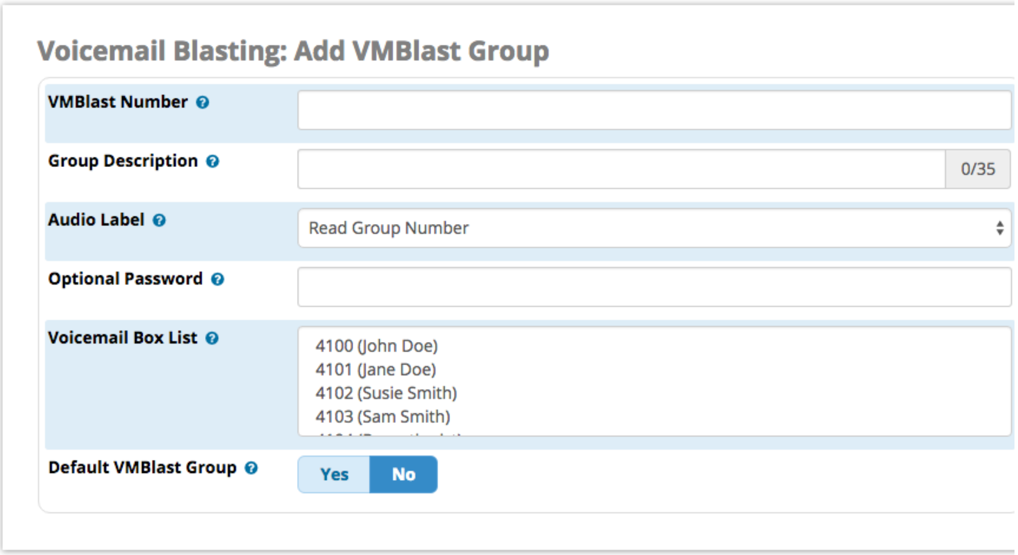
- From the top menu click **Applications**
- In the drop down click **Voicemail Blasting**

Adding a VMBlast Group

Click the **Add New VM Blast Group** button.



Fill out the form as described below.



VMBlast Number

Enter the number that users will dial to access the VMBlast Group. This number must not conflict with an existing extension number.

Group Description

Provide a descriptive name for the VMBlast Group. This is a mandatory field.

Audio Label

Select which message to play to the person leaving the voicemail.

- **Read Group Number**: The default setting. The system will read the VMBlast group number. This can help the caller confirm they have called the proper VMBlast group number before leaving a message.
- **Beep Only - No Confirmation** - The system will play a beep to the caller, and the caller can begin recording after the beep.

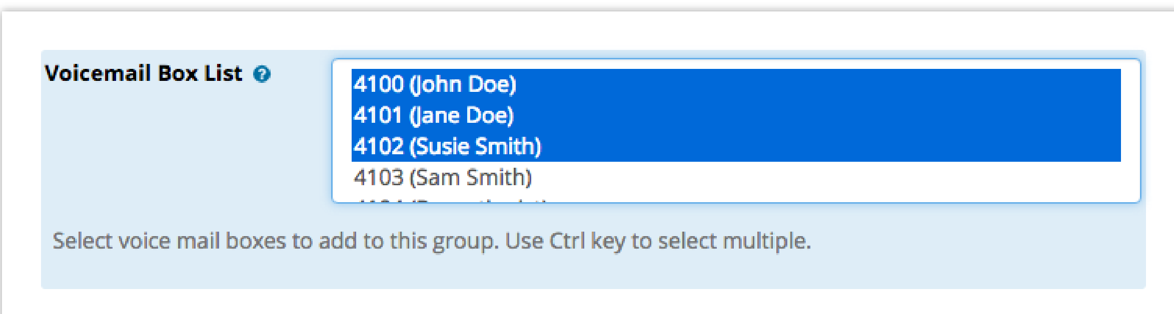
Optional Password

(Optional) Specify a numerical password to protect the VMBlast group from accidental use.

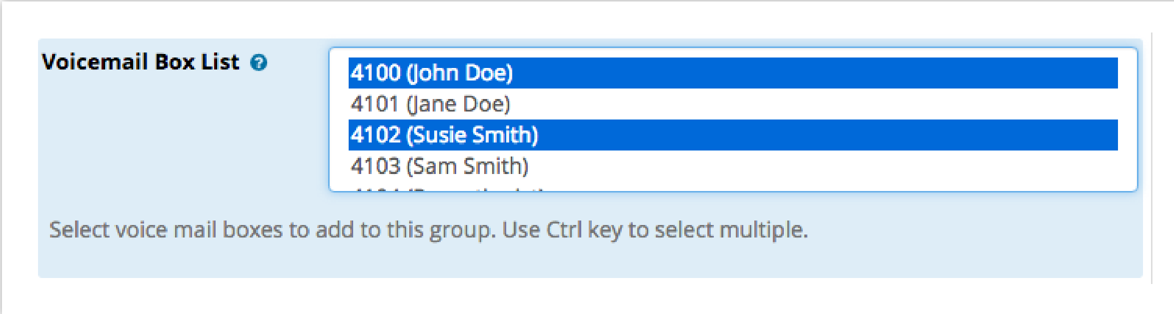
Voicemail Box List

Click on voicemail boxes to add them to this group.

Hold down the SHIFT key and click to select multiple sequential users:



Hold down the CTRL or command key and click to select multiple non-sequential users:



Default VMBlast Group

Yes/No: Whether to designate this VMBlast group as the default.

Each PBX system can have a single Default Voicemail Blast Group. If you designate a new group as the default, any other group that was previously the default will no longer be the default. Extensions can be automatically added (or removed) from the default group in the Extensions (or Users) module. This prevents the need to revisit the Voicemail Blasting module each time a new extension is created.

Save

Click the **Submit** button, then click the **Apply Config** button.

4 Connectivity

4.1 Autoclip config

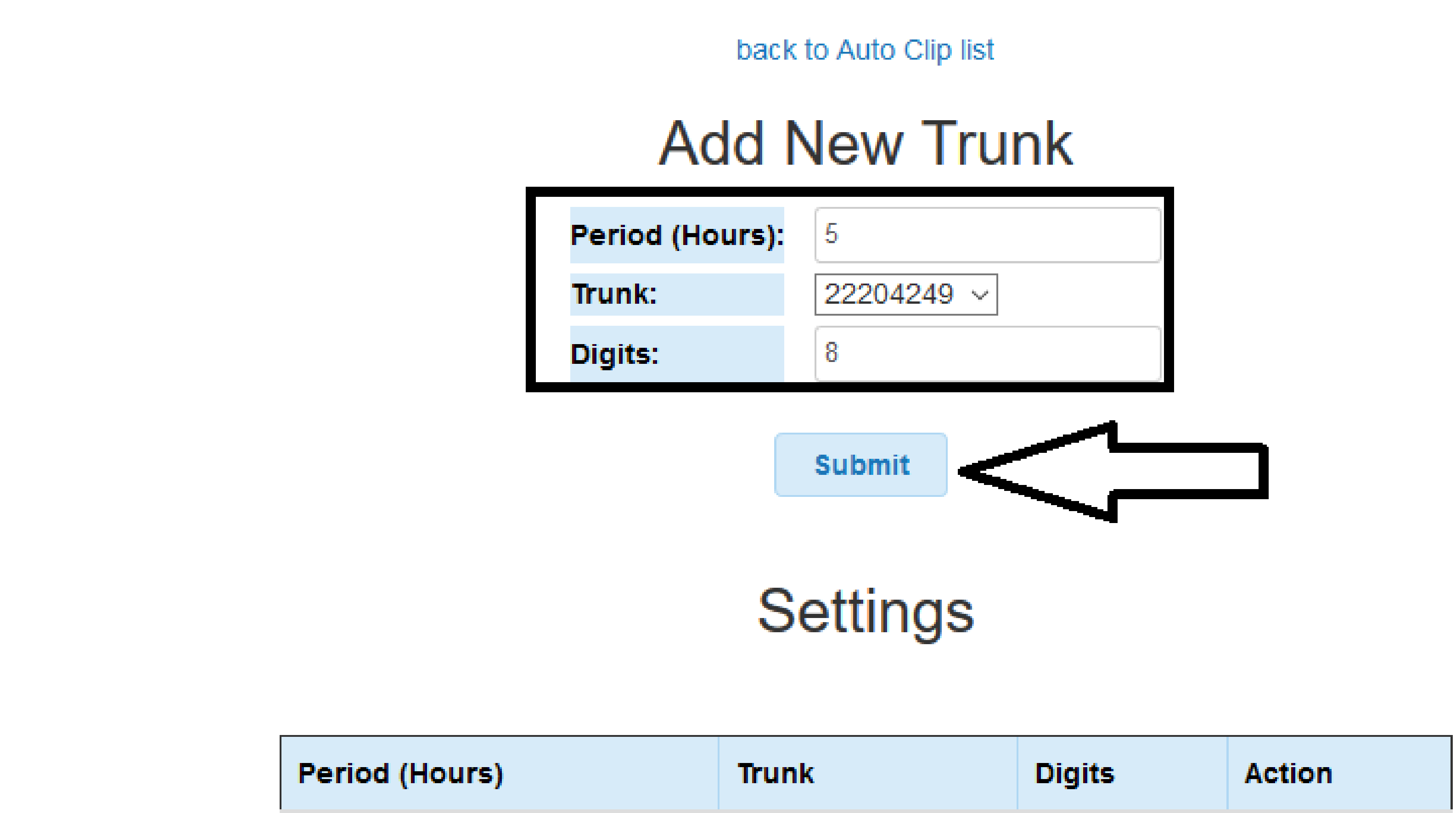
XonTel MS PBX can automatically stores records of outgoing call to Autoclip route table. When the called person calls back, the call will be routed directly to the original caller's extension.

How to Set up Autoclip Routes?

Follow the steps below to configure Autoclip route.

Step 1. Configure Autoclip Parameters.

Go to **Connectivity** > **Autoclip config** to configure Autoclip parameters on **Autoclip config** page.



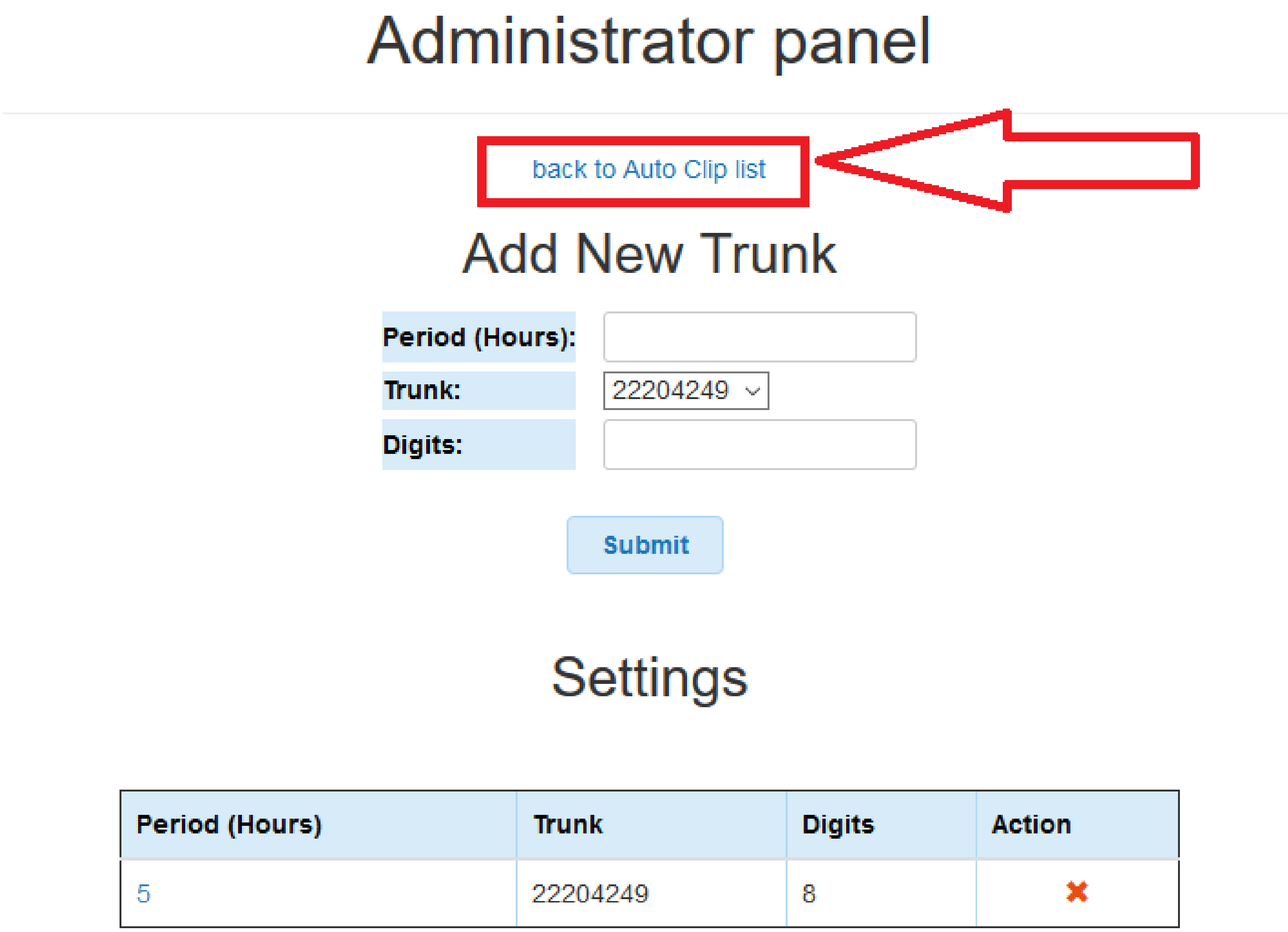
- **Period (Hours):** set the time duration in hours for which records are kept in Autoclip list.

- **Trunk:** only the call back to PBX through the same trunk will be matched against the Autoclip list. For example, if the PBX user **300** call to user **97978301** through the trunk, only when the number **97978301** calls back to the trunk, Autoclip feature will work.

- **Digits:** define how many digits from the last digit of the incoming caller ID will be used to match the Autoclip record.

Step 2. Click Submit and Apply.

To test Autoclip feature click " **back to Auto Clip list** " to check the Autoclip records.



Info panel

AutoClip List

delete

☐	Calldate	Extension Number	Called Number	Trunk name	Delete
☐	2020-02-07 21:28:50	371	65996600	22204249	✖

The Autoclip feature will work as we will see in the following steps:

1. Extension user **371** makes a call to number **65996600**, the called party doesn't answer the call.

2. The number **65996600** calls back to PBX.

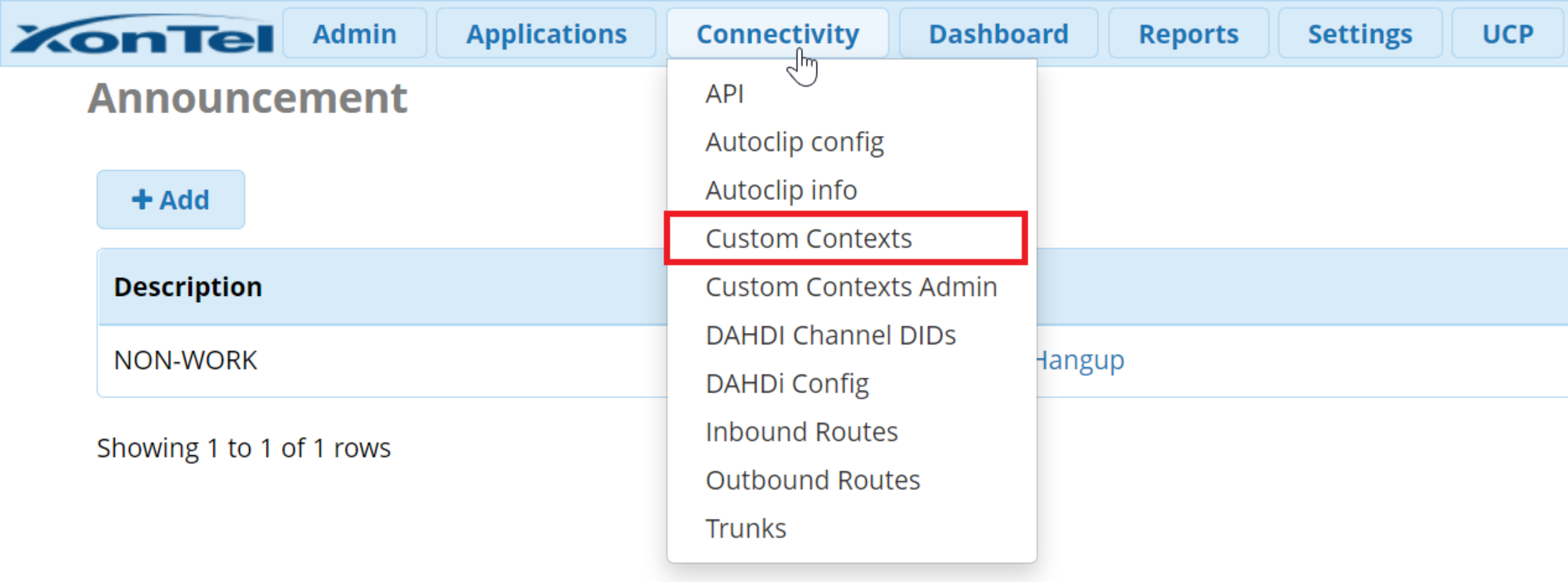
3. As we set **Digits Match** to "**8**", so the call will match against the Autoclip records, and the call will be forwarded directly to the extension user **371**.

4. The record will be deleted automatically in Autoclip and next time when the number **65996600** calls the PBX, no Autoclip route will be matched and the call will go to the inbound route destination.

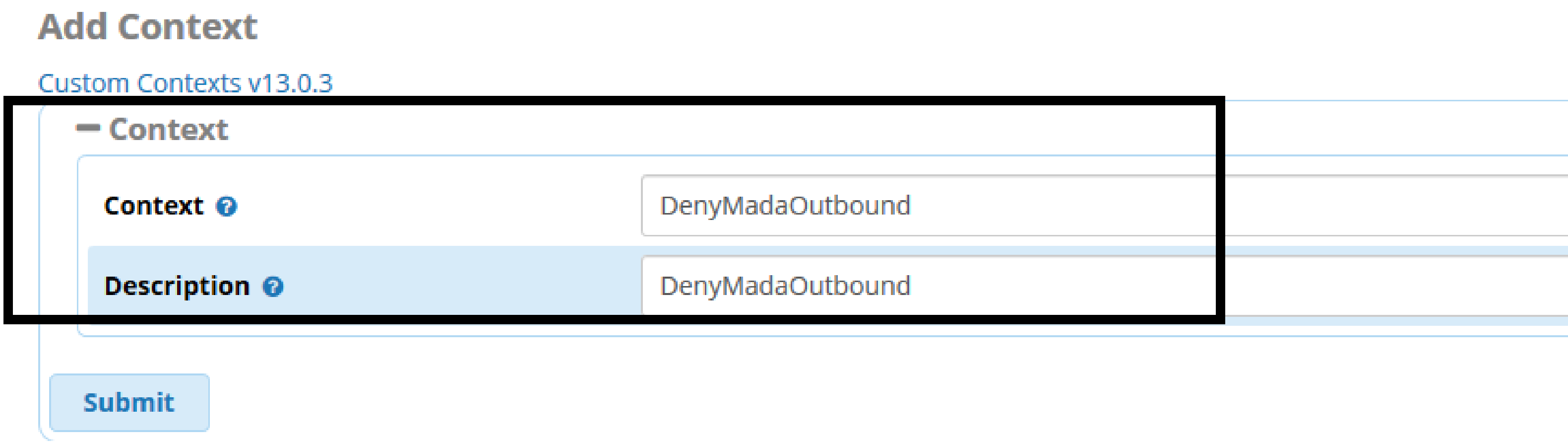
4.2 Custom Context

In this section, we will explain a general way to allow or deny the use of features, trunks, and outbound routes from the PBX.

1. Go to Custom Contexts configuration page.

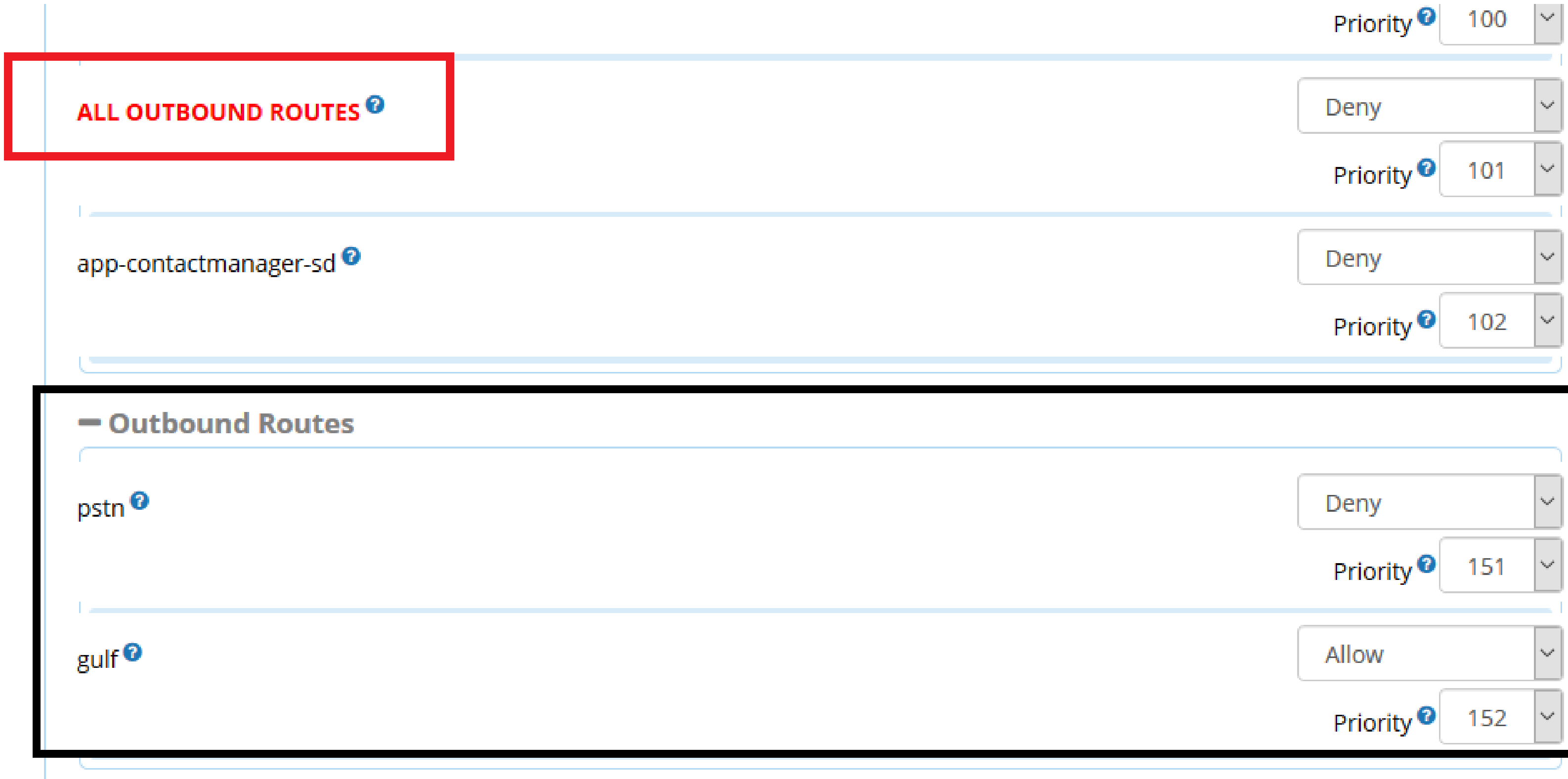


2. Create new Custom Context.



3. In the Custom Context that you create go to the ALL OUTBOUND ROUTES section.

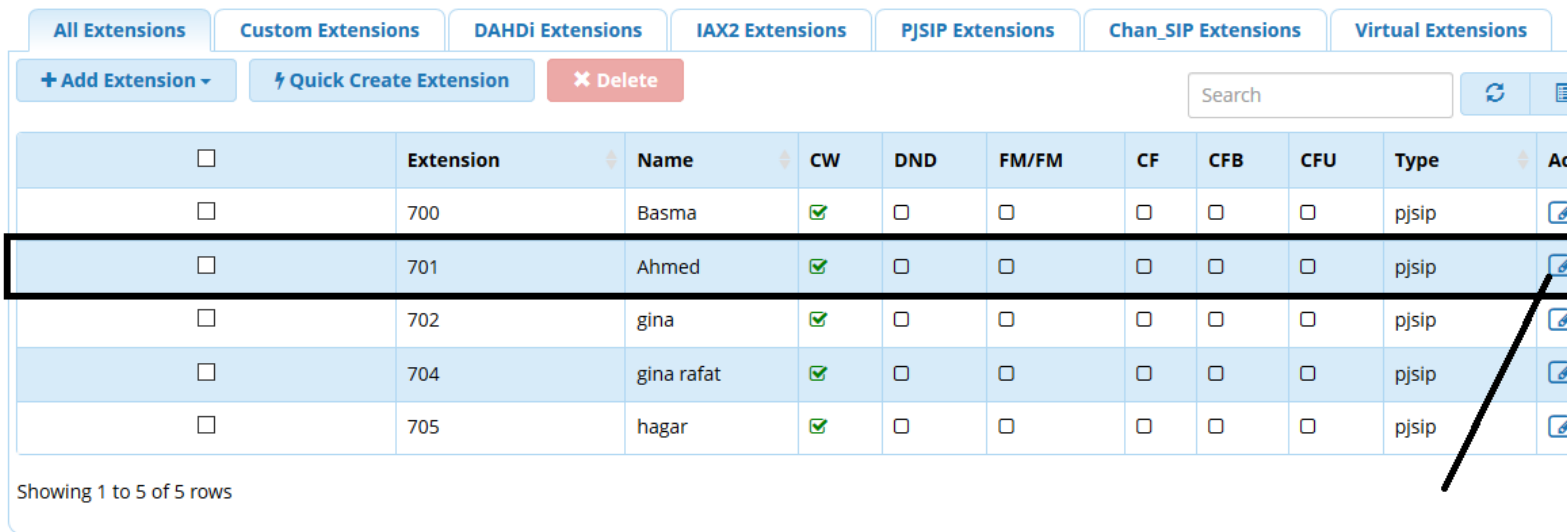
4. Select which outbound route this context can use.



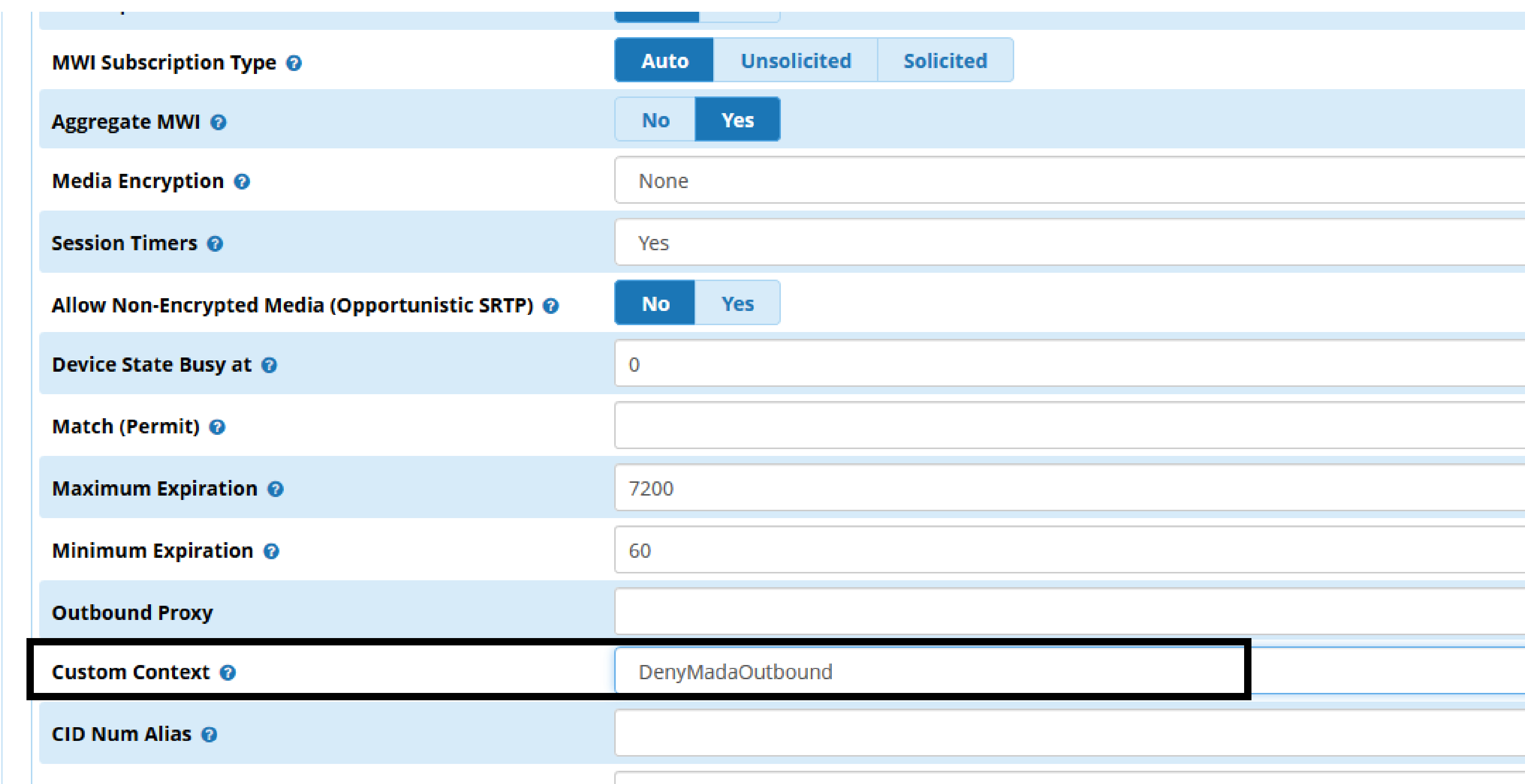
5. Save and apply the changes.

6. Go to the Applications | Extensions menu

7. Choose the extension to restrict its calls.



8. From extension Advanced options in the Custom Context menu, choose the Custom Context that you create.



4.3 Inbound Routes

Inbound routing is one of the key pieces to a functional PBX. The Inbound Routes module is the mechanism used to tell your PBX where to route inbound calls based on the phone number or DID dialed. This module is used to handle SIP, PRI and analog inbound routing. Setting up inbound routing properly is a critical step in the deployment of a PBX system. Inbound routes are often used in conjunction with time conditions and IVRs. A typical setup will go from an inbound route to a time condition, then to an IVR or after-hours answering service depending on the time condition met.

Settings depend on installed modules. You may have more settings than are shown here, or settings may be missing.

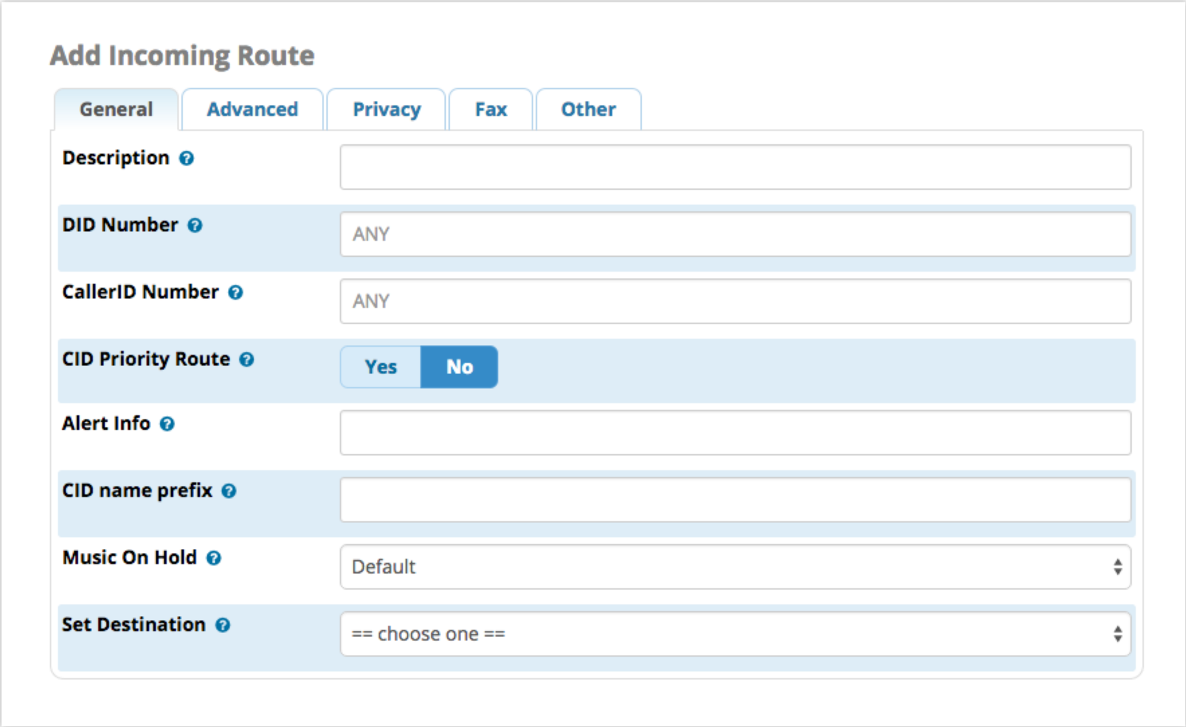
Logging In

- From the top menu click **Connectivity**
- From the drop down click **Inbound Routes**

Adding an Inbound Route

The PBX allows two specific types of inbound routing: DID & CID Routing. These two routing methods can be used on their own or in conjunction with one another. Leaving both fields blank will create a route that matches all calls.

General



Description

Enter a unique description for the route.

DID (Direct Inward Dialing) Number

Routing is based on the trunk on which the call is coming in. In the DID field, you will define the expected "DID Number" if your trunk passes the DID on incoming calls. Leave this blank to match calls with any or no DID info. The DID number entered must match the format of the provider sending the DID. You can also use a pattern match to match a range of numbers. Patterns must begin with an underscore (_) to signify they are patterns. Within patterns, X will match the numbers 0-9 and specific numbers can be matched if they are placed between square parentheses. This field can also be left blank to match calls from all DIDs. This will also match calls that have no DID information.

CID (Caller ID) Number

Routing calls based on the caller ID number of the person that is calling. Define the caller ID number to be matched on incoming calls. Leave this field blank to match any or no CID info. In addition to standard dial sequences, you can also put "Private," "Blocked," "Unknown," "Restricted," "Anonymous" or "Unavailable" in order to catch these special cases if the telco transmits them. Caller ID can be specified as a dial pattern when prefxed with an underscore, so for example to intercept all calls from area code 902, CID can be specified as "_902NXXXXXX" (without the quotes).

CID Priority Route

Yes/No: Whether to designate this route as a Caller ID Priority Route. This will only affect routes that do not have an entry in the DID field. If set to **Yes**, calls with this CID will be routed to this route, even if there is a route to the DID that was called. Normal behavior is for the DID route to take the calls. If there is a specific DID/CID route for this CID, that route will still take the call when that DID is called.

The default priority levels are matched in the following sequence:

With CID Priority Route disabled:

1. Routes with a specific DID and CID will always be first in priority.
2. Routes with a specific DID but no CID will be second in priority.
3. Routes with no DID, but with a specific CID will be third in priority.
4. Routes with no specific DID or CID will be last in priority.

With CID Priority Route enabled:

1. Routes with a specific DID and CID will always be first in priority.
2. Routes with no DID, but with a specific CID will be second in priority.
3. Routes with a specific DID but no CID will be third in priority.
4. Routes with no specific DID or CID will be last in priority.

Alert Info

This is used to send a string of text in the SIP ALERT_INFO headers. It's often used for SIP endpoints that ring differently or auto-answer calls based on the ALERT_INFO text that is received.

CID name prefix

This 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. For example, a number dedicated for sales might be prefixed with ”Sales:.” A call from John Doe would display as, “Sales:John Doe.”

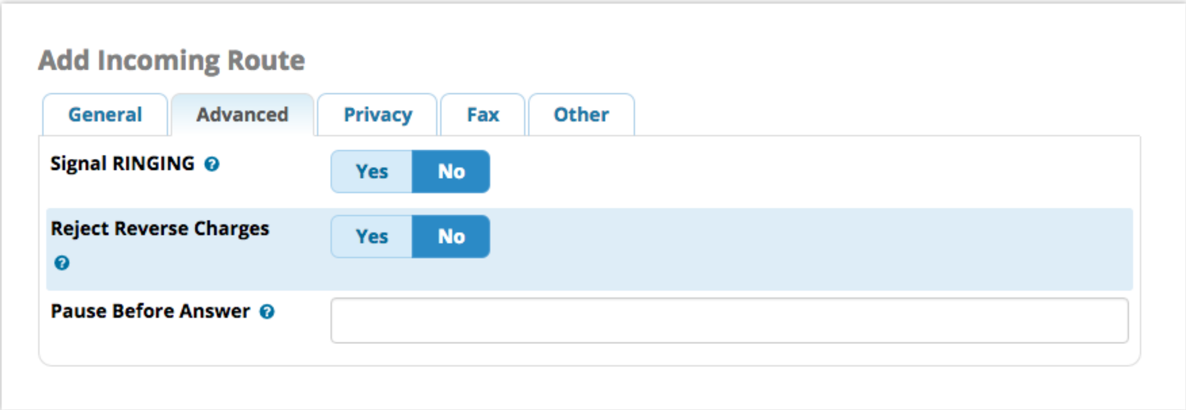
Music On Hold

Music on Hold (MoH) allows you to define the specific music on hold for calls on this inbound route. Whenever a caller is placed on hold, they will hear the music on hold defined here. This is typically used for companies that advertise in their music on hold and take calls in multiple languages. For example, calls to an English DID might play English advertisements while calls to a Spanish DID would play Spanish advertisements.

Set Destination

The PBX provides multiple ways to route a call. This is the place where the desired call target is selected.

Advanced



Signal RINGING

Yes/No: Whether to send “ringing” tones before the system lets the other side know that the call has been answered. Some providers and devices require RINGING to be sent before ANSWER. You’ll notice the need for this if you can send calls directly to a phone/extension, but if you send it to an IVR, it won’t connect the call.

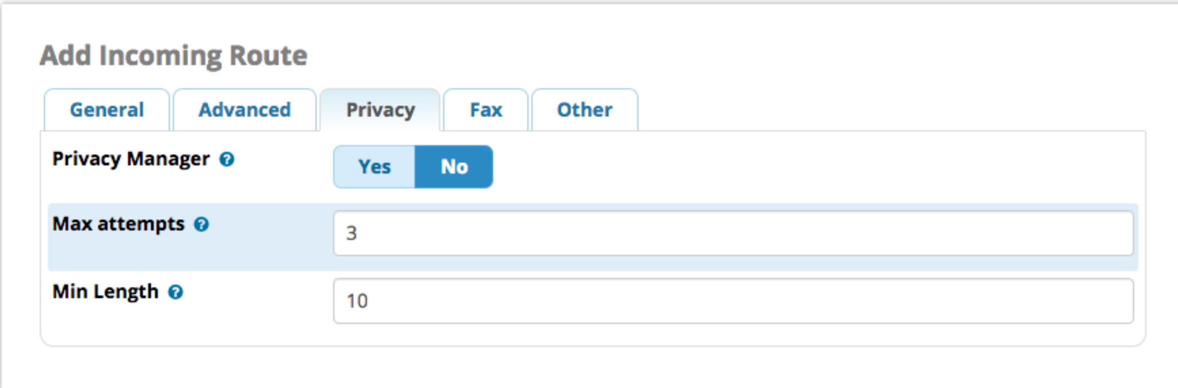
Reject Reverse Charges

Yes/No: Whether to reject calls that indicate a billing reversal, if supported. On PRI channels, the carrier will send a signal if the caller indicates a billing reversal.

Pause Before Answer

An optional delay to have the PBX pause before processing this route. This is not really useful on digital connections, but may be handy if external fax, modem, or security systems are installed on the trunk and you would like them to be able to seize the line prior to the PBX answering the call.

Privacy



Privacy Manager

Yes/No: Whether to enable the PBX "Privacy Manager" functionality on this route. When enabled, calls without an associated caller ID will be prompted to enter their 10-digit telephone number. Callers will have 3 attempts to enter this information before the call is disconnected. If a user/extension has call screening enabled, the incoming caller will be prompted to say their name when the call reaches the user/extension.

Max attempts

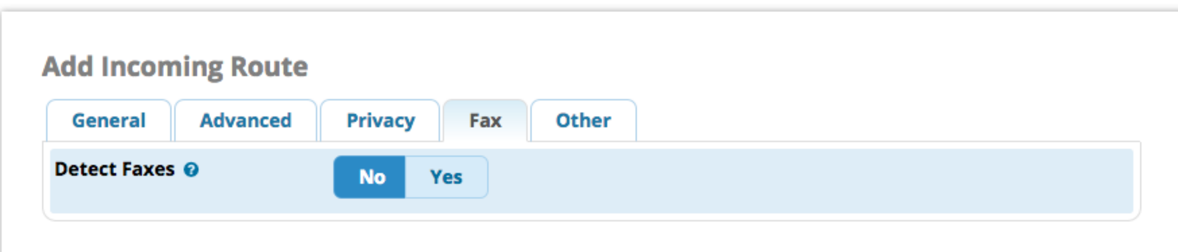
Maximum number of attempts the caller has to enter a valid CallerID.

Min Length

Minimum amount of digits the CallerID needs to contain in order to be considered valid.

Fax

This section only has one option unless you select **Detect Faxes: Yes**.



Detect Faxes

No/Yes: Whether to enable the "fax detect" functionality on this route.

- **No:** No attempts are made to auto-determine the call type. All calls are sent to the defined destination.
- **Yes**: The system will try to auto-determine the type of call. If the call is a fax, it will be routed to the fax destination. Otherwise, it will be routed to the regular destination. Use this option if you receive both voice and fax calls on the same line. (Please note, the best practice is to dedicate routes for your fax services, as "fax detection" is not 100% reliable.)

If **Detect Faxes = Yes**, you will see the following options:

Fax Detection type

Type of fax detection to use.

- Dahdi: Use Dahdi fax detection; requires "faxdetect=" to be set to "incoming" or "both" in Dahdi.conf.
- NVFax: Use NV Fax Detection; Requires NV Fax Detect to be installed and recognized by asterisk.
- SIP: use sip fax detection (t38). Requires asterisk 1.6.2 or greater and 'faxdetect=yes' in the sip config files.

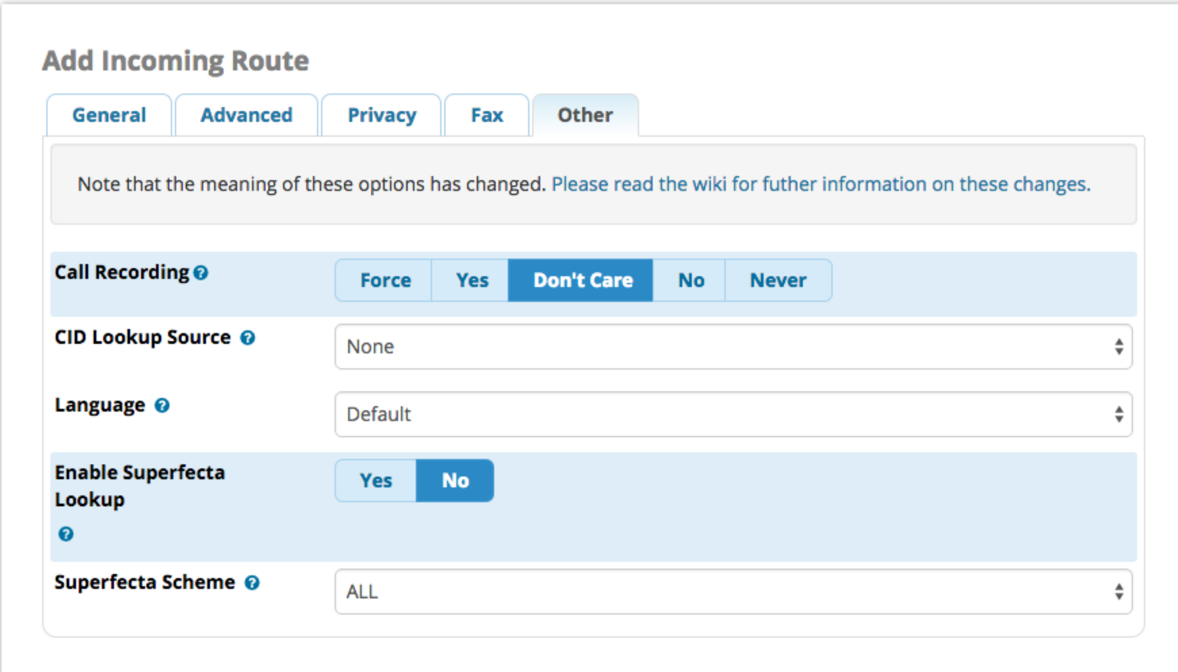
Fax Detection Time

How long to wait and try to detect fax. Please note that callers to a Dahdi channel will hear ringing for this amount of time (i.e. the system wont "answer" the call, it will just play ringing).

Fax Destination

Where to send the faxes.

Other



Call Recording

Force/Yes/Don't Care/No/Never: This setting controls or overrides the call recording behavior for calls using this route.

CID Lookup Source

A CID lookup source resolves numeric caller IDs of incoming calls. This gives you more detailed caller ID information and CDR reports. The sources are defined in the Caller ID Lookup Sources module and can be linked to web-based services, local databases, or CRM systems. Lookup sources are also useful if your trunks do not pass the caller ID name information with the number.

Language

This allows the language setting to be configured before the call reaches a destination. This is useful when you use privacy manager and want to have the prompts played in the proper language. Please note that not all of the voice prompts are recorded in every language. If the prompt is not available, it will play in the default English setting. If this is left blank, the system will default to English.

The available language codes are:

- **English** - en
- **Chinese** - cn
- **German** - de
- **Spanish** - es

- **French** - fr
- **Hebrew** - he
- **Hungarian** - hu
- **Italian** - it
- **Portuguese** - pt
- **Portuguese** (Brazil) - bp
- **Russian** - ru
- **Swedish** - sv

Enable Superfecta Lookup

Yes/No: Whether to use Caller ID Superfecta to look up caller ID. Sources can be added/removed in CID Superfecta module.

Superfecta Scheme

The Caller ID Superfecta scheme to use. Sources can be added/removed in CID Superfecta module.

4.4 Outbound Routes

Outbound routing is a set of rules that the PBX uses to decide which trunk to use for an outbound call. Having multiple trunks allows you to control cost by routing calls over the least costly trunk for a particular call. Outbound routes are used to specify what numbers are allowed to go out a particular route.

You will want to make sure you define routes for all types of calls. Not defining a route can leave your users frustrated when they need to make an important call.

When a call is placed, the actual number dialed by the user is compared with the dial patterns in each route (from highest to lowest priority) until a match is found. If no match if found, the call fails. If the number dialed matches a pattern in more than one route, only the rules with the highest priority in the route are used.

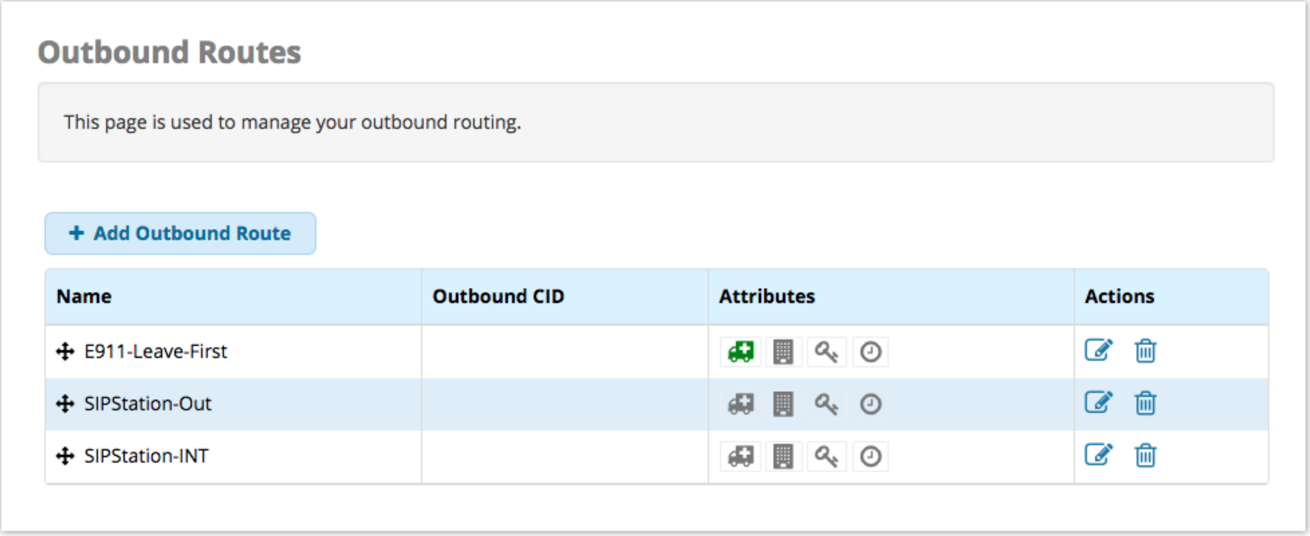
Important
The emergency route should normally be placed first, at the top of the list.

"Outbound Route Dial Patterns" can be used to strip off leading digits before passing them to a trunk. This is most useful if you use a specific dialing code to access a particular route. For example, "9" to access an outside line.

Outbound dial rules work in conjunction with trunk dial rules. Trunk dial rules are ONLY used for adding numbers to, or subtracting numbers from the number being sent to the trunk. Trunk dial rules are never used to allow or restrict numbers that may be dialed.

Logging In

- From the top menu click **Connectivity**
- In the drop down click **Outbound Routes**



The outbound routes home page shows a list of routes, in order of priority from highest to lowest. The columns are **Name, Outbound CID, Attributes,** and **Actions**.

Name

The name of the route, along with an arrow symbol ✥ indicating you can drag and drop the route to change its order in the list.

Outbound CID

The outbound caller ID for this route.

Attributes

- **Green = Yes**
- **Gray = No**

= Emergency Route

= Intra-Company Route

= Password-Protected

= Time Group Assigned

Actions

= View or Edit

= Delete

Adding an Outbound Route

Click the **Add Outbound Route** button.

Route Settings Tab

Route Name

Name of this route. Usually used to describe what type of calls this route matches (for example, "local" or "longdistance"). Cannot contain spaces.

Route CID

Optional route Caller ID to be used for this route. If set, this will override all CIDs specified *except*:

- Extension/device EMERGENCY CIDs if this route is checked as an EMERGENCY route type
- Trunk CID if trunk is set to force its CID
- Forwarded call CIDs (CF, Follow Me, Ring Groups, etc)
- Extension/user CIDs if the Override Extension option is set to No

Override Extension

Yes/No: If set to **Yes**, the extension's Outbound CID will be ignored in favor of the route CID set above. The extension's Emergency CID will still be used if the route is an Emergency Route and the Extension has a defined Emergency CID.

Route Password

(Optional) A route can prompt users for a password before allowing calls to progress. This is useful for restricting calls to international destinations or 1-900 numbers. A numerical password or the path to an authenticate password file can be used. Leave this field blank to not prompt for a password.

Route Type

(Optional) Whether the route is considered an emergency or intra-company route.

- **Emergency**: This will enforce the use of a device's Emergency CID setting (if set). Select this option if the route is used for emergency dialing (i.e.: 911).
- **Intra-Company**: The system will treat route as an intra-company connection, preserving the internal caller ID information instead of the outbound CID of either the extension or trunk.

Music On Hold

You can choose which music category (MoH) to use. For example, choose a type appropriate for a destination country that may have announcements in the appropriate language.

Time Group

If this route should only be available during certain times, then select a time group created under the Time Groups module. The route will be ignored outside of times specified in that time group. If left as default, "Permanent Route," then it will always be available.

Route Position

Where to insert this route or relocate it relative to the other routes.

Trunk Sequence for Matched Routes

The trunk sequence controls the order of trunks that will be used when the above dial patterns are matched. For dial patterns that match long distance numbers, for example, you would want to pick the lowest cost route for long distance, followed by more expensive routes.

Time Group

By default, the route is a **Permanent Route**, meaning it is available at all times. To restrict the route to only being available during certain times, you can select a time group from the drop-down menu. Then, the route would be ignored outside of times specified in the time group.

Route Position

Where to insert this route or relocate it relative to the other routes. You can select a position from the drop-down menu. You will also be able to move the route later by dragging and dropping it in the routes list on the module home page.

Trunk Sequence for Matched Routes

Controls the order of trunks that will be used when the above dial patterns are matched. For dial patterns that match long distance numbers, for example, you'd want to pick the cheapest routes for long distance (i.e., VoIP trunks first) followed by more expensive routes (POTS lines).

Select one or more trunks from the drop-down menus. You can also change the order of trunks by dragging and dropping the routes using the arrow icon . The top route will be tried first, followed by the next route down, and so forth.

Optional Destination on Congestion

Destination for calls that encounter trunk congestion. Default = **Normal Congestion**. You can select a different destination if desired. For example, you might play a customized system recording.

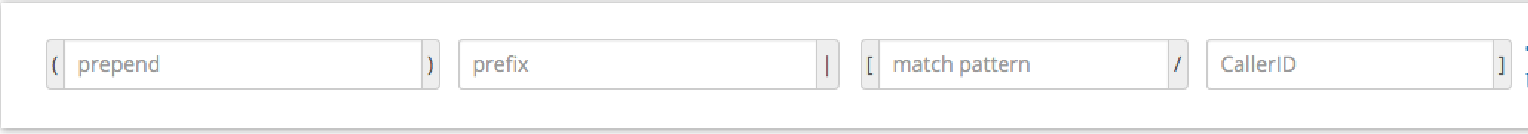
Dial Patterns Tab

A dial pattern is a unique set of digits that will select this route and send the call to the designated trunks. If a dialed pattern matches this route, no subsequent routes will be tried. If time groups are enabled, subsequent routes will be checked for matches outside of the designated times.

A dial pattern can have up to four elements: **Prepend**, **Prefix**, **Match Pattern**, and **CallerID**. Each element has its own field in the Outbound Routes Dial Patterns tab.

The format is:

(prepend) prefix | [match pattern / caller ID]



You can enter any combination of numbers and the following special patterns:

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, example [123] is 1 OR 2 OR 3. Note that multiple numbers can be separated 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.
. (dot)	It matches one or more characters and (acts as a wildcard)

Prepend

The prepend will be added to the beginning of a successful match. If the dialed number matches the patterns specified by the subsequent columns, then this will be prepended to the sequence before sending it to the trunks.

Prefix

Prefix to remove upon a successful match. The dialed number is compared to this and the subsequent columns for a match (prefix + match pattern). Upon a match, this prefix is removed (stripped) from the dialed number before sending the sequence to the trunks.

Match Pattern

The dialed number will be compared against the prefix + this match pattern. Upon a match, the match pattern portion of the dialed number will be sent to the trunks.

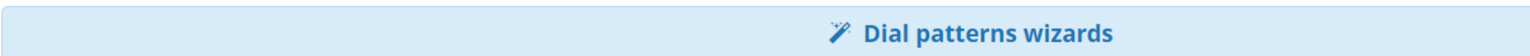
CallerID

If caller ID is supplied, the dialed number will only match the prefix + match pattern if the caller ID being transmitted matches this. When extensions make outbound calls, the caller ID will be their extension number and NOT their outbound CID. The above special matching sequences can used for caller ID matching similar to other number matches.

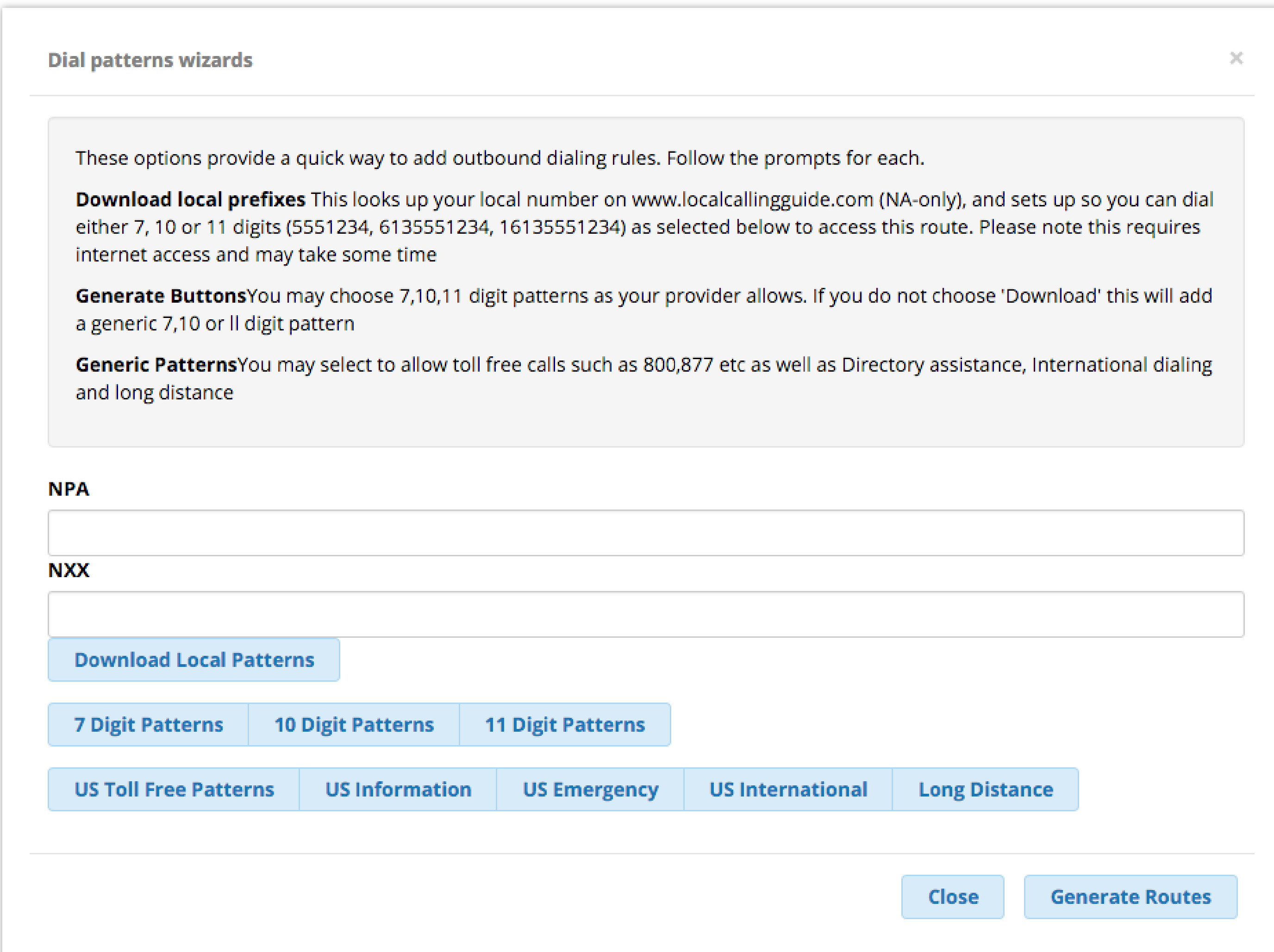
Dial Patterns Wizards

These are pre-constructed dial patterns. Selecting a pre-made pattern will automatically populate the Dial Pattern fields.

To use a wizard, click the **Dial patterns wizard's** button.



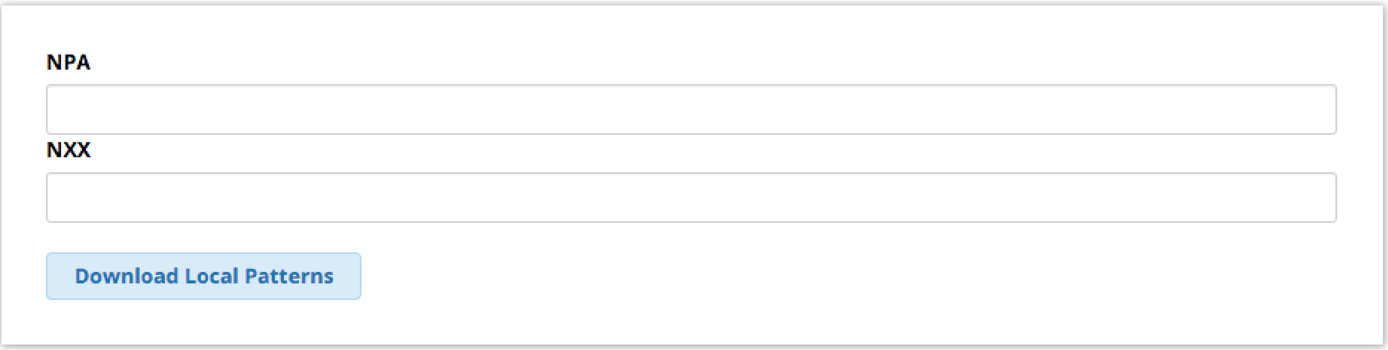
This displays a pop-up window where you can generate various dial patterns.



The information on this wizard comes from a variety of sources and is not guaranteed to be 100% complete or correct. For authoritative information, please consult the appropriate company or trunk provider.

How to Generate Local Dial Patterns

The Download Local Patterns feature will look up the NPA-NXX (area code and prefix) on www.localcallingguide.com. This feature is only available for North American numbers. Internet access is required in order to use this feature.



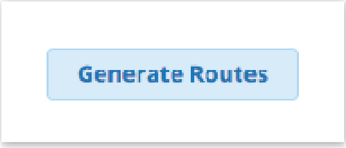
-
- Enter your local **NPA** (area code)
- Enter your local **NXX** (prefix)
- Select the **Download Local Patterns** button. It will turn dark blue when selected.

- Now, select one or more pattern options in the next line of buttons. You can choose from 7-, 10-, and 11-digit patterns.

Unselected:

Selected:

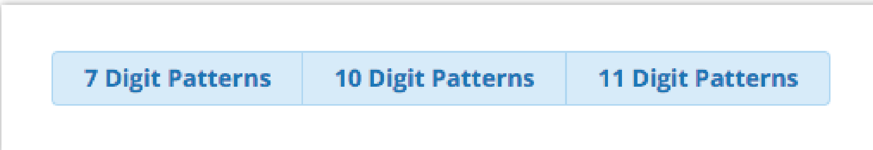
- Click the **Generate Routes** button.



- Be patient; this may take some time. After the system has looked up and downloaded local dial patterns, the Wizard window will disappear, and your dial pattern fields will be populated with the 7-, 10-, and/or 11-digit patterns you requested.

How to Generate Generic 7-, 10-, and/or 11-Digit Dial Patterns

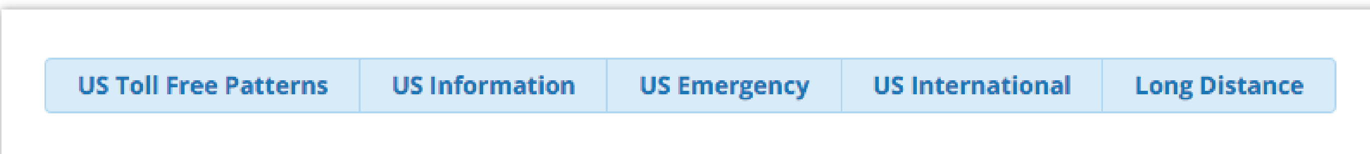
- Leave the **NPA** and **NXX** fields blank
- **Do not** select the **Download Local Patterns** button (it should be light blue in color)
- Then, select one or more pattern options:



- Click the **Generate Routes** button.
- Your new dial patterns will be added to your list in the Dial Patterns tab. If the Wizard window does not automatically disappear, click **Close** to close it.

How to Generate Toll-Free, US Information, US Emergency, US International, and Long Distance Dial Patterns

- Choose one or more of the buttons near the bottom of the Dial Patterns Wizard. Note: the settings above, such as 7/10/11 digit patterns and NPA-NXX do not affect these dial patterns.



- **US Toll Free Patterns:** 11-digit dial patterns 1800NXXXXXX, 1888NXXXXXX, 1877NXXXXXX, 1866NXXXXXX, 1855NXXXXXX, and 1844NXXXXXX
- **US Information:** 3-digit dial patterns 211, 311, 411, 511, 611, and 711.
- **US Emergency:** 3-digit dial patterns 911 and 933, along with three other versions of 911 containing a prefix (1-911, 9-911, and 91-911).
- **US International:** Matches any number that begins with 011 (dial pattern of "011 + wildcard").
- **US International:** A generic 11-digit dial pattern starting with 1 ("1NXXNXXXXXX").

- Click the **Generate Routes** button.
- Your new dial patterns will be added to your list in the Dial Patterns tab. If the Wizard window does not automatically disappear, click **Close** to close it.

Import/Export Patterns Tab

Here, you can import dial pattern CSV files or export your dialplan as a CSV file.

Upload

Create a CSV file with a dial pattern list. If there are no headers, then your CSV file must have 4 columns of patterns in the same order as in the GUI. You can also supply headers: **prepend**, **prefix**, **match pattern** and **callerid** in the first row. If there are less than 4 recognized headers, then the remaining columns will be blank.

Click the **Choose File** button to import a CSV file. Select the file from your computer. After you have made your selection, the filename will appear next to the Choose File button. The new dial patterns are added to your list in the Dial Patterns tab after you click the module's **Submit** button.

After clicking **Submit**, the dial patterns in your CSV file will replace the entire list in your Dial Patterns tab, instead of adding to or syncing with any dial patterns you previously entered in that tab.

Export

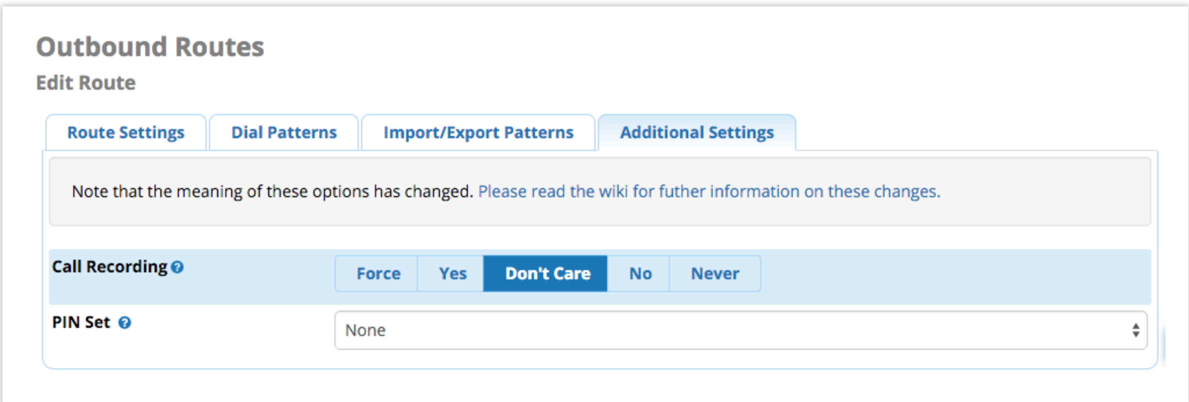
Click the **Export** button to download a list of patterns as a CSV file with headers listed as: **prepend**, **prefix**, **match pattern** and **callerid** in the first row.

This feature will export the latest dialplan that has been submitted. If this is a brand-new route or you have just made changes to your dial patterns, you would need to click the module's **Submit** button before this feature will work correctly.

Additional Settings

The settings shown here will vary depending upon whether you have additional add-ons installed. If you have modules such as Outbound Call Limiting, Class of Service, Extension Routing, Fax Pro, and Page Pro, you will see their associated options.

Below is the view without add-ons:



Call Recording

Force/Yes/Don't Care/No/Never: This sets the call recording behavior for calls going out this route.

PIN Set

Select a PIN set to use. Default = **None.** For more information on this feature,

Save

Make sure to press the **Submit** button when done editing your outbound route, followed by the **Apply Config** button to apply the changes. You can also create a duplicate route by clicking the **Duplicate Route** button.

Changing the Order of Outbound Routes

Remember, the system searches for a matching dial pattern by starting with the top route and working its way down. If a match is found, the system does not continue going down the list looking for a "better" route. Therefore, route order is important, especially if there is some overlap. For example, the number 5555551212 will match both a dial pattern of 555555XXXX and NXXNXXXXXX.

To change the order of routes in the module home page:

- Simply drag and drop. Hold the mouse button down over the arrow symbol ✥ or the route's name, and drag the entire line to a new position in the list.
- A pop-up notification will appear to let you know you've changed the order.
- Click the **Apply Config** button to apply the changes.

Editing or Deleting an Outbound Route

- To **Edit**, click the edit button next to an outbound route in the list on the module home page. When finished, click the **Submit** button, then click the **Apply Config** button.
- To **Delete**, click the trash button next to an outbound route in the list on the module home page. Confirm deletion by clicking **OK** in the pop-up window. Then click the **Apply Config** button.
- Alternatively, if already viewing an outbound route, click the **Delete** button, click **OK** in the pop-up window, and click the **Apply Config** button.

4.5 Trunks

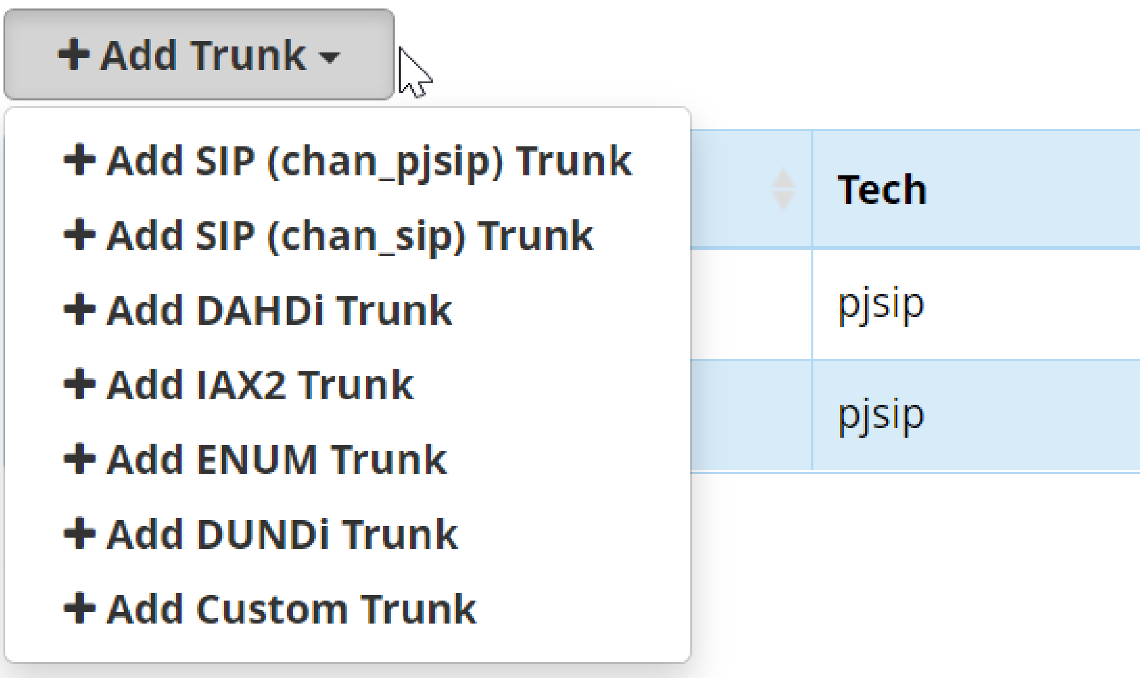
The Trunks module is where you control connectivity to the PSTN and your VoIP provider(s). This is where you also control to interconnect other PBX's for multi-site applications. The most common trunks are SIP Other than the Extensions module, the Trunks module is one of the most critical modules on the system and allows for a great deal of flexibility.

Logging in

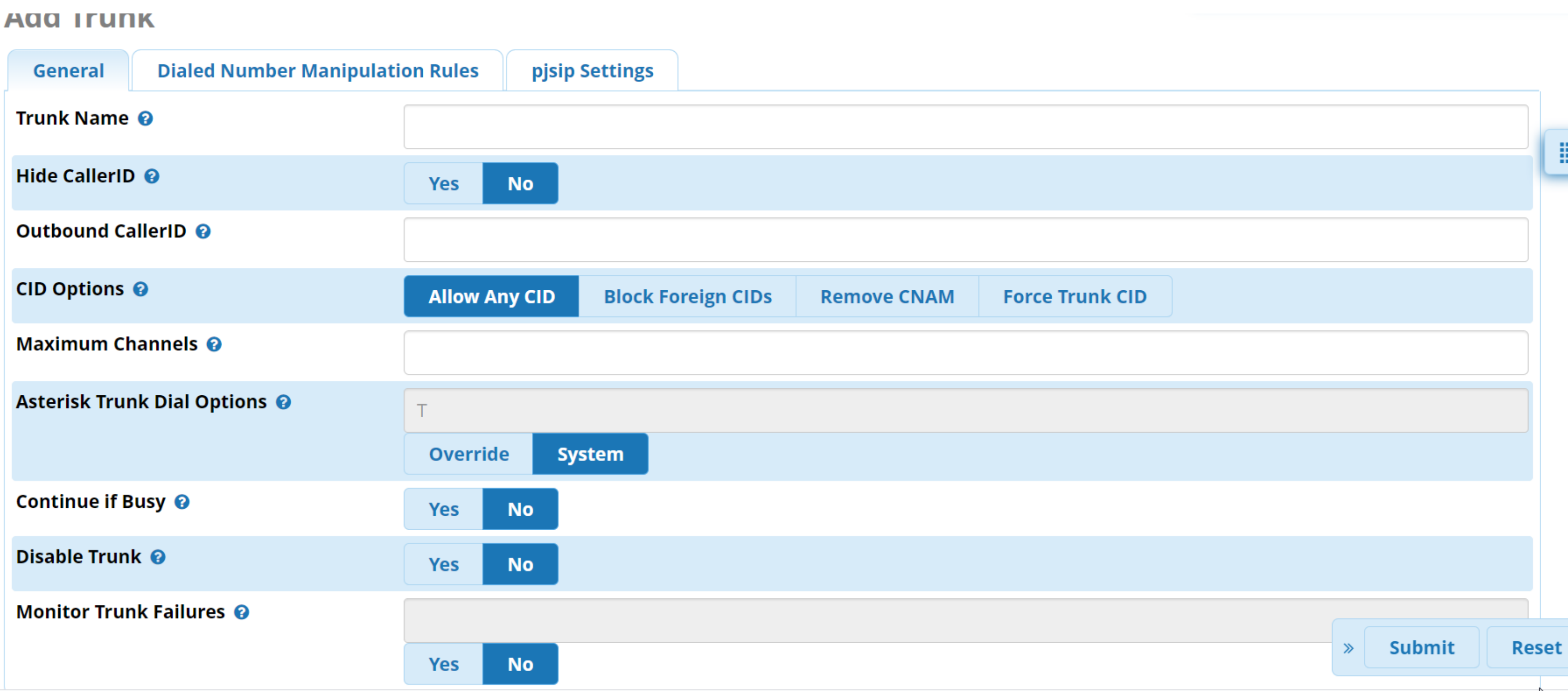
From the top menu click **Connectivity**
In the drop down click **Trunks**

Adding a Trunk

You will want to click on the trunk type you wish to create. Please note in this guide we will cover the generic settings that are universal to all trunk types first. Following this we will cover technology specific settings.



General Settings



Trunk Name

Set a descriptive name for the trunk.

Outbound CallerID

Use this field to specify caller ID for calls placed out of this trunk with the <NXXNXXXXXX> format. You can also use the format: “hidden”
<NXXNXXXXXX> to hide the caller ID sent out over digital lines, if supported (E1/T1/J1/BRI/SIP/IAX2).

CID Options

This setting determines what CIDs will be allowed out of this trunk. Please NOTE that Emergency CIDs defined on an extension or device will ALWAYS be used if this trunk is part of an emergency route regardless of these settings.

Allow Any CID

All CIDs, including foreign CIDs from forwarded external calls, will be transmitted.

Block Foreign CIDs

This will block any CID that is the result of a forwarded call from off the system. CIDs that are defined for an extension or device will be transmitted.

Remove CNAM

This will remove the CNAM (Name) from any CID sent out of this trunk.

Force Trunk CID

This will always use the CID defined for the trunk, except if the trunk is part of an emergency route with an emergency CID defined for the extension or device. Intra-Company routes will always transmit an extension's internal number and name.

Maximum Channels

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. To count inbound calls against this maximum, use the auto-generated context: from-trunk-[trunkname] as the inbound trunk's context (see extensions_additional.conf). Leave blank to specify no maximum.

Continue if Busy

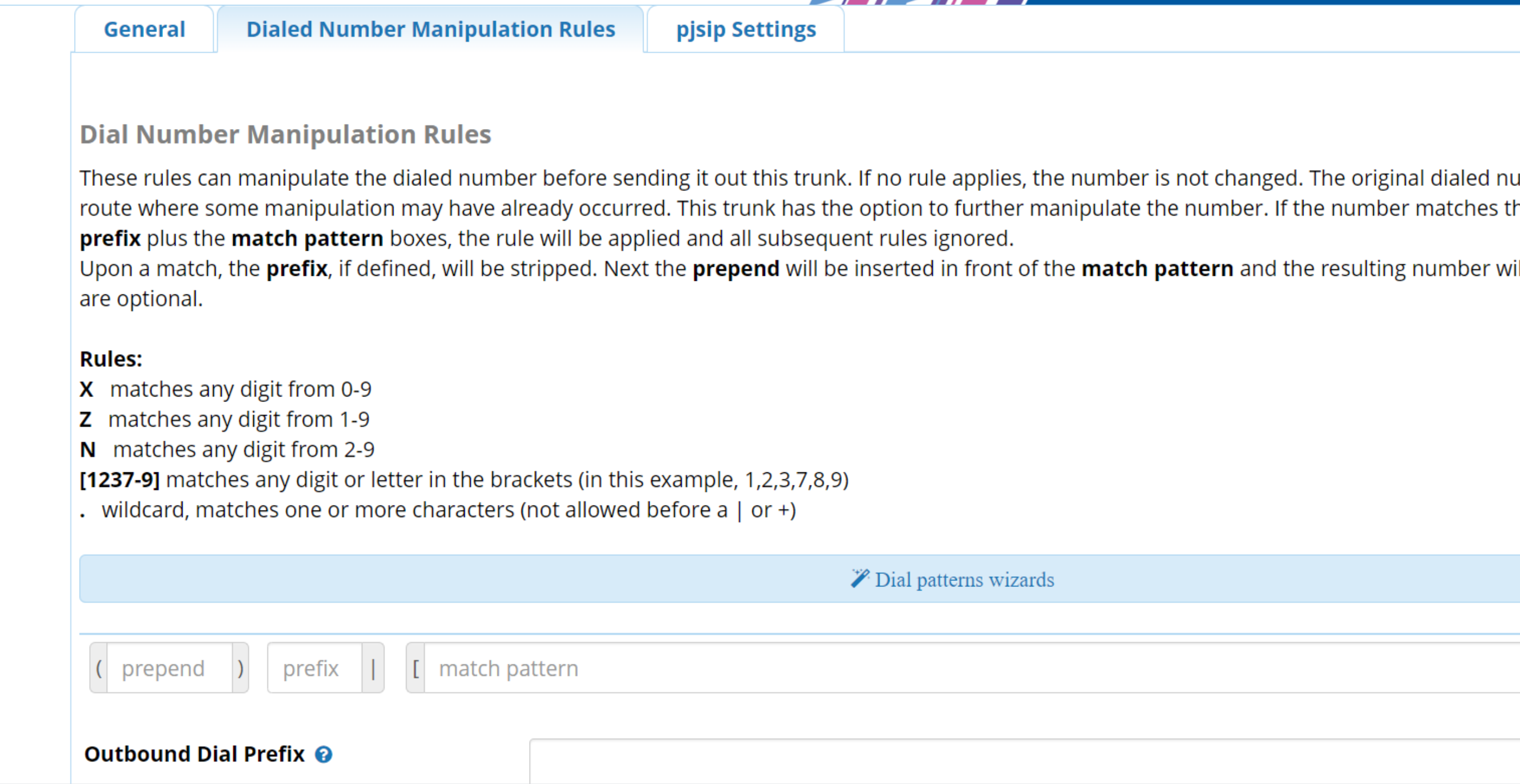
Normally the next trunk is only tried upon a trunk being 'Congested' in some form, or unavailable. Checking this box will force a failed call to always continue to the next configured trunk or destination even when the channel reports BUSY or INVALID NUMBER. This should normally be unchecked

Disable Trunk

Check this to disable this trunk in all routes where it is used.

Dial Pattern Manipulation Rules

These rules can manipulate the dialled number before sending it out of this trunk. If no rule applies, the number is not changed. The original dialled number is passed down from the route where some manipulation may have already occurred. This trunk has the option to further manipulate the number. If the number matches the combined values in the prefix plus the match pattern boxes, the rule will be applied and all subsequent rules ignored. Upon a match, the prefix, if defined, will be stripped. Next, the prepend will be inserted in front of the match pattern and the resulting number will be sent to the trunk. All fields are optional.



Pattern	Description
X	Matches any digit from 0-9
Z	Matches any digit from 1-9
N	Matches any digit from 2-9
[1237-9]	Matches any digit in the brackets (example: 1,2,3,7,8,9)
. (dot)	Wildcard, matches one or more dialled digits.

Prepend

Digits to prepend upon a successful match. If the dialled number matches the patterns in the prefix and match pattern boxes, this will be prepended before sending to the trunk

Prefix

Prefix to remove upon a successful match. If the dialled number matches this, plus the match pattern box, this prefix is removed before adding the optional prepend box and sending the results to the trunk.

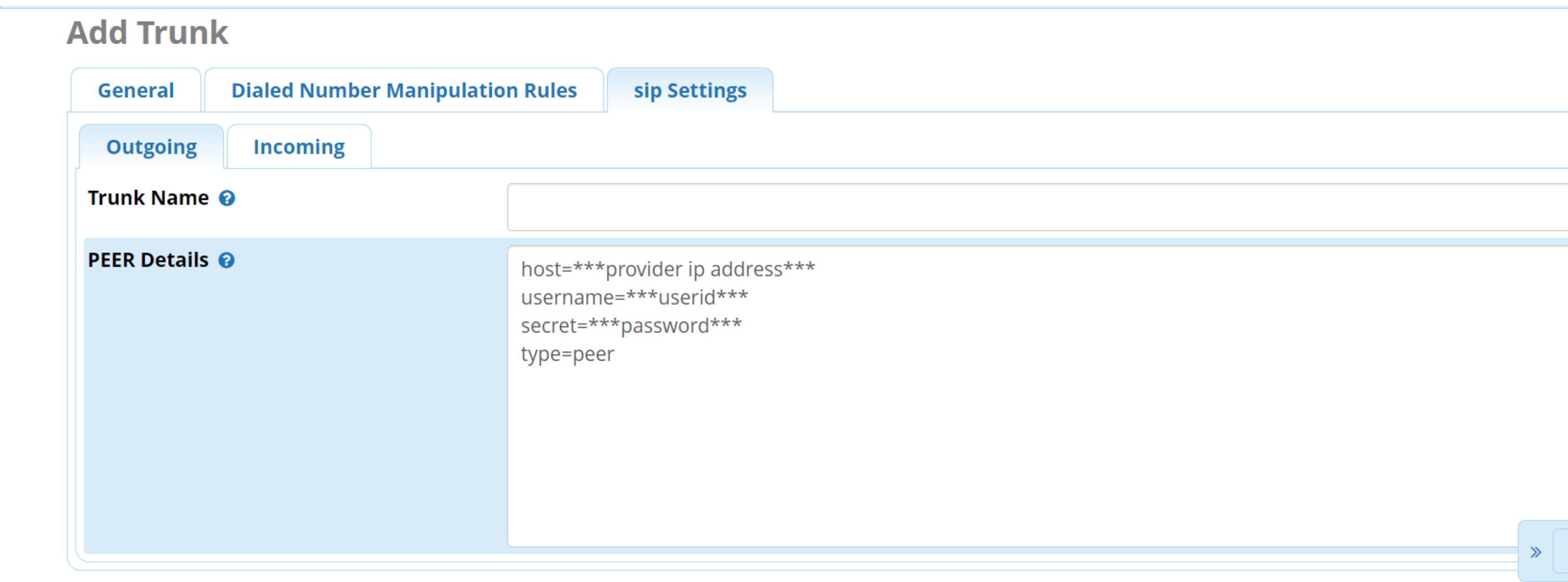
Match pattern

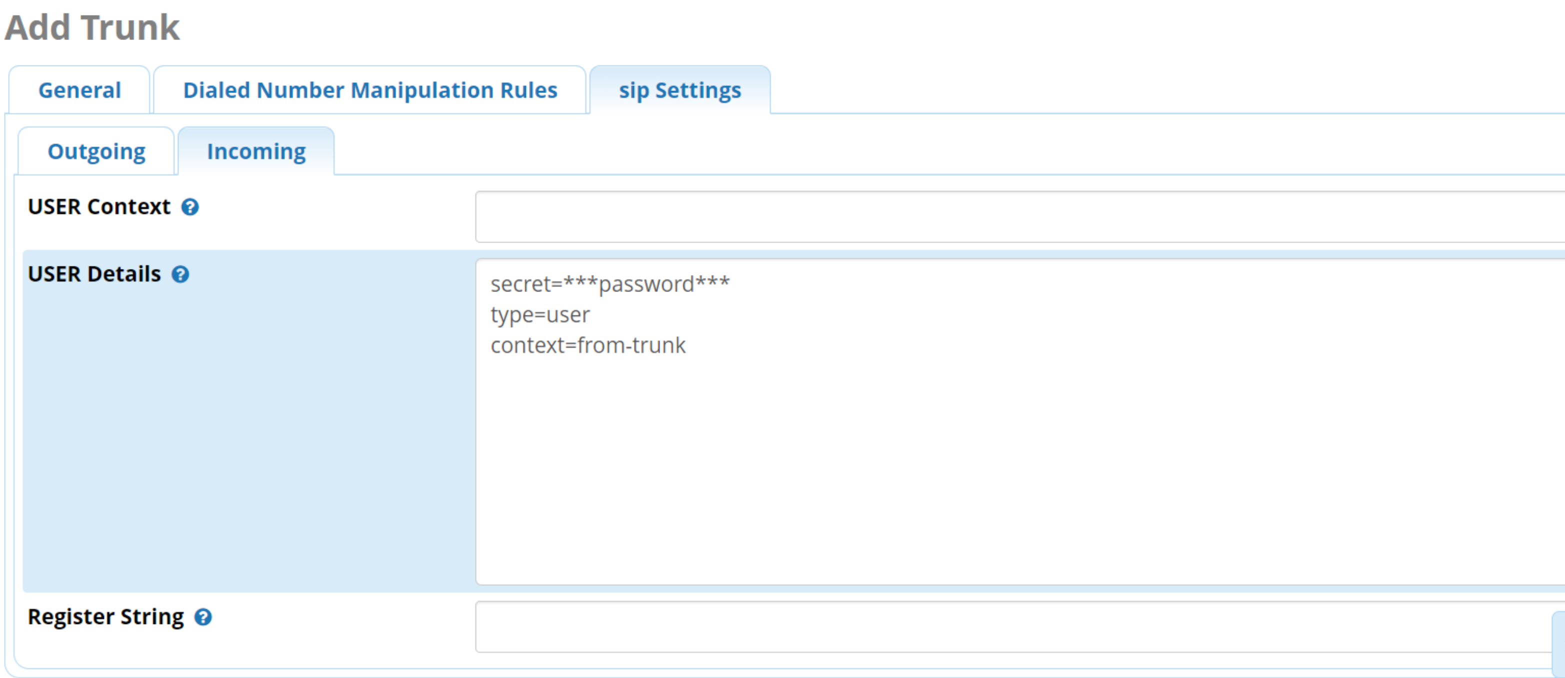
The dialled number will be compared against the prefix, plus this pattern. Upon a match, this portion of the number will be sent to the trunks after removing the prefix and appending the prepend digits. You can completely replace a number by matching on the prefix only, replacing it with a prepend and leaving the match pattern blank.

Outbound Dial Prefix

The outbound dialling prefix is used to prefix a dialling string to all outbound calls placed on this trunk. For example, if this trunk is behind another PBX or is a Centrex line, then you would put “9” here to access an outbound line. Another common use is to prefix calls with “w” (to add a 500ms wait per w) on a POTS line that needs time to obtain a dial tone to avoid eating digits. Most users should leave this option blank.

SIP (chan_sip)/IAX2 Specific Settings





Trunk Name

Give the trunk a descriptive name such as “mysiptrunk.”

PEER Details

Here you give the PEER connection parameters supplied by your VoIP provider. You may need to add to the provided default settings and in some cases remove default settings depending on your provider or application.

Order is important, as it will be retained. For example, if you use an “allow/deny” directive, then make sure the “deny” is first (reading top down).

USER Context

This is most often the account name or number your provider expects.

USER Details

Here you supply the USER connection parameters supplied by your VoIP provider. You may need to add to the provided default settings and in some cases remove default settings depending on your provider or application.

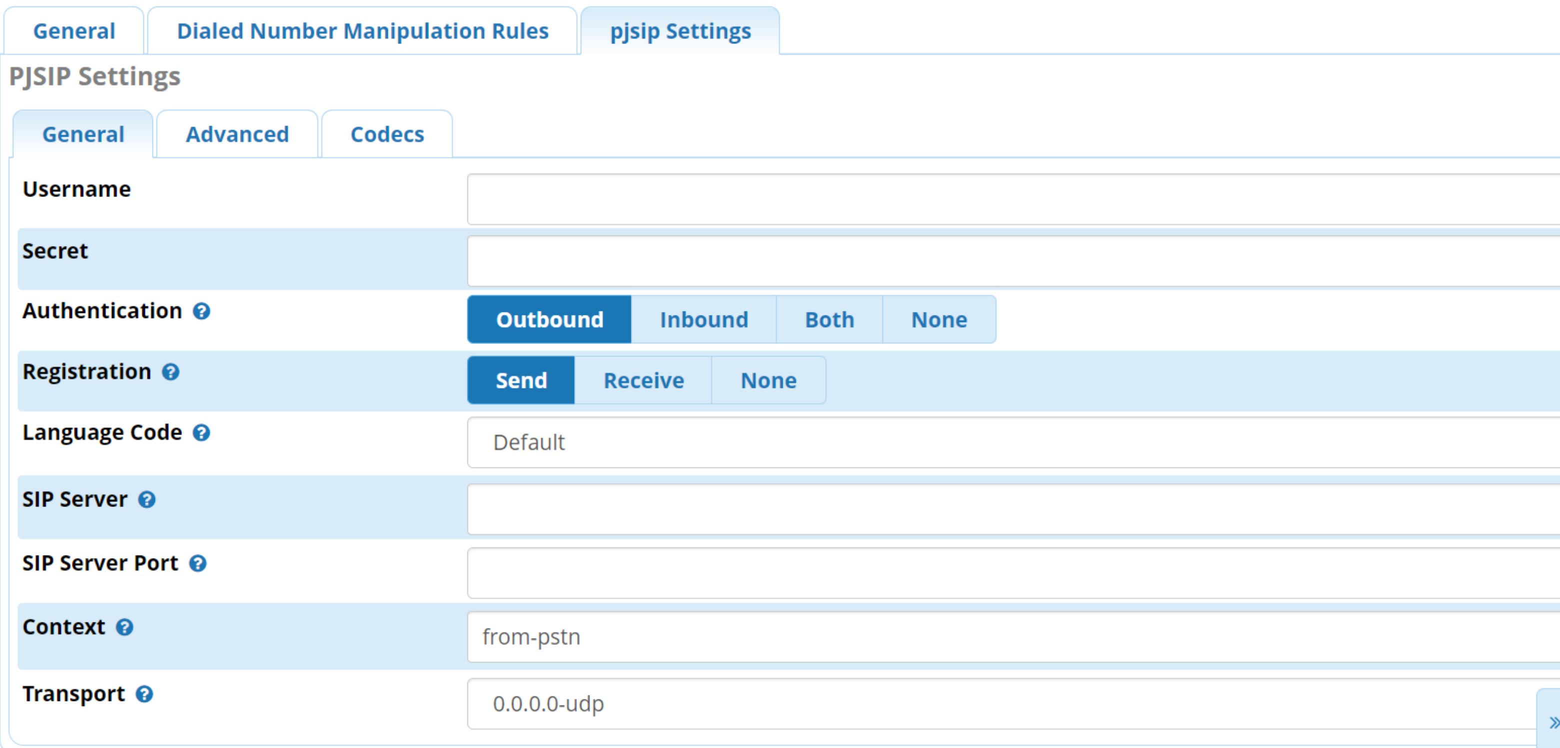
Order is important, as it will be retained. For example, if you use an “allow/deny” directive, then make sure the “deny” is first (reading top down).

Register String

Most VoIP providers require your system to register with theirs. If required, you will need to enter the string the provider specifies, such as username:password@some.voipprovider.com. In some cases, you may need to provide a DID and the end of the string:

such as **username:password@some.voipprovider.com/7045551212.**

SIP TRUNK(PJSIP)



Username

SIP Username

Secret

SIP Password

Authentication

Usually, this will be set to 'Outbound', which authenticates calls going out, and allows unauthenticated calls in from the other server. If you select 'None', all calls from or to the specified SIP Server are unauthenticated. Setting this to 'None' may be insecure!

Registration

You normally Send registration, which tells the remote server where to send your calls. If the other server is not on a fixed address, it will need to register to this server (Receive), so this server can send calls to it. You would select None if both machines have a fixed address and do not require registration.
Warning: If you select 'None', registration attempts for the Username and Secret specified above will be rejected. Setting this incorrectly may result in firewall services detecting this as an attack and blocking the machine trying to register. Do not change this unless you control both servers, and are sure it is required!

SIP Server

SIP Server you register to

SIP Server Port

SIP Port

Context

Context to send the Inbound Call to.

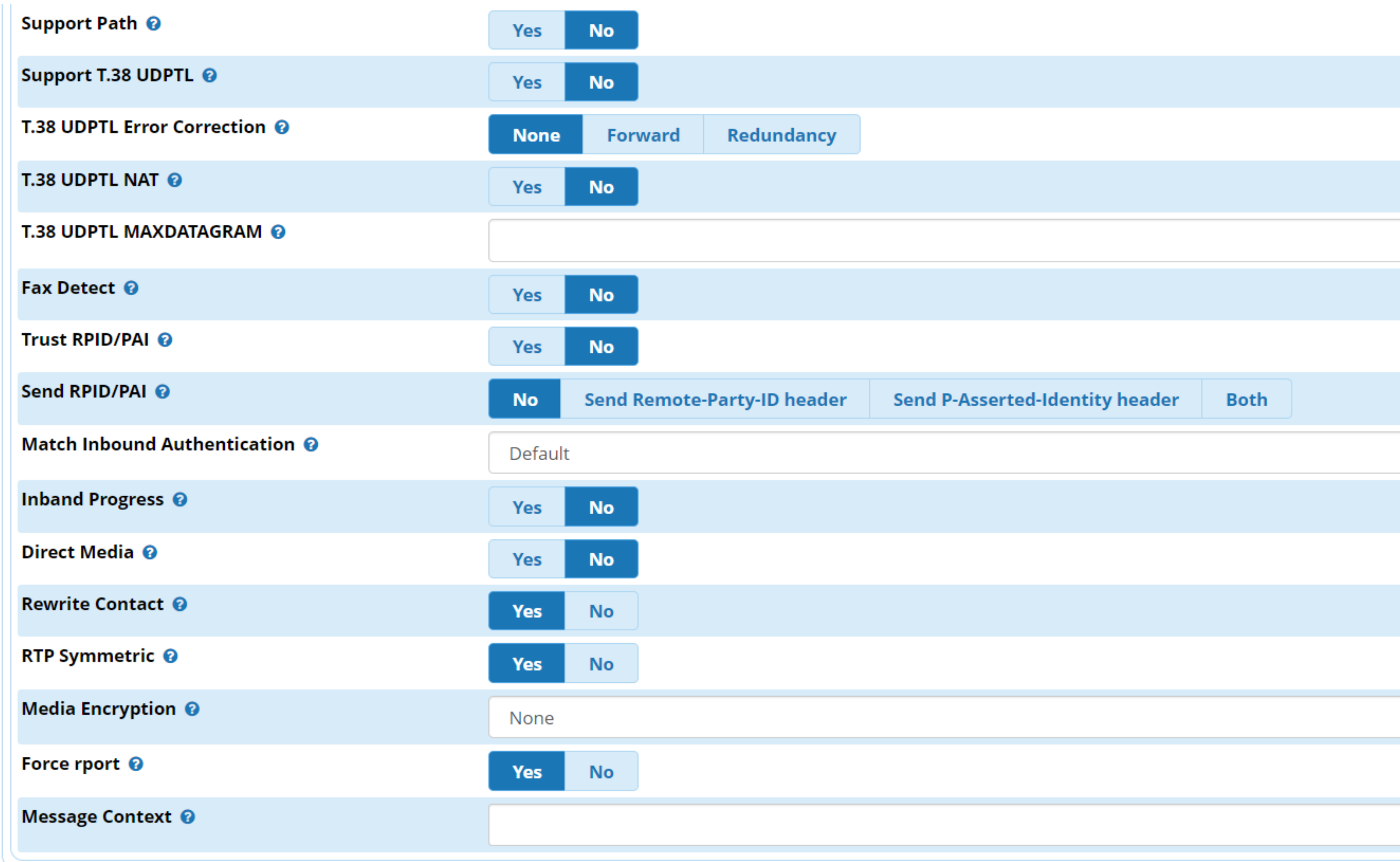
Transport

The Transport to use for connection

PJSIP Settings

General | Advanced | Codecs

DTMF Mode	Auto
Permanent Auth Rejection	Yes / **No**
Forbidden Retry Interval	30
Fatal Retry Interval	30
General Retry Interval	60
Expiration	3600
Max Retries	10000
Qualify Frequency	60
Outbound Proxy	
Contact User	
From Domain	
From User	
Client URI	
Server URI	
Media Address	
AOR	
AOR Contact	
Match (Permit)	



DTMF

The DTMF signaling mode used by this trunk, usually RFC for most trunks

- Auto [Asterisk 13] - DTMF is sent as RFC 4733 if the other side supports it or as INBAND if not.
- rfc4733 - DTMF is sent out of band of the main audio stream. This supersedes the older RFC-2833 used within the older chan_sip.
- inband - DTMF is sent as part of audio stream.
- info - DTMF is sent as SIP INFO packets.

Permanent Auth Rejection

Determines whether failed authentication challenges are treated as permanent failures.

Forbidden Retry Interval

How long to wait before retry when receiving a 403 Forbidden response.

Fatal Retry Interval

How long to wait before retry when receiving a fatal response. If 'Forbidden Retry Interval' is also set then 'Forbidden Retry Interval' takes precedence over this one when a 403 is received. Also, if 'Permanent Auth Rejection' is enabled then a 401 and 407 become subject to this retry interval.

Expiration

Expiration time for registrations in seconds.

Max Retries

Maximum number of registration attempts.

Qualify Frequency

Interval at which to qualify.

Outbound Proxy

SIP Proxy

Contact User

Contact User to use in request.

From Domain

Domain to use in From header for requests to this trunk

From User

Username to use in From header for requests to this trunk

Client URI

Client SIP URI used when attempting outbound registratio

Server URI

SIP URI of the server to register against

Media Address

This address will be provided to clients. If blank, will use the default settings

AOR

AOR to use in trunk. This setting is automatically generated by the PBX if left blank
trunk_name

AOR Contact

Permanent contacts assigned to AoR

Match (Permit)

IP addresses or networks to match against. The value is a comma-delimited list of IP addresses. IP addresses may have a subnet mask appended. The subnet mask may be written in either CIDR or dot-decimal notation. Separate the IP address and subnet mask with a slash ('/'). This setting is automatically generated by the PBX if left blank

Support Path

When this option is enabled, outbound REGISTER requests will advertise support for Path headers so that intervening proxies can add to the Path header as necessary.

Support T.38 UDPTL

Whether T.38 UDPTL support is enabled or not

T.38 UDPTL Error Correction

T.38 UDPTL error correction method

T.38 UDPTL NAT

Whether NAT support is enabled on UDPTL sessions

T.38 UDPTL MAXDATAGRAM

T.38 UDPTL maximum datagram size.

Fax Detect

This option can be set to send the session to the fax extension when a CNG tone is detected.

Trust RPID/PAI

Trust the P-Asserted-Identity and/or Remote-Party-ID header

Send RPID/PAI

Send the P-Asserted-Identity and/or Remote-Party-ID header

Match Inbound Authentication

Matches the endpoint based on the selected options.

Inband Progress

Determines whether chan_pjsip will indicate ringing using inband progress.

Direct Media

Determines whether media may flow directly between endpoints.

Rewrite Contact

Allow Contact header to be rewritten with the source IP address-port

RTP Symmetric

Enforce that RTP must be symmetric. This should almost always be on.

Media Encryption

Determines whether res_pjsip will use and enforce usage of media encryption for this endpoint. [media_encryption]

Force rport

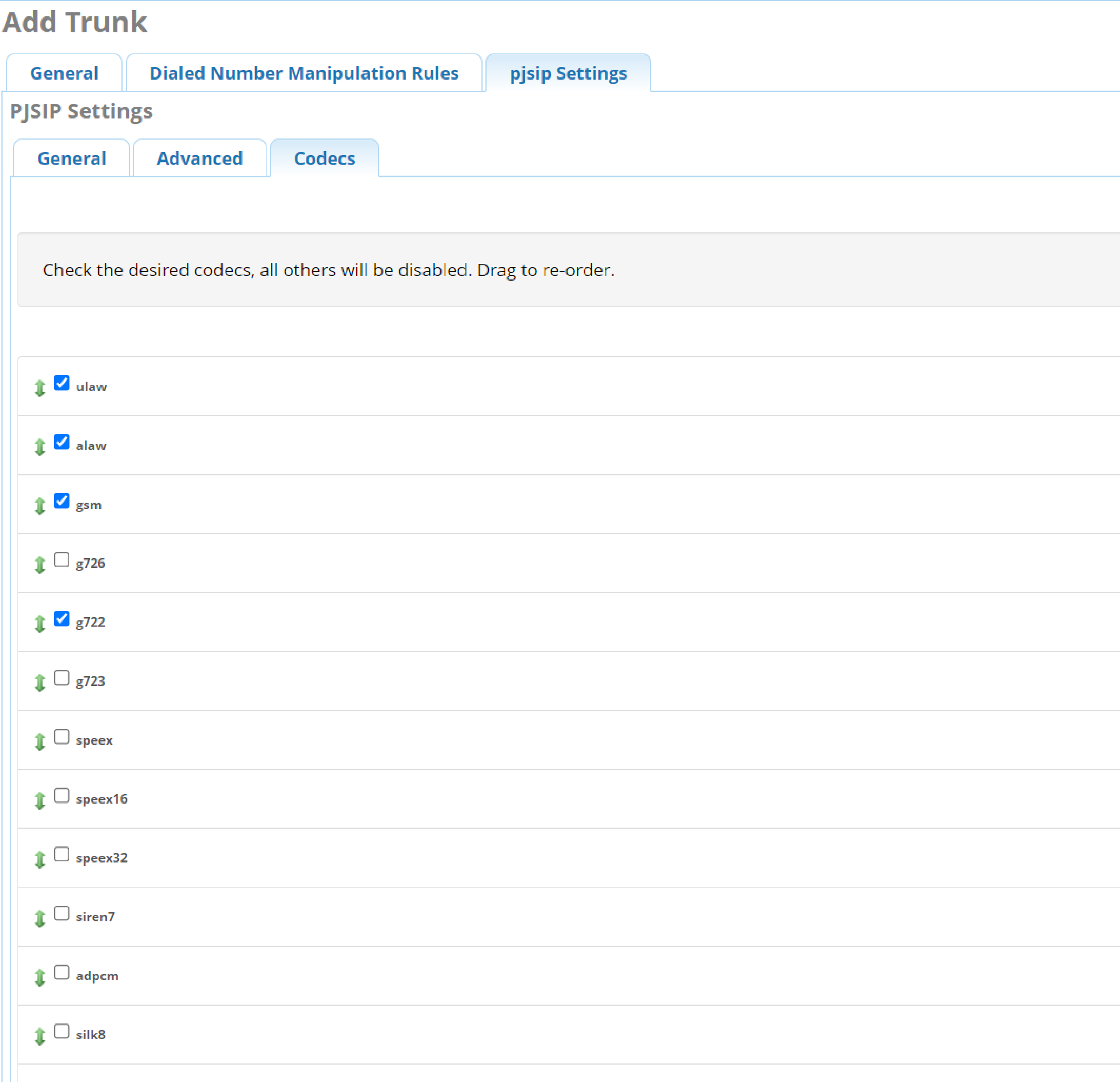
Force RFC3581 compliant behavior even when no rport parameter exists. Basically always send SIP responses back to the same port we received SIP requests from.

Message Context

Context to route incoming MESSAGE requests to

Codecs

Check the desired codecs, all others will be disabled. Drag to re-order.



5. Reports

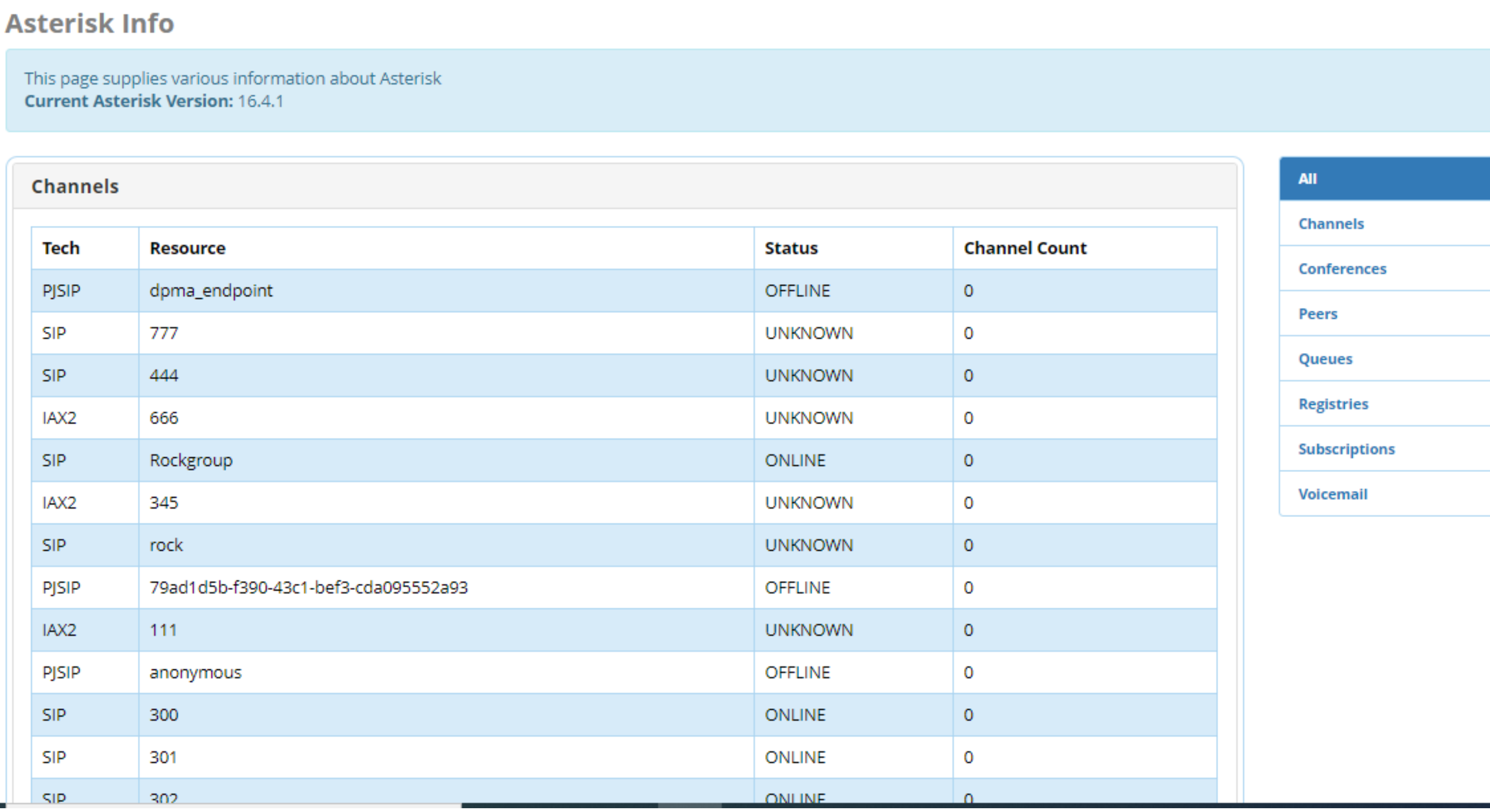
5.1 Asterisk Info

The Asterisk Info page gives you the ability to look at key things in Asterisk such as extension registration information or "BLF Hints" amongst other items and is usually used to debug issues.

This is for advanced users who understand Asterisk.

Logging in

- From the top menu click **Reports**
- From the drop down click **Asterisk Info**


Asterisk Info

This page supplies various information about Asterisk
Current Asterisk Version: 16.4.1

Channels

Tech	Resource	Status	Channel Count
PJSIP	dpma_endpoint	OFFLINE	0
SIP	777	UNKNOWN	0
SIP	444	UNKNOWN	0
IAX2	666	UNKNOWN	0
SIP	Rockgroup	ONLINE	0
IAX2	345	UNKNOWN	0
SIP	rock	UNKNOWN	0
PJSIP	79ad1d5b-f390-43c1-bef3-cda095552a93	OFFLINE	0
IAX2	111	UNKNOWN	0
PJSIP	anonymous	OFFLINE	0
SIP	300	ONLINE	0
SIP	301	ONLINE	0
SIP	302	ONLINE	0

All
Channels
Conferences
Peers
Queues
Registries
Subscriptions
Voicemail

All

All will show us a snap shot of the following information:

- **Active SIP Channels** - How many active SIP channels. Please note this does not mean active calls, as a single call can be 2 or more SIP channels.
- **Active IAX2 Channels** - How many active IAX2 channels. Please note this does not mean active calls, as a single call can be 2 or more IAX2 channels.
- **SIP Registry** - How many SIP connections Asterisk is registered to. This is usually only for SIP trunks because a phone registers to Asterisk, not Asterisk registering to the device.
- **IAX2 Registry** - How many IAX2 connections Asterisk is registered to. This is usually only for IAX2 trunks because a phone registers to Asterisk, not Asterisk registering to the device.
- **SIP Peers** - How many SIP peers are online and offline. A SIP peer is a extension or trunk.
- **IAX2 Peers** - How many IAX2 peers are online and offline. A IAX2 peer is a extension or trunk.
- **Active PJSIP Channels** - How many active SIP channels. Please note this does not mean active calls, as a single call can be 2 or more SIP channels.
- **PJSIP Registry** - How many PJSIP connections Asterisk is registered to. This is usually only for PJSIP trunks because a phone registers to Asterisk, not Asterisk registering to the device

Channels

Here we will see any active channels. A channel is a single communication between 2 devices, such as from Asterisk to a phone or from a trunk to Asterisk.

A typical call will show 2 channels:

- 1 From the trunk to Asterisk
- 1 from Asterisk to the phone.

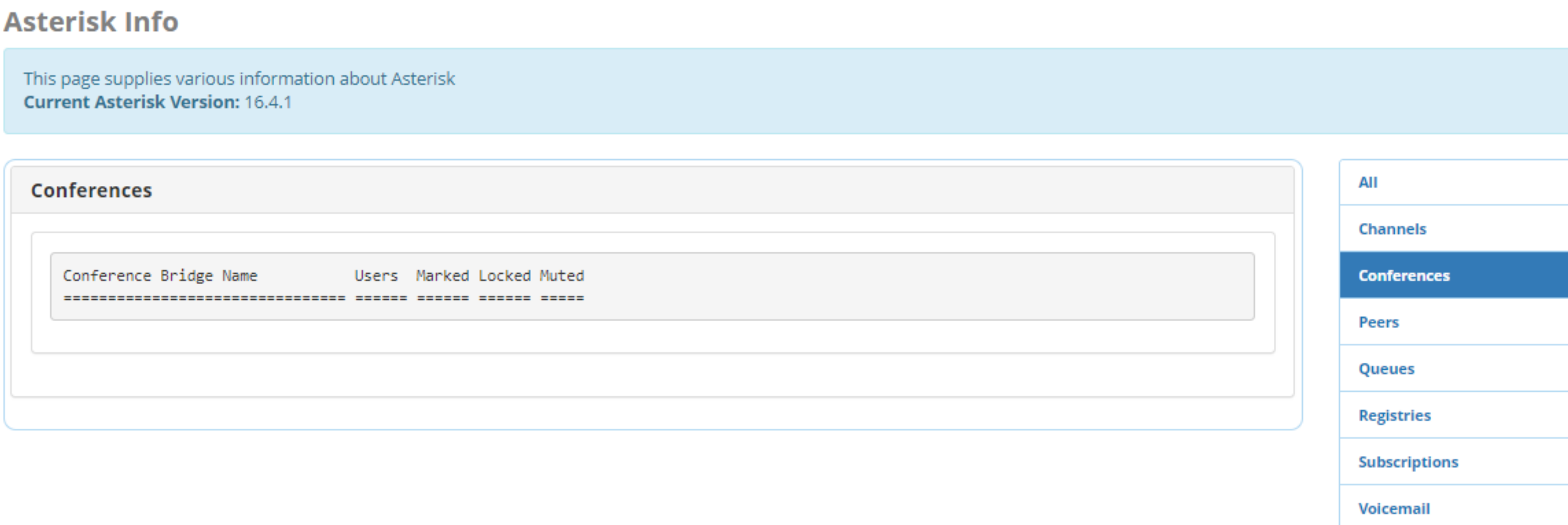
Channels

Tech	Resource	Status	Channel Count
PJSIP	dpma_endpoint	OFFLINE	0
SIP	777	UNKNOWN	0
SIP	444	UNKNOWN	0
IAX2	666	UNKNOWN	0
SIP	Rockgroup	ONLINE	0
IAX2	345	UNKNOWN	0
SIP	rock	UNKNOWN	0
PJSIP	79ad1d5b-f390-43c1-bef3-cda095552a93	OFFLINE	0
IAX2	111	UNKNOWN	0
PJSIP	anonymous	OFFLINE	0
SIP	300	ONLINE	0

- All
- Channels
- Conferences
- Peers
- Queues
- Registries
- Subscriptions
- Voicemail

Conferences Report

Conferences will show you any active conference calls on your system.



Peers

Peers are devices or trunks that are registering and connecting to Asterisk. When viewing a peer, we get some useful information.

PJSIP Info

PJSip Info combines the Registry and Peers report into 1 view, but only showing you the PJSIP Peers and Registries not SIP or IAX2.

CHANSIP Info

CHANSIP Info combines the Registry and Peers report into 1 view, but only showing you the SIP Peers and Registries not IAX2.

CHANSIP

```
Name/username            Host                                Dyn Forcerport Comedia    ACL Port     Status      Description
300/300                  192.168.1.2                          D  No         No          A  5060     OK (7 ms)
301/301                  192.168.1.180                        D  No         No          A  5060     OK (7 ms)
302/302                  192.168.1.179                        D  Yes        Yes         A  5060     OK (7 ms)
303/303                  192.168.1.168                        D  No         No          A  5060     OK (7 ms)
304/304                  192.168.1.181                        D  No         No          A  5060     OK (6 ms)
305/305                  192.168.1.99                         D  No         No          A  5060     OK (7 ms)
306/306                  192.168.1.133                        D  Yes        Yes         A  5060     OK (7 ms)
307                      (Unspecified)                        D  No         No          A  0        UNKNOWN
308                      (Unspecified)                        D  No         No          A  0        UNKNOWN
309/309                  192.168.111.131                      D  No         No          A  5060     OK (113 ms)
311/311                  192.168.111.130                      D  Yes        Yes         A  5060     OK (95 ms)
312/312                  192.168.111.133                      D  No         No          A  5060     OK (79 ms)
313/313                  192.168.1.4                          D  Yes        Yes         A  5060     OK (9 ms)
314/314                  37.34.211.246                        D  Yes        Yes         A  45521    OK (54 ms)
315/315                  37.37.124.164                        D  Yes        Yes         A  1024     OK (65 ms)
```

IAX Info

IAX Info combines the Registry and Peers reports into 1 view, but only showing you the IAX2 Peers and Registries not SIP.

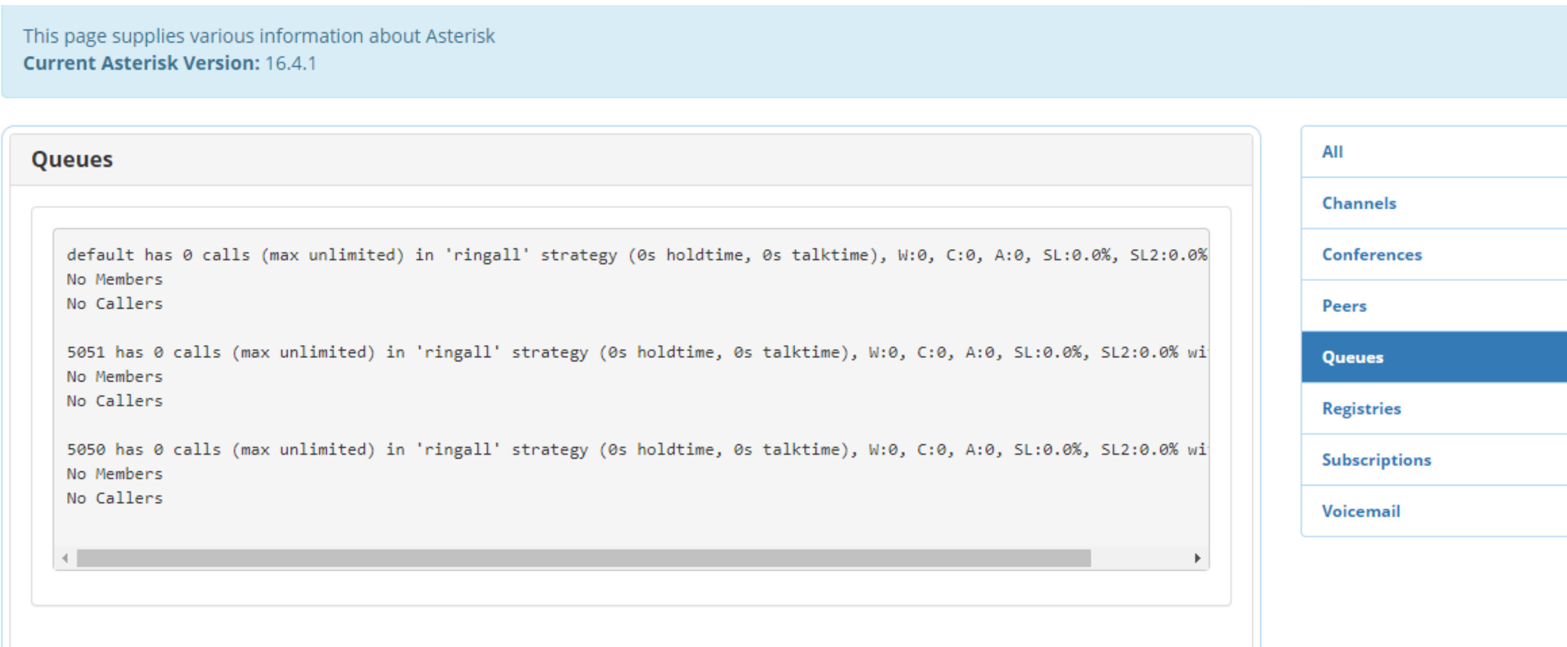
IAX2

```
Name/Username    Host                                    Mask                                Port          Status
111              (null)                           (D)  (null)                               (null)        UNKNOWN
666              (null)                           (D)  (null)                               (null)        UNKNOWN
345              (null)                           (D)  (null)                               (null)        UNKNOWN
3 iax2 peers [0 online, 3 offline, 0 unmonitored]
```

Queues Report

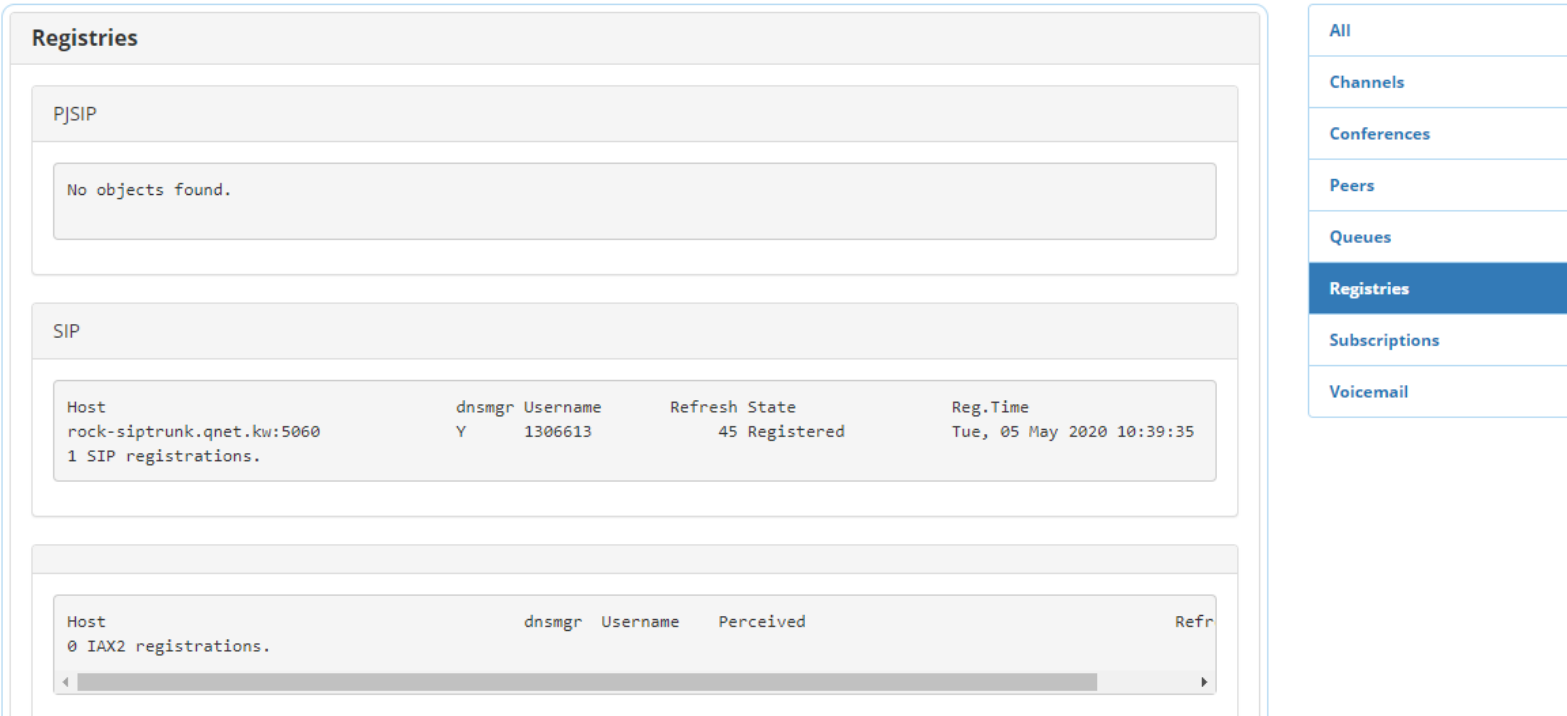
The Queues report will show us each queue we have setup and some quick real time stats, such as:

- Waiting Callers
  - Logged in members and their current state o Paused
  - Not Paused
  - In use
  - Not-In use



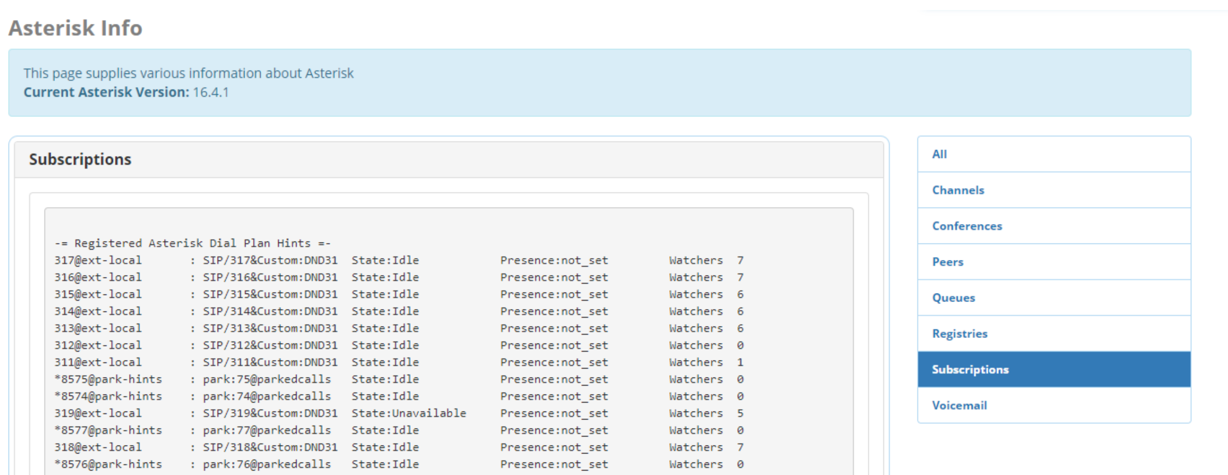
Registries

Registries will show you each connection that Asterisk is registered to. This is usually a trunk, as Asterisk registers to a trunk. This only shows what Asterisk is registered to, not what is registering to Asterisk. Please see "Peers" to see devices and trunks that are registered to Asterisk.



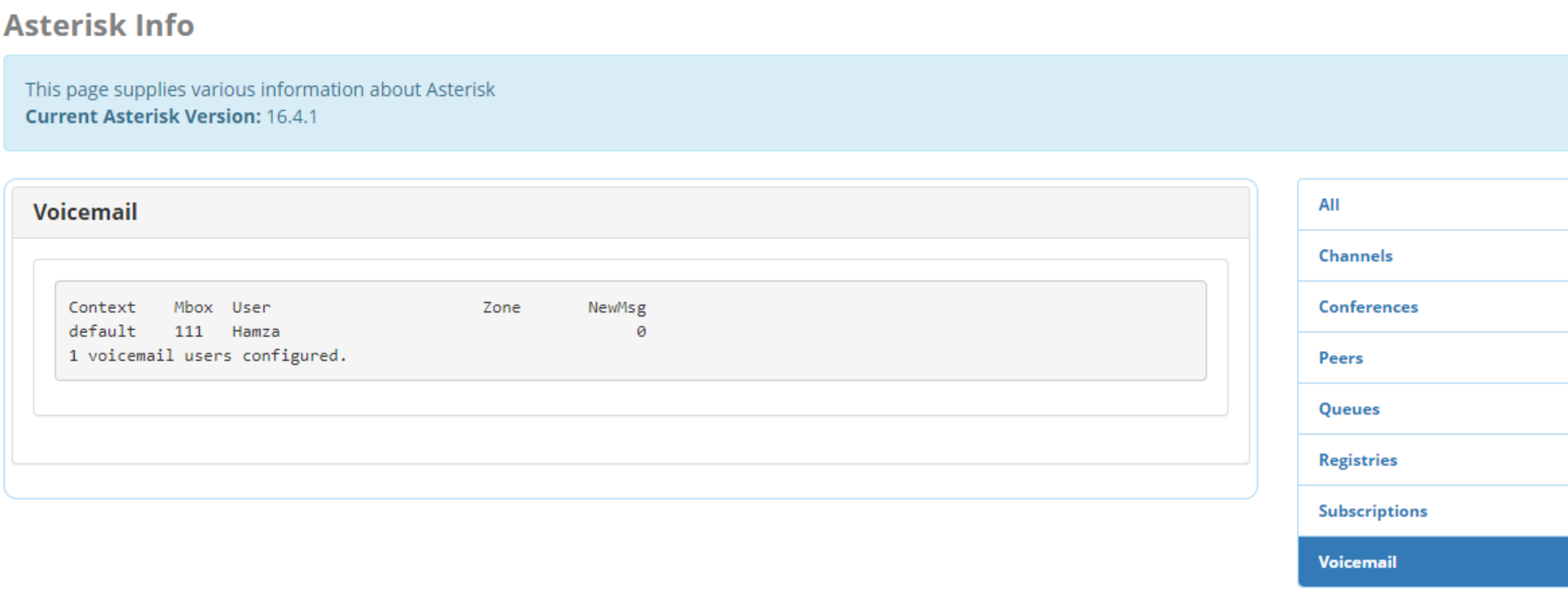
Subscription Report

Subscriptions will show you a list of all hints that are created on your system. A hint is what you subscribe a BLF button on your phone to. A good example might be having a BLF button to use 101, so anytime user 101 is on a call the button will be red. Here we can see all the hints and how many users (watchers) are subscribed to any single hint.



Voicemail Users Report

At a glance we can see all the voicemail boxes that are created and how many new voicemails each user has.



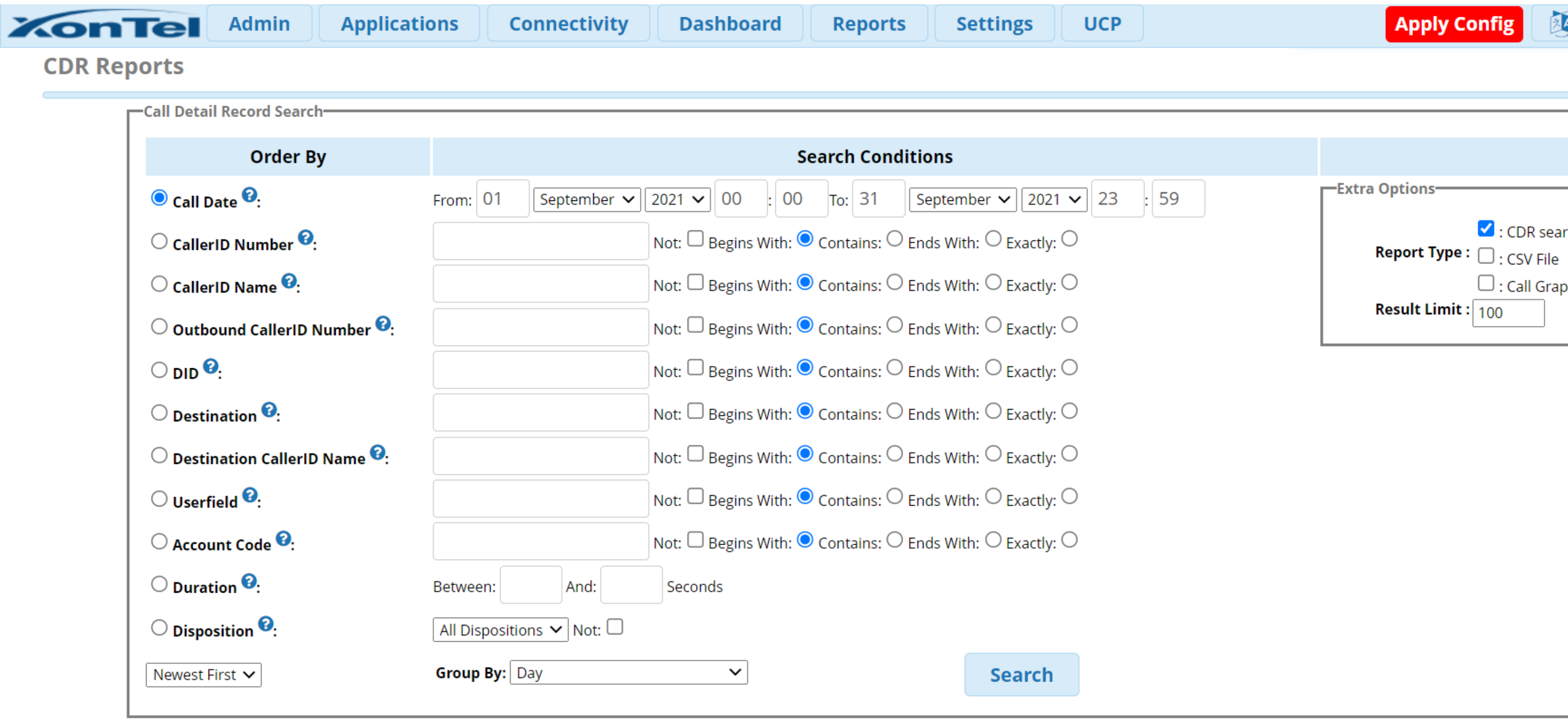
5.2 CDR Report

Call Reports is designed to be the raw data of all call activity on your phone system. It can be a very challenging module to work with because it’s not in a very user-friendly format and there is so
much raw Call Detail Records (CDR). It is really meant as a way to export the data so you can build your own custom reports around the raw data.

Logging in

- From the top menu click **Reports**
- From the drop down click **CDR Reports**

Viewing Data



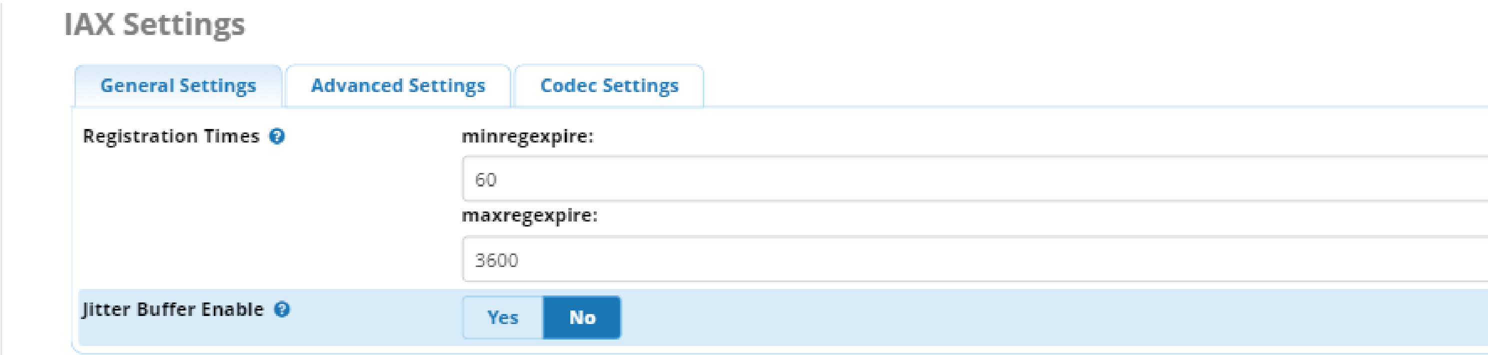
From the landing page you can run reports against your CDR database and filter on the following:

- Call Date
- CallerID Number
- CallerID Name
- Outbound CallerID Number
- DID
- Destination
- Destination CallerID Name
- Userfield
- Account Code
- Duration
- Disposition

6. Asterisk General Settings

6.1 Asterisk IAX Settings

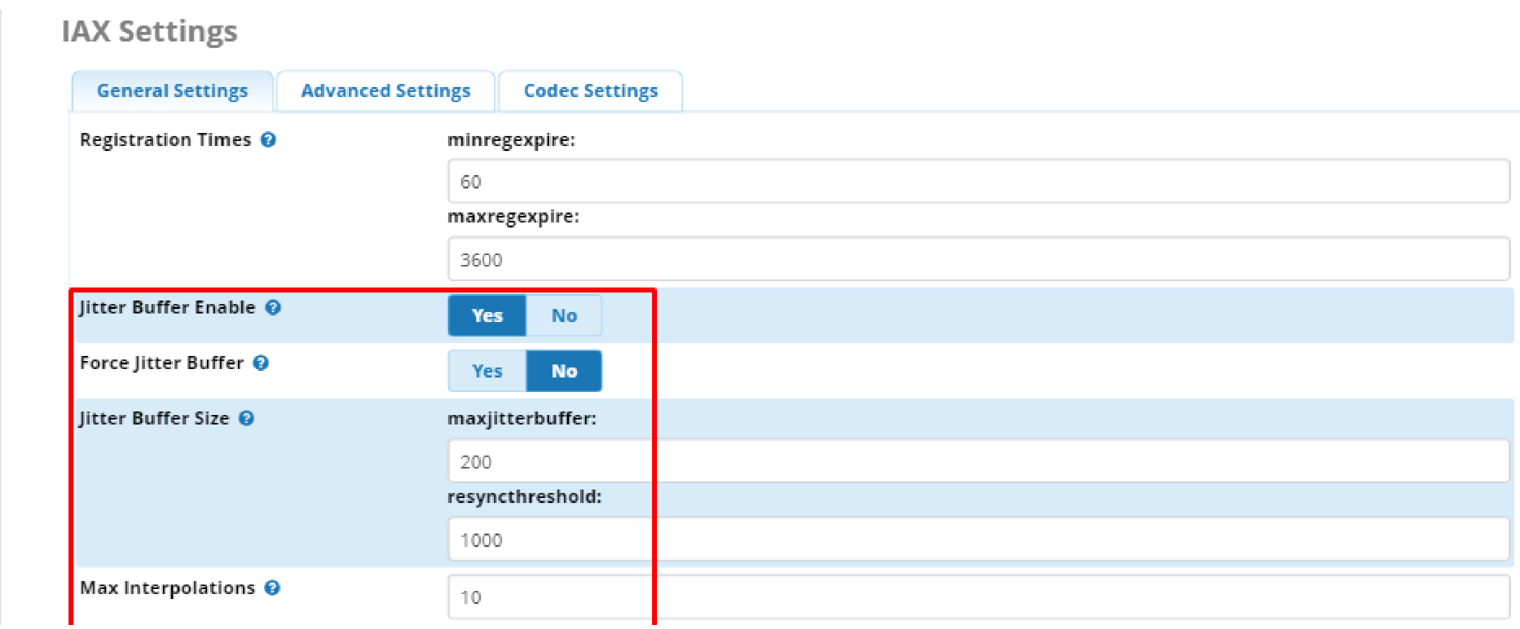
From **Settings** choose **Asterisk IAX Settings**



General Settings

Registration Times - The time span of the registration request for IAX nodes.

- minregexpire - minimum registration time.
- maxregexpire - minimum registration time.



Jitter Buffer Enable

Force Jitter Buffer - if set to Yes, Jitter Buffer will be used on the receiving side of the IAX channel, not on SIP.

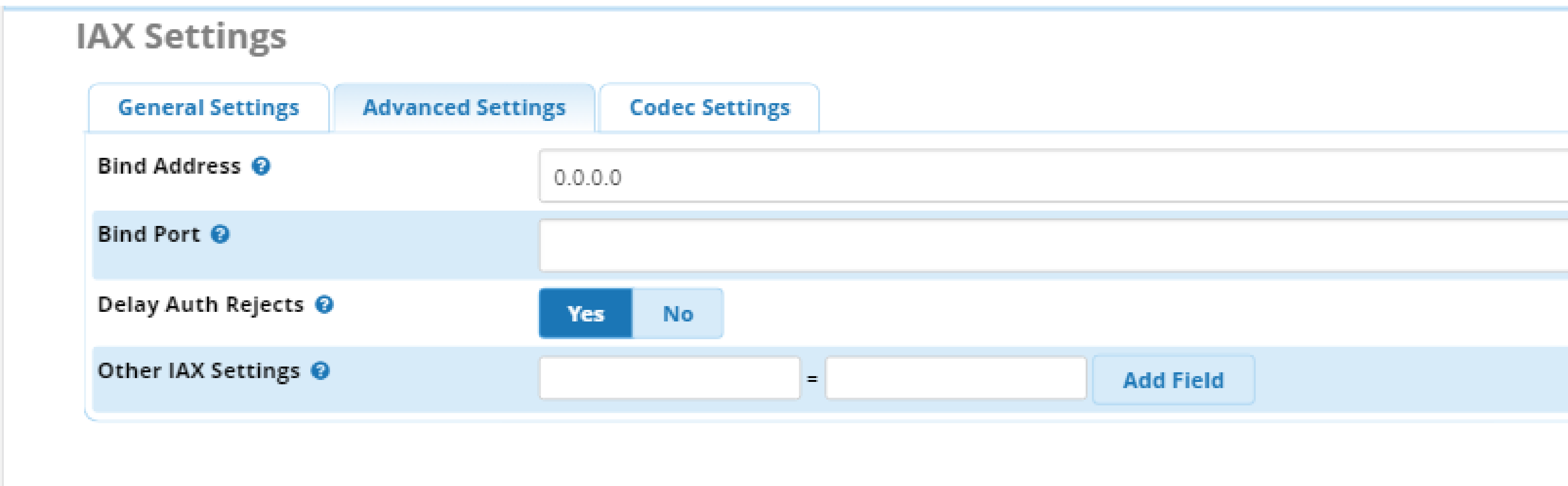
Enabling Force Jitter Buffer can add additional latency to the stream.

Jitter Buffer Size

- maxjitterbuffer - maximum jitterbuffer value.
- resyncthreshold - the period during which the delays could have been changed, the resynchronization will be performed.

Max Interpolations - The number of interpolation frames that must return prevents interpolation for a long time of silence.

Advanced Settings



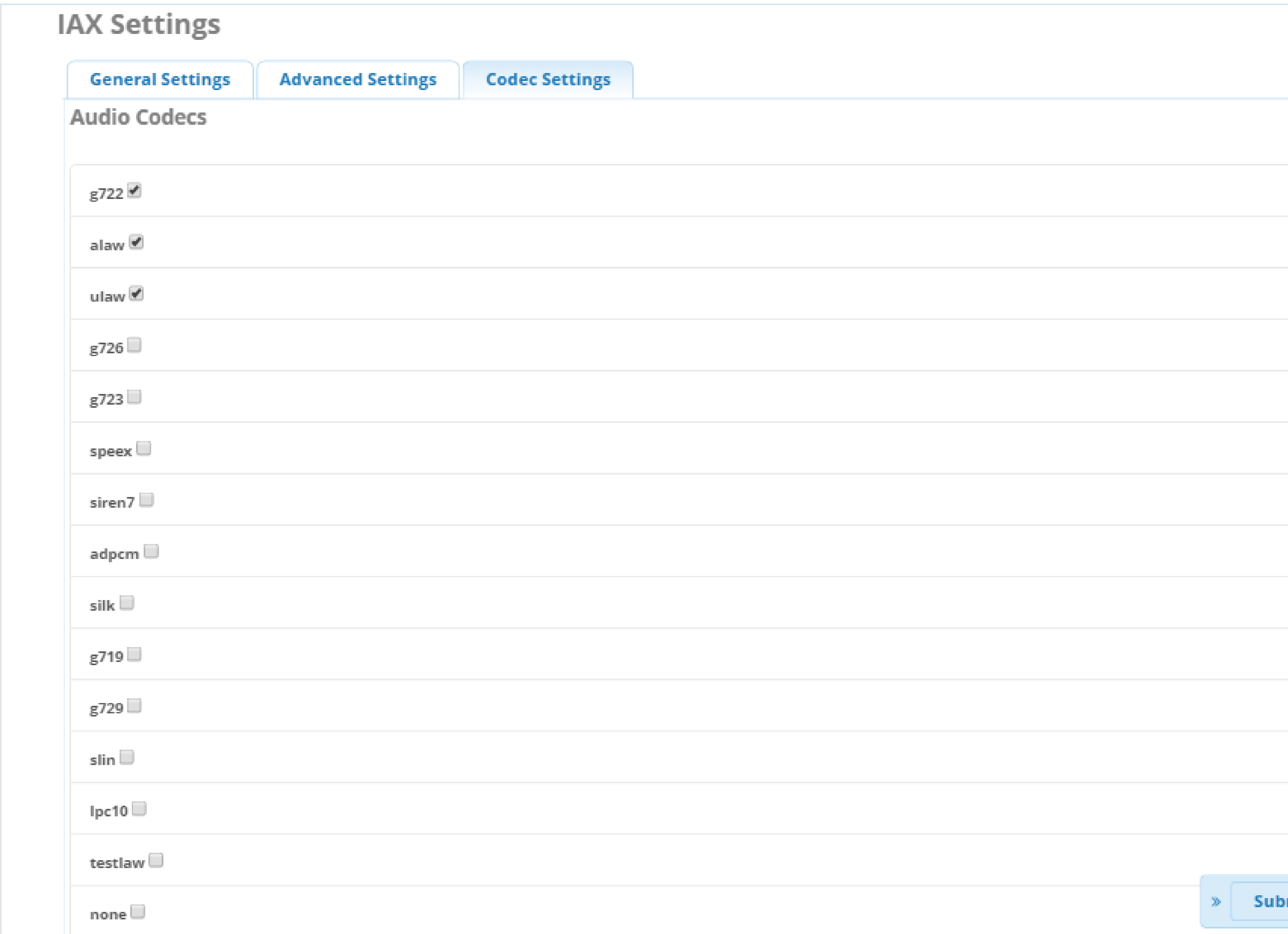
- Bind Address – IP address for listening. The default is 0.0.0.0
- Bind Port - UDP port for communication with Asterisk, by default **- 4569**, we recommend leaving this field blank.
- Delay Auth Rejects - rejection of authentication delays. Enable / disable the parameter. By default - enabled.
- Other IAX Settings - other IAX settings. It used if necessary to introduce additional parameters that can be set up under General config iax . conf .
- Input format: [parameter] = [value]

Now click on Codec Settings

Section for managing codecs for connecting IAX.

The following codecs are marked by default:

- Alaw - providing dynamic range (64 Kbps).
- Ulaw - Provides the best voice quality (64 Kbps).



Codec Settings

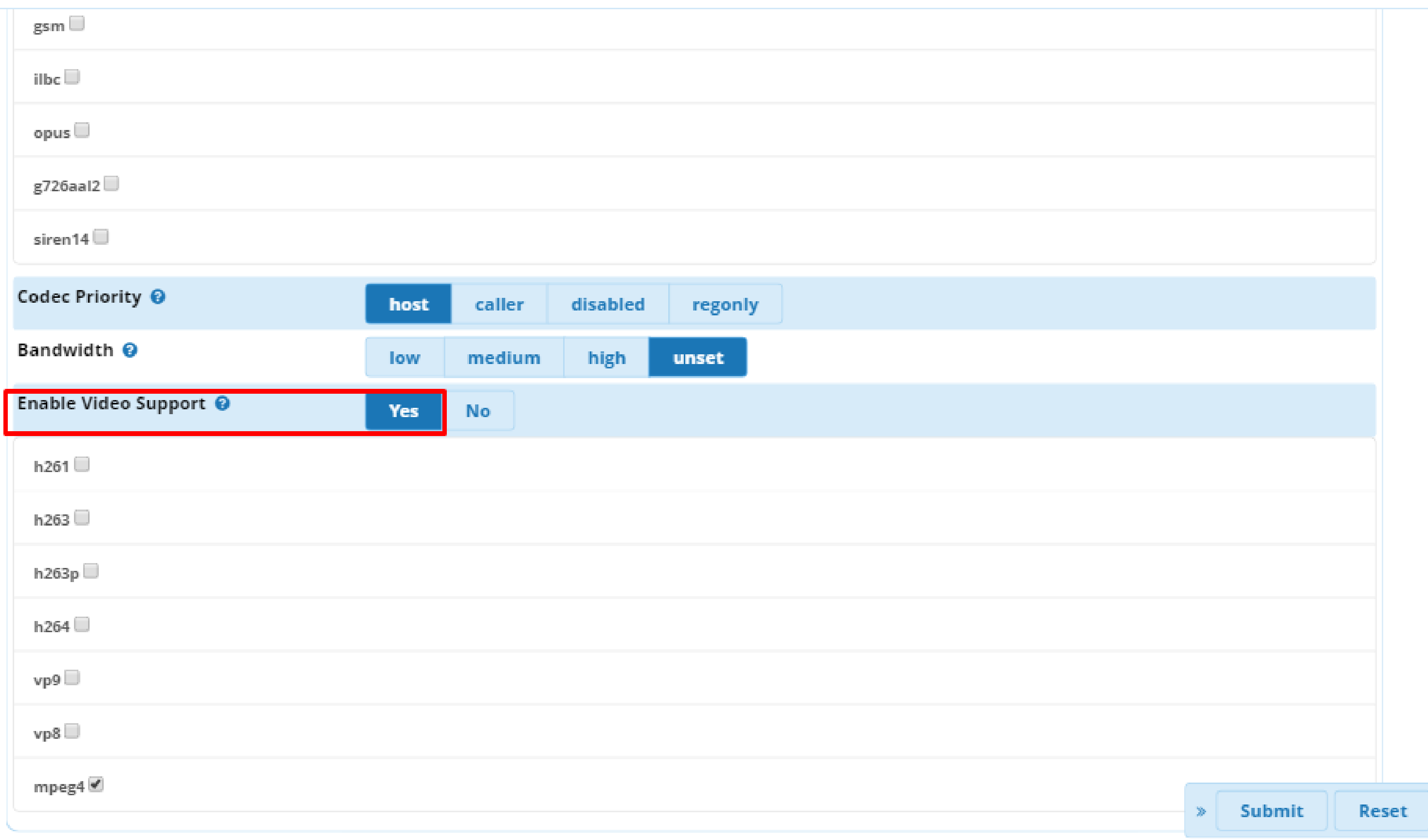
Codec Priority - definition of codecs for incoming calls via the IAX protocol. The selected option will apply to all incoming numbers. At the same time, the order can be defined separately for each number, but these settings will have priority over those configured in this module.

Priorities:

- Host - host-side codec management.
- Caller - control of codecs on the calling side.
- Disabled - disable codec priority control for both sides.
- Regonly - accepting a call when the codec matches the system format.

Bandwidth - bandwidth parameter control: low, medium, high, unset. By default, unset is not specified.

Enable Video Support - enable the use of video when possible.



Video Codecs

If video will be used, then it is necessary to check the correspondence of the codecs.

The priority of using the codecs will depend on the order in which the codec is selected.

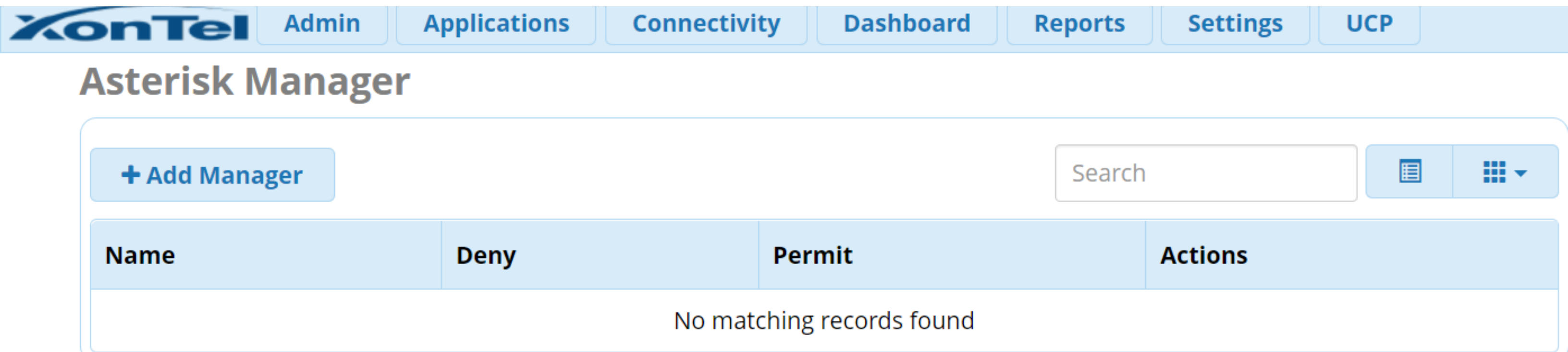
Summing up: if you are going to connect several PBXs via an IAX trunk, you need not only to create a trunk, but also go through this module - Asterisk IAX Settings to improve the settings.

6.2 Asterisk Manager Users

The Asterisk API (aka Asterisk Manager API) is the Application Program Interface for/to the Asterisk Manager and allows for external systems to connect via TCP/IP to issue commands and read events. Common examples of usage include Dialers, CRM, Management Console and so on.

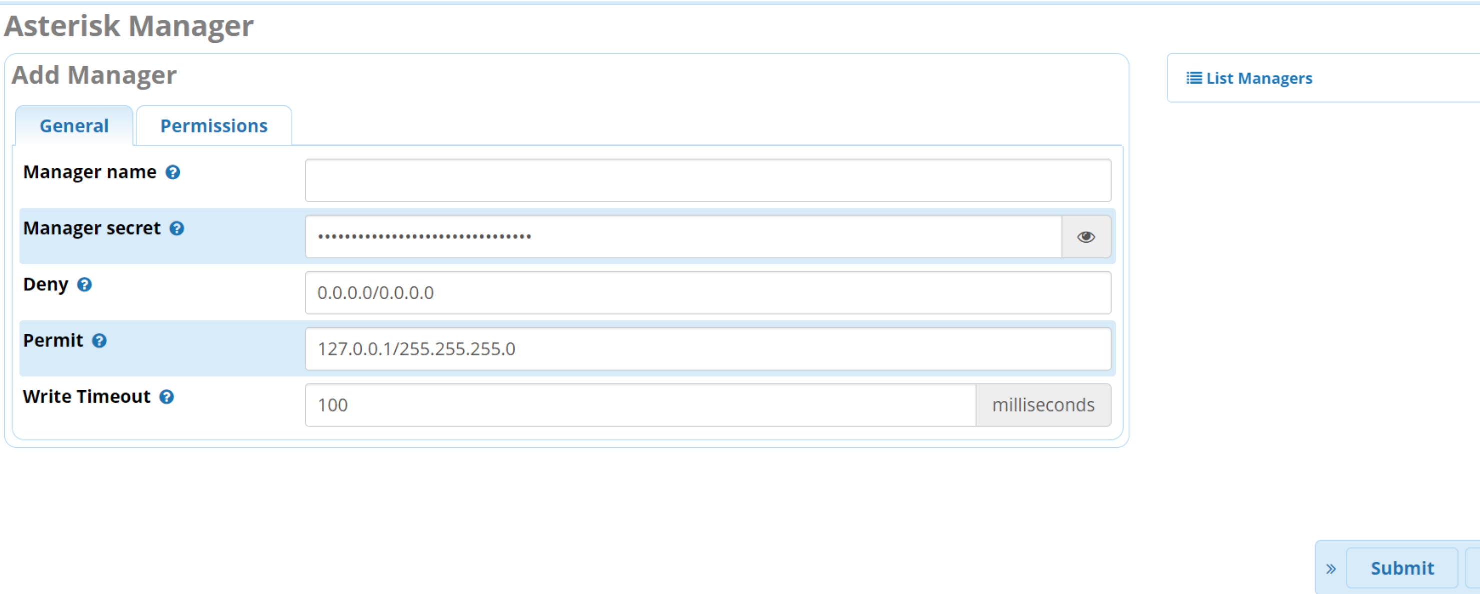
Logging In

- From the top menu click **Settings**
- In the drop down click **Asterisk Manager Users**



Add Manager

Manager Information



Manager Name

Name of the Manager account. No spaces are allowed.

Manager Secret

Password for the Manager.

Deny

Here you define an IP Address/Subnet Mask Deny statement. If you wish to add more than one network, use the "&" character as a separator. i.e. 192.168.0.0/255.255.0.0&10.0.10.0/255.255.255.0

Permit

Here you define an IP Address/Subnet Mask Permit statement. You may define more than one network or device as with the Deny statement.

Write Timeout
Sets the timeout used by Asterisk when writing data to the AMI connection for this user

Permissions

You may assign various read/write permissions to each Manager.

Asterisk Manager

Add Manager

General | Permissions

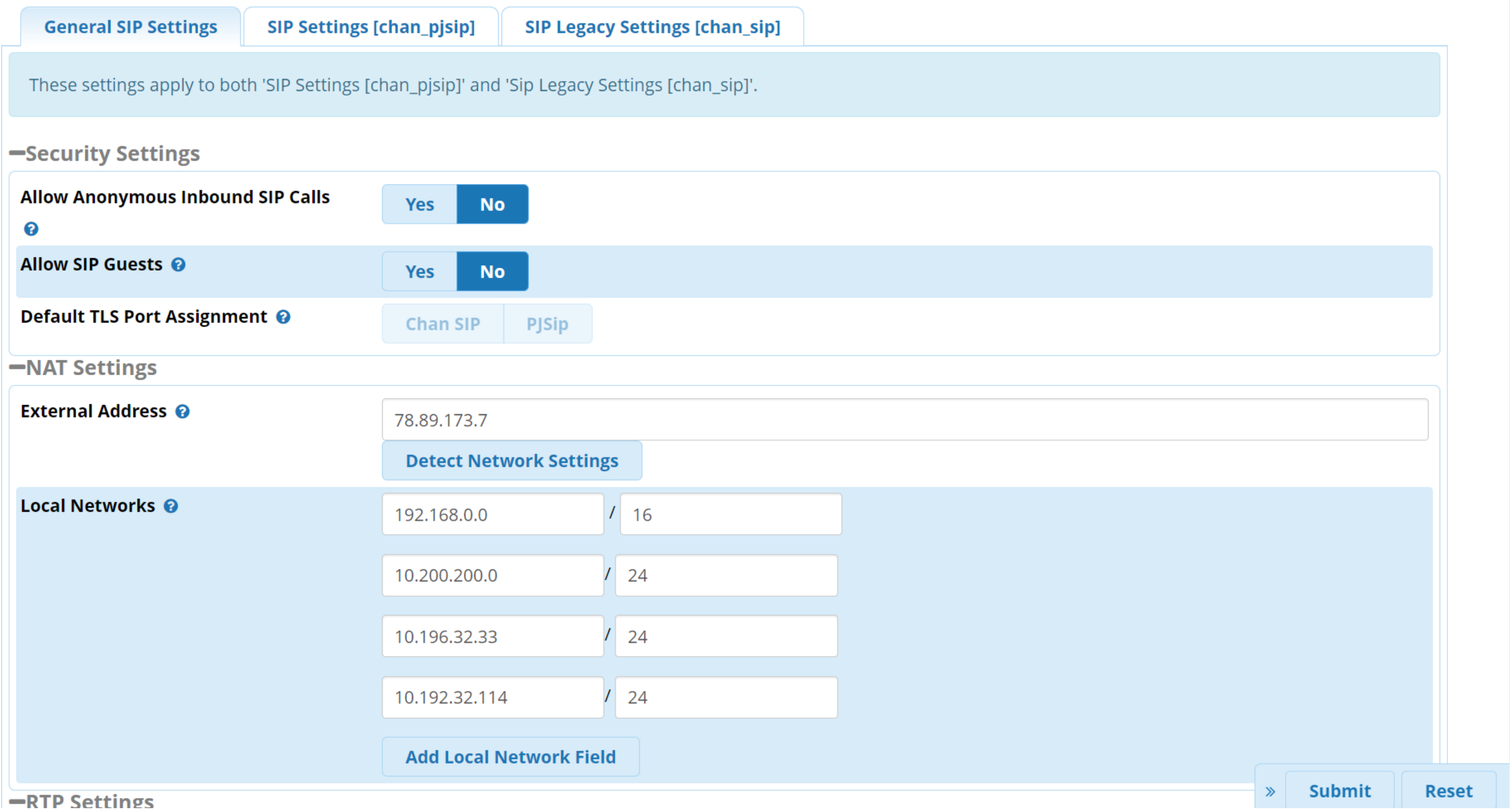
For information on individual permissions please see the Asterisk Manager Documentation

Permission	Read	Write
system	Yes No	Yes No
call	Yes No	Yes No
log	Yes No	Yes No
verbose	Yes No	Yes No
command	Yes No	Yes No
agent	Yes No	Yes No
user	Yes No	Yes No
config	Yes No	Yes No
dtmf	Yes No	Yes No
reporting	Yes No	Yes No
cdr	Yes No	Yes No
dialplan	Yes No	Yes No
originate	Yes No	Yes No
Toggle All	Yes No	Yes No

6.3 Asterisk SIP Settings

To configure, go to **Settings → Asterisk Sip Settings.**

Let's start with the **General SIP Settings** tab



Security Settings

1. **Allow Anonymous inbound SIP Calls** - allow incoming anonymous SIP calls.

YES - allow incoming calls from unknown IP addresses, IP-PBX will process calls according to the from-pstn context. Allowing anonymous calls poses additional security risks.

If SIP URI dialing is allowed or ENUM service is used, then YES must be set for incoming traffic to work.

The configurable parameter is not related to sip.conf, it is used in the dialplan according to the default context. If the context is changed, then the parameter may become unnecessary, likewise if the Allow SIP Guests parameter is No.

2. **Allow SIP Guests**

When set Asterisk will allow Guest SIP calls and send them to the Default SIP context. Turning this off will keep anonymous SIP calls from entering the system. Doing such will also stop 'Allow Anonymous Inbound SIP Calls' from functioning. Allowing guest calls but rejecting the Anonymous SIP calls below will enable you to see the call attempts and debug incoming calls that may be mis-configured and appearing as guests.

3. Assigning the default TLS port

Controls the SIP protocol that is listening on port 5061. If the options are not available, then the protocol must be enabled, or it does not support TLS. If you change it, you need to restart Asterisk.

NAT Settings

External Address - field for external IP address. When you click on Detect Network Settin, the PBX automatically detects the external IP address parameters and internal local subnets.

Local Network - local subnets from which SIP devices will be connected (IP address / mask). To add subnets, use the Add Local Network button.

Informer: These parameters apply to both chan_sip and chan_pjsip.

RTP setup

RTP Settings

RTP Port Ranges	Start:	10000
	End:	20000
RTP Checksums	Yes / No	
Strict RTP	Yes / No	
RTP Timeout	30	
RTP Hold Timeout	300	
RTP Keep Alive	0	

Media Transport Settings

- STUN Server Address
- TURN Server Address
- TURN Server Username
- TURN Server Password

ICE Blacklist

What is ICE Blacklist?

ICE Blacklist

What is ICE Blacklist?

IP Addresses: /

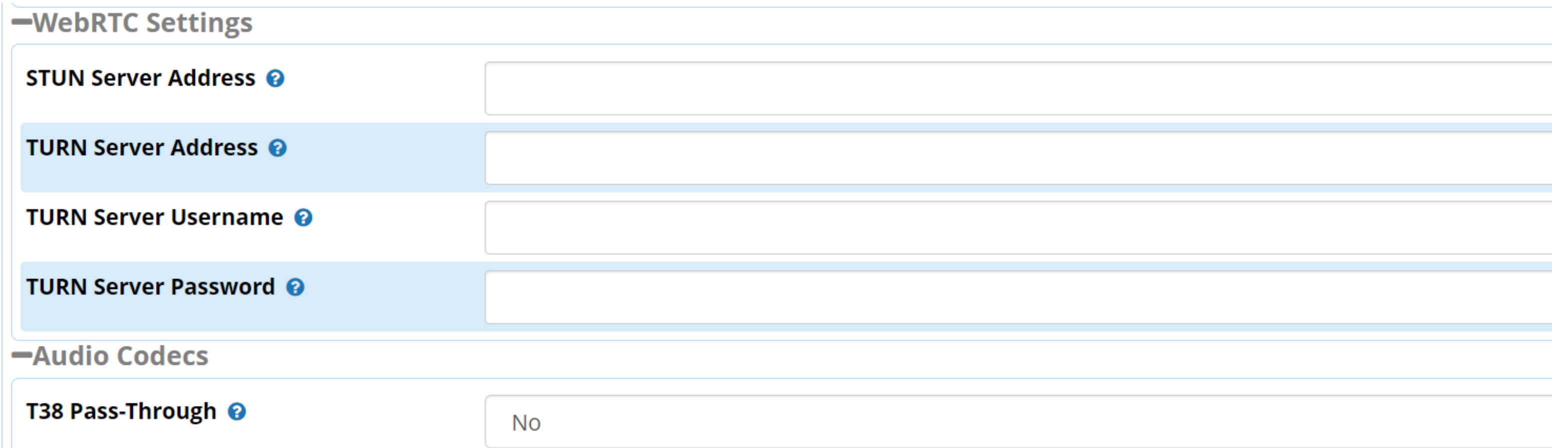
Add Address

ICE Host Candidates

What is ICE Host Candidates?

Candidates: =>

Add Address



1. **RTP Rangers**: start and end UDP ports for RTP traffic. Start / End.
2. **RTP Checksums**: calculates the checksum for UDP from RTP traffic.
3. **Strict RTP** - discard RTP packets that do not come from the source of the RTP stream within the session.
4. **RTP Timeout** - Terminate call if rtptimeout seconds of no RTP or RTCP activity on the audio channel when we're not on hold. This is to be able to hangup a call in the case of a phone disappearing from the net, like a power loss or someone tripping over a cable.
5. **RTP Hold Timeout** - Terminate call if rtpholdtimeout seconds of no RTP or RTCP activity on the audio channel when we're on hold (must be > rtptimeout).
6. **RTP Keep Alive** - Send keepalives in the RTP stream to keep NAT open during periods where no RTP stream may be flowing (like on hold).

Media Transport Settings

1. **STUN Server Address**: Hostname or address for the STUN server used when determining the external IP address and port an RTP session can be reached at. The port number is optional. If omitted the default value of 3478 will be used. This option is blank by default.
2. **Turn Server Address**: Hostname or address for the TURN server to be used as a relay. The port number is optional. If omitted the default value of 3478 will be used. This option is blank by default.
3. **Turn Server Username** - Username used to authenticate with TURN relay server. This option is disabled by default.

4. **Turn Server Password** - Password used to authenticate with TURN relay server. This option is disabled by default.

ICE Settings

1. **ICE Blacklist** - Subnets to exclude from ICE host, srflx and relay discovery. This is useful to optimize the ICE process where a system has multiple host address ranges and/or physical interfaces and certain of them are not expected to be used for RTP. For example, VPNs and local interconnections may not be suitable or necessary for ICE. Multiple subnets may be listed. If left unconfigured, all discovered host addresses are used.

The format for these overrides is: [address] / [subnet]

This is most commonly used for WebRTC

2. **ICE Host Candicates**: When Asterisk is behind a static one-to-one NAT and ICE is in use, ICE will expose the server's internal IP address as one of the host candidates. Although using STUN (see the 'stunaddr' configuration option) will provide a publicly accessible IP, the internal IP will still be sent to the remote peer. To help hide the topology of your internal network, you can override the host candidates that Asterisk will send to the remote peer.

IMPORTANT: Only use this functionality when your Asterisk server is behind a one-to-one NAT and you know what you're doing. If you do define anything here, you almost certainly will NOT want to specify 'stunaddr' or 'turnaddr' above.

The format for these overrides is: [local address] => [advertised address]>

This is most commonly used for WebRTC

WebRTC Settings

1. **STUN Server Address**: Hostname or address for the STUN server used when determining the external IP address and port an RTP session can be reached at. The port number is optional. If omitted the default value of 3478 will be used. This option is blank by default.

2. **Turn Server Address**: Hostname or address for the TURN server to be used as a relay. The port number is optional. If omitted the default value of 3478 will be used. This option is blank by default.

3. **Turn Server Username** - Username used to authenticate with TURN relay server. This option is disabled by default.

4. **Turn Server Password** - Password used to authenticate with TURN relay server. This option is disabled by default.

Audio Codecs

T38 Passthrough - Asterisk: t38pt_udptl. Enables T38 passthrough which makes faxes go through Asterisk without being processed.

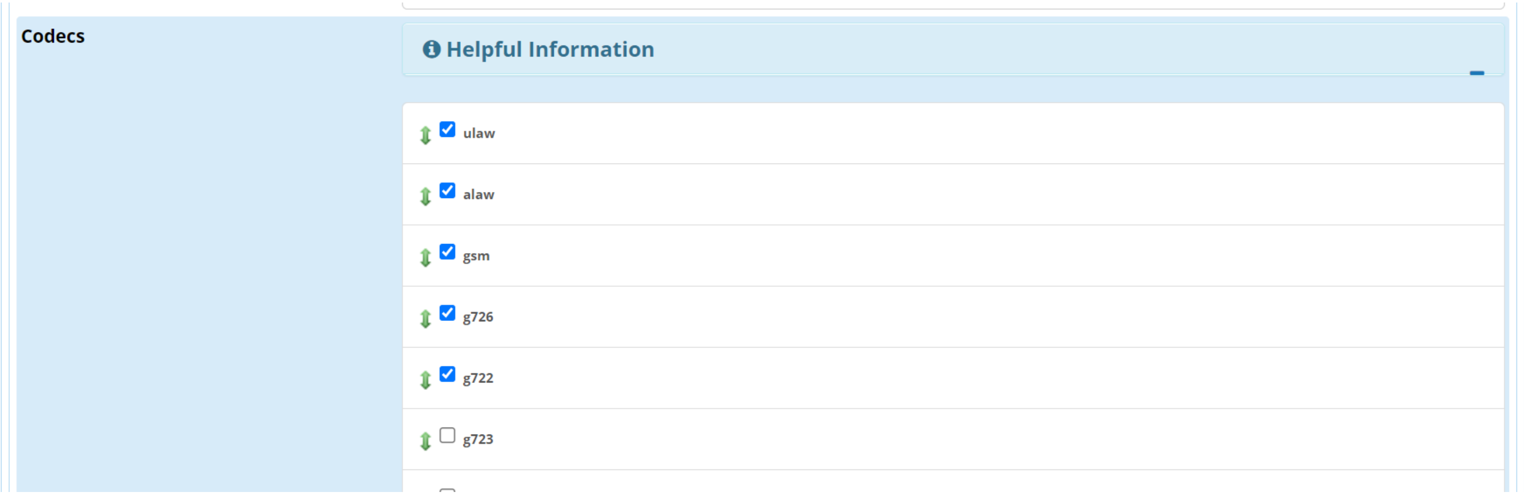
- No - No passthrough
- Yes - Enables T.38 with FEC error correction and overrides the other endpoint's provided value to assume we can send 400 byte T.38 FAX packets to it.
- Yes with FEC - Enables T.38 with FEC error correction
- Yes with Redundancy - Enables T.38 with redundancy error correction
- Yes with no error correction - Enables T.38 with no error correction.

Audio Codecs

Used for encoding and decoding audio data.

It is important to take into account the order of the codecs - it affects the priority of setting the codec within the SDP messages.

Main codecs:
g722, alaw, ulaw, ilbc, g723, g729.

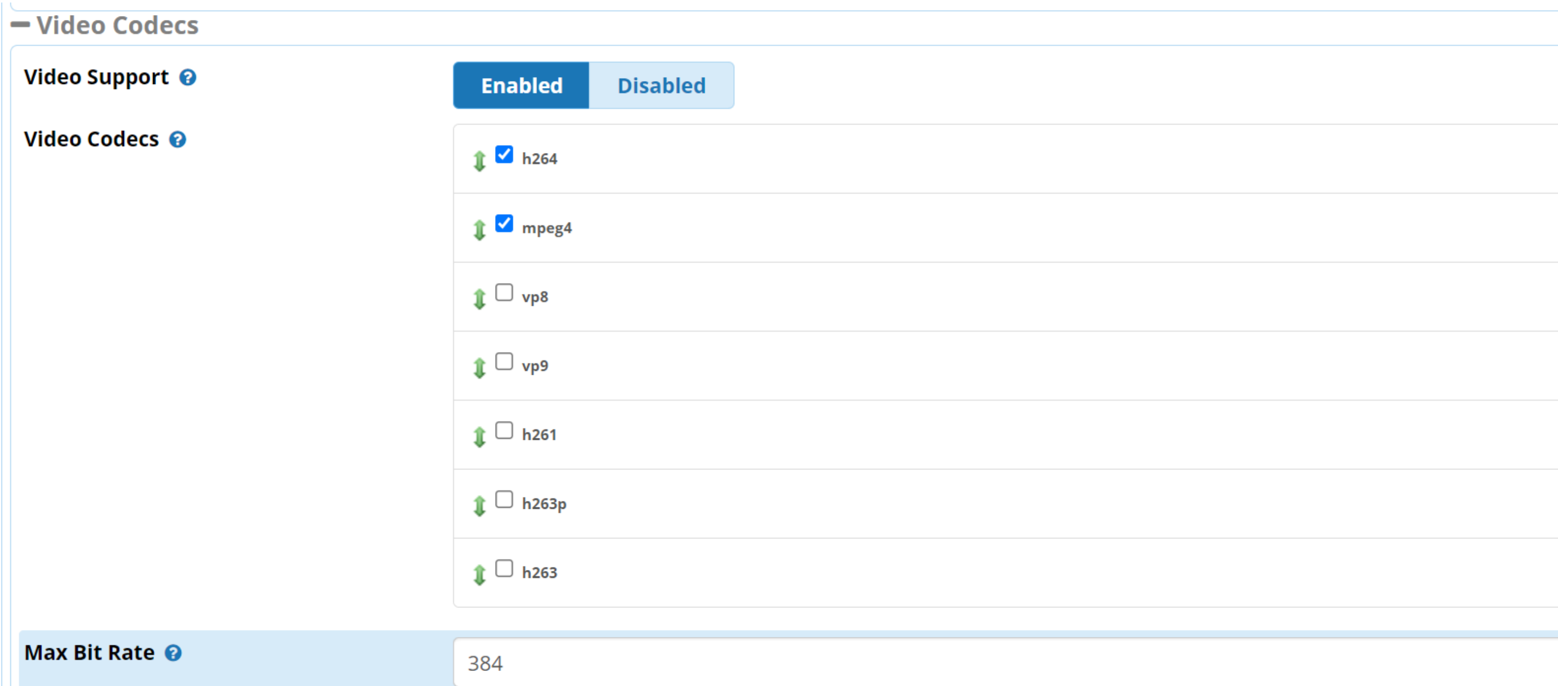


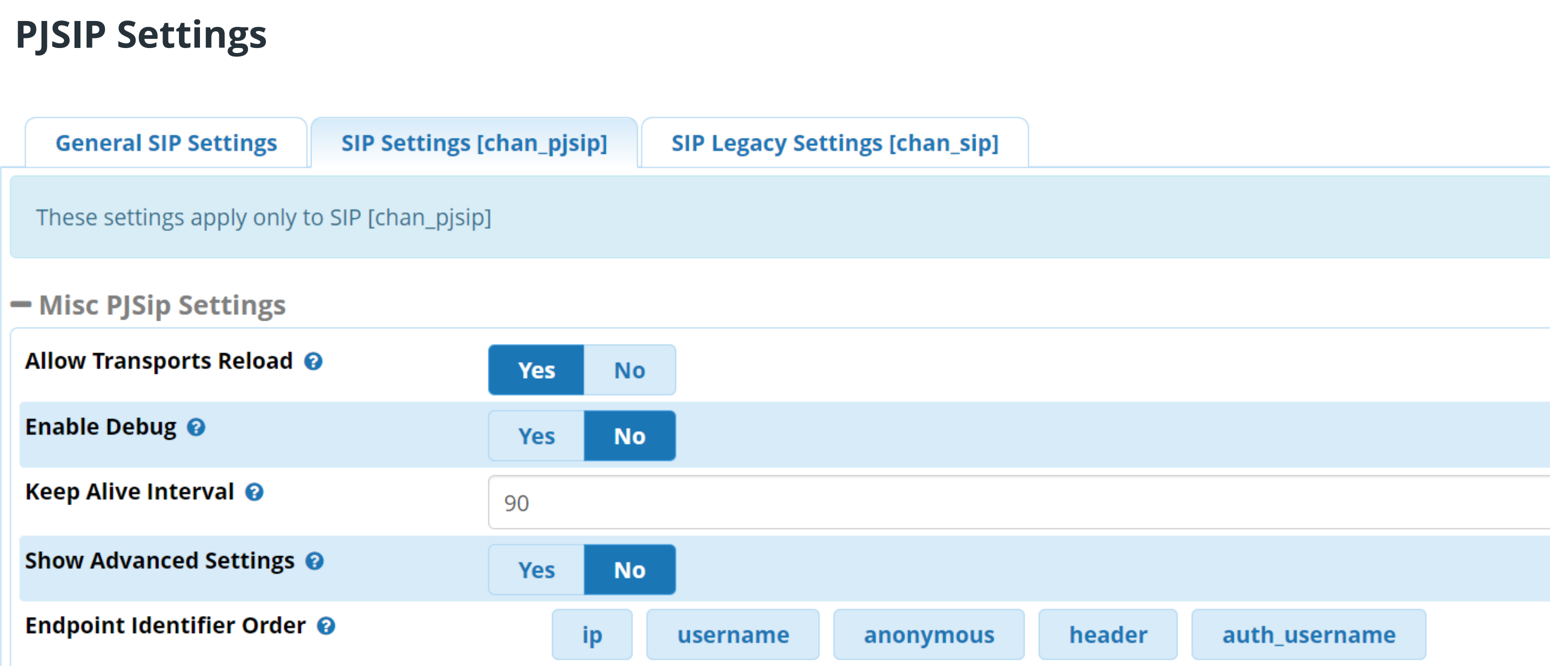
Video Codec

1. **Video Support**: Check to enable and then choose allowed codecs. If you clear each codec and then add them one at a time, submitting with each addition, they will be added in order which will affect the codec priority.

2. **Video Codec**: Choose the video codec that you want to apply in PBX

3. **Max Bit Rate -** Maximum bitrate for video calls in kb/s





Misc PJSIP Settings

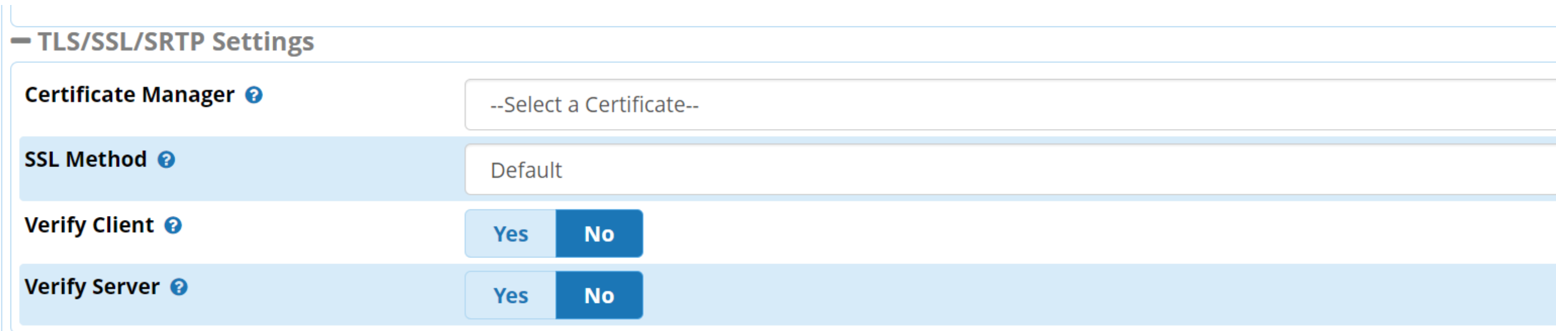
Allow Transport Reload: Allow transports to be reloaded when the PBX is reloaded

Enable Debug: Enable/Disable SIP debug logging.

Keep Alive interval - The interval (in seconds) to send keepalives to active connection-oriented transports.

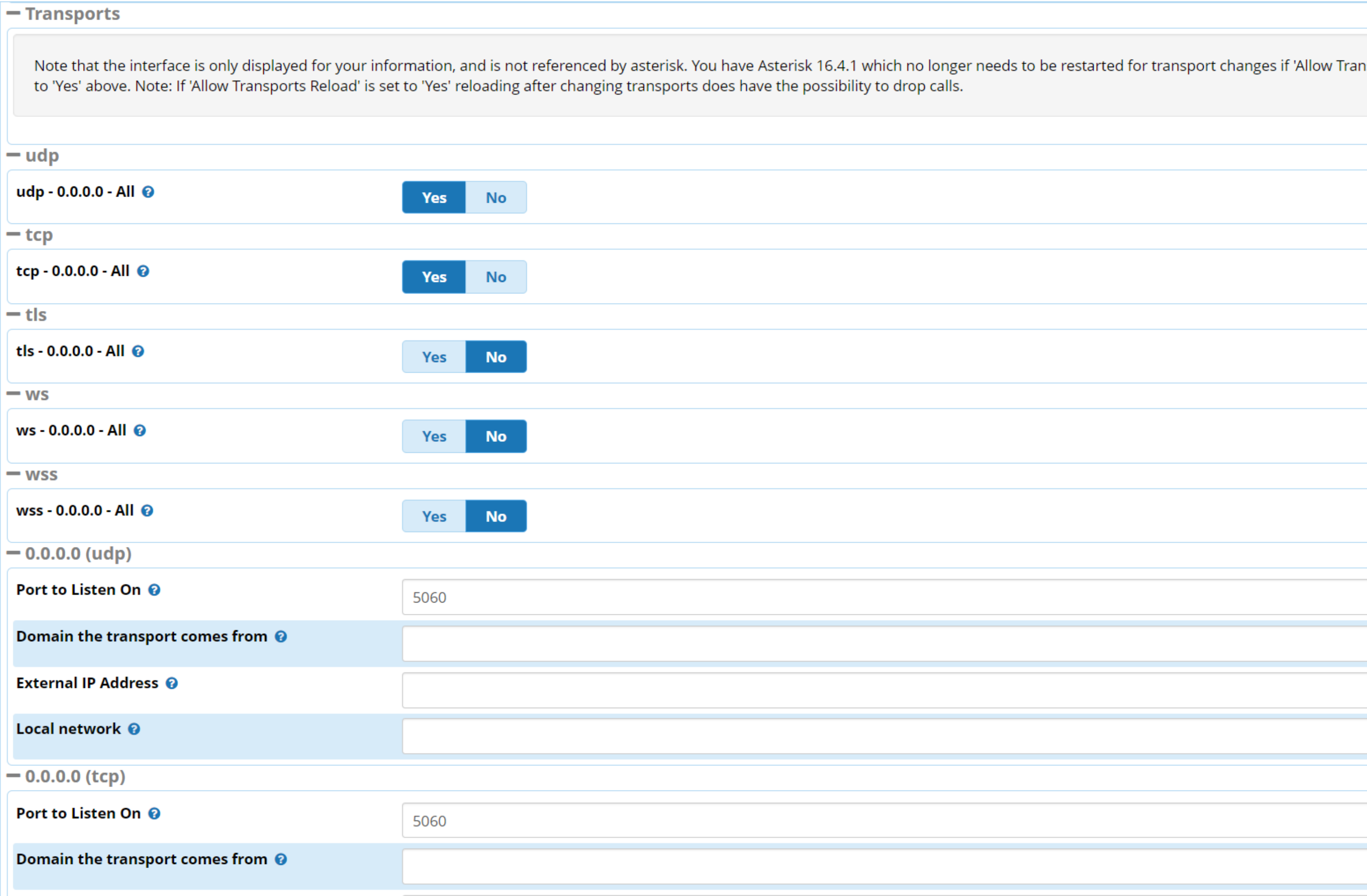
TLS/SSL/RTSP Settings

- **Certificate Manager**: Select a certificate to use for the TLS transport. These are configured in the module Certificate Manager
- **SSL Method**: Method of SSL transport (TLS ONLY). The default is currently TLSv1, but may change with future releases.
- **Verify Client** : Require verification of client certificate (TLS ONLY).
- **Verify Server**: Require verification of server certificate (TLS ONLY).



Transports Settings

Note that the interface is only displayed for your information, and is not referenced by asterisk. You have Asterisk 16.4.1 which no longer needs to be restarted for transport changes if 'Allow Transports Reload' is set to 'Yes' above. Note: If 'Allow Transports Reload' is set to 'Yes' reloading after changing transports does have the possibility to drop calls.



Chan SIP Settings Nat Settings tab
- NAT settings

Chan SIP Settings / Nat Settings

1. NAT:

- Yes - use Nat;
- No - use broadcast according to RFC3581. In short, this RFC allows you to send a response to the port from which the request was received, instead of the port taken from the Via header in the SIP packet;
- ever - do not use NAT according to RFC3581;
- route - this option is suitable for clients that do not use the rport field in SIP message headers (according to RFC3581).

2. IP Configuration - in this field you can specify the parameters of the external IP. You can manually enter your external IP address and also use DDNS (Dynamic DNS).

3. Override External IP: external static IP address or fully qualified domain name as seen on the WAN side of the router.

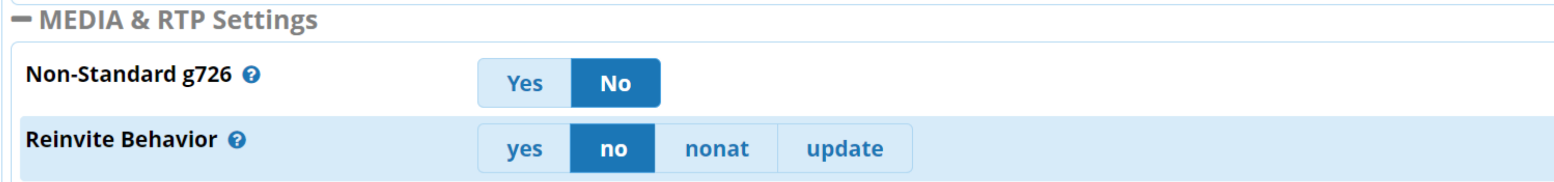
By default, External IP is inherited from the General Sip Settings tab.

TLS / SSL / SRTP Settings - TLS / SSL / SRTP settings

1. Enable TLS - enable support for secure connections via TLS.
2. Certificate Manager - enable certificate for TLS support. It can be easily configured in the Certificate Manager module.
3. SSL Method - SSL transport transmission method (only for TLS). The default is sslv2.
4. Don't Verify Server - do not ask for server certificate verification (setting only affects TLS).

Media & RTP Settings

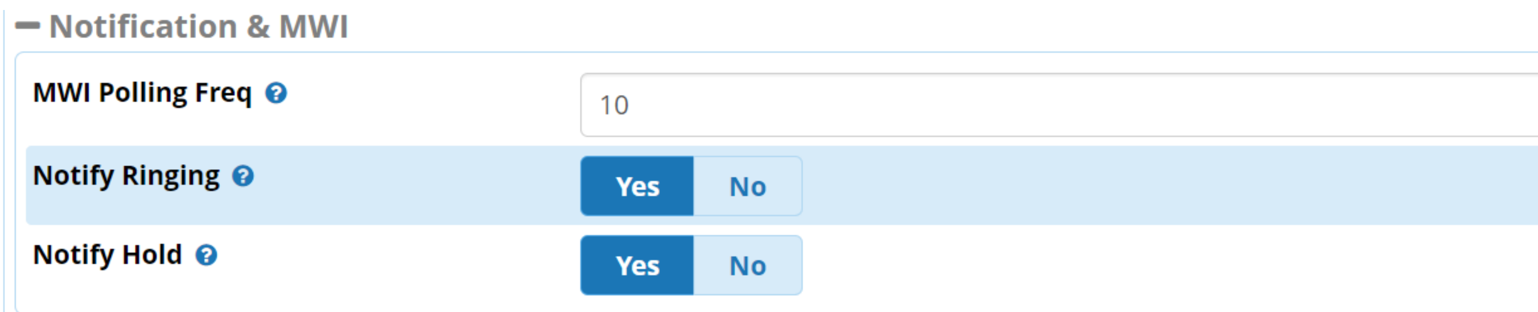
1. Non-Standard g726 - Asterisk: g726nonstandard. If the peer negotiates G726-32 audio, use AAL2 packing order



instead of RFC3551 packing order (this is required for Sipura and Grandstream ATAs, among others). This is contrary to the RFC3551 specification, the peer _should_ be negotiating AAL2-G726-32 instead.

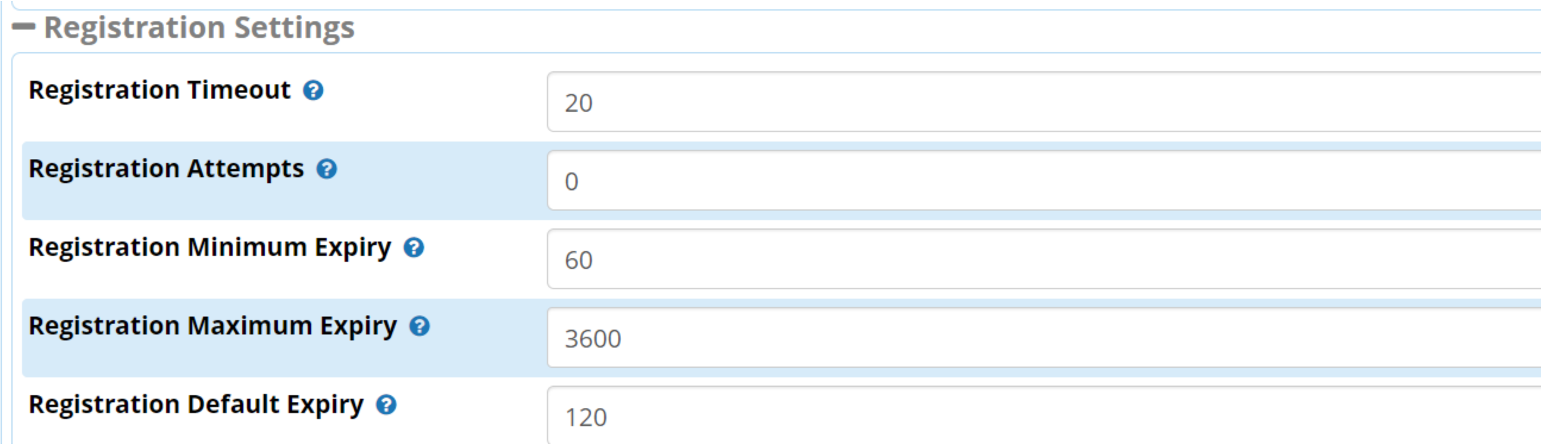
2. Reinvite Behavior - an option that allows you to redirect the RTP data stream if the peer is not behind NAT (using RTP this can be detected by IP addresses).

Notification & MWI - Configuring the verification process and MWI



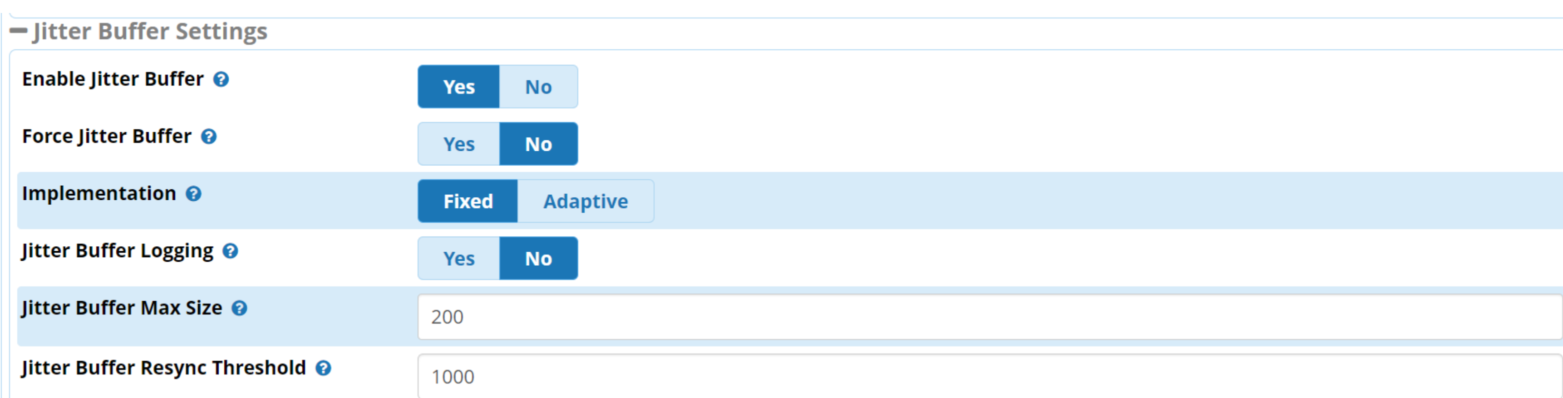
1. MWI Polling Freq - frequency in seconds, within which the MWI status change will be checked (light indication, Message Waiting Indication) and the status will be sent to peers.
2. Notify Ringing - this option allows you to control the state of the subscriber, realizing that his phone is being used (INUSE) by receiving a SIP 180 RINGING packet. Convenient when using BLF functionality.
3. Notify Hold - control of the subscriber and transfer to the INUSE state if the call is on hold (ONHOLD event).

Registration Settings - Registration settings



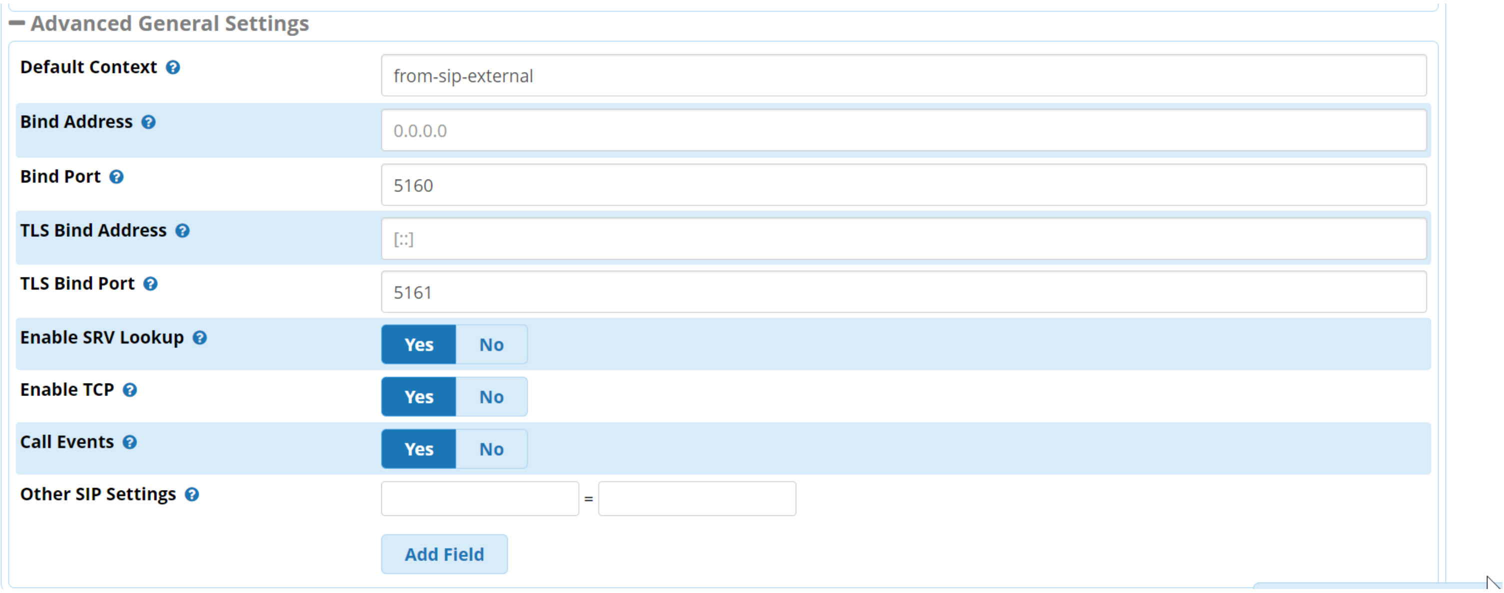
1. Registration Timeout - registration timeout. By default, it is 20 seconds. In other words, every 20 seconds a registration request will be sent until the maximum number of attempts is exceeded.
2. Registration Attempts - the number of registration attempts after which the server will decide to stop sending requests. If set as 0, then the number of requests will not be limited. In a normal situation, a value of 0 is fine - Asterisk will keep sending registration requests until another attempt is successful.
3. Registration Minimum Expiry - the minimum time during which a registration session will be considered expired.
4. Registration Maximum Expiry - the maximum time during which the registration session will be considered expired (for incoming registrations).
5. Registration Default Expiry - default duration of inbound and outbound registrations.

Jitter Buffer Settings - Using Jitter Buffer



Enable Jitter Buffer - this option enables the use of a jitter buffer on the receiving side within one SIP channel.

Advanced General Settings



1. Default Context is the default call handling context unless otherwise specified. MS PBX itself assigns this option as from-sip-external. Make changes only if you fully understand what you are doing.

2. Bind Address - this field specifies the IP address where Asterisk will wait for telephone processing requests, on the port specified in the Bind Port option. If specified as 0.0.0.0, Asterisk will accept requests on all addresses specified in the OS settings. We recommend that you leave this option unchanged. By the way, chan_sip does not support IPv6 for UDP transport. If you specify [::], Asterisk will listen for all IPv4 and all IPv6 addresses.

3. Bind Port - local UDP (and TCP, if enabled in the Enable TCP option) port on which Asterisk listens for chan_SIP calls. If you leave this field blank, then port 5060 will be used by default.

4. TLS Bind Address - TCP port on which Asterisk listens for TLS (secure) calls. Configuration like [::], listens to IPv4 and IPv6 on all interfaces.

5. TLS Bind Port - Local incoming TCP Port that Asterisk will bind to and listen for TLS SIP messages.
   Enable SRV Lookup - See current version of Asterisk for limitations on SRV functionality.

6. Enable TCP – Enable TCP Transport in chan_sip

7. Call Events - Generate manager events when sip ua performs events (e.g. hold).

8. Other SIP Settings - You may set any other SIP settings not present here that are allowed to be configured in the General section of sip.conf. There will be no error checking against these settings so check them carefully.
   They should be entered as:
   [setting] = [value]
   in the boxes below. Click the Add Field box to add additional fields. Blank boxes will be deleted when submitted.

6.4 Music On Hold

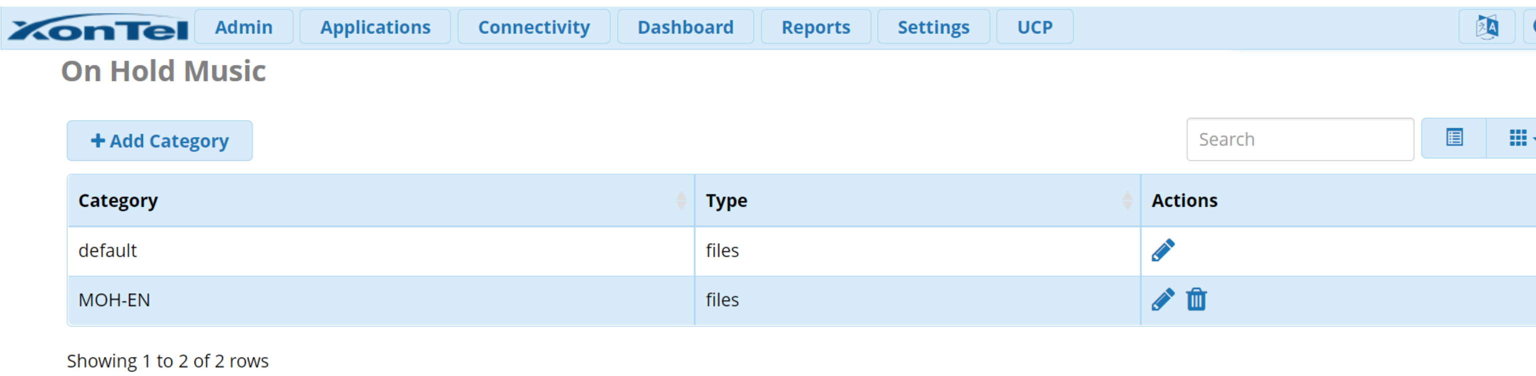
The Music on Hold module is intended to reassure callers that they are still connected to their calls

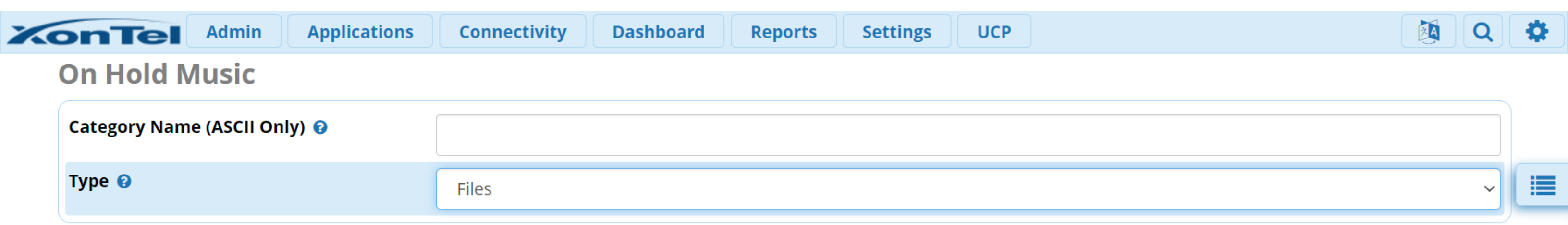
Logging In

- From the top menu click **Settings**
- In the drop down click **Music on Hold**

Adding Music Categories

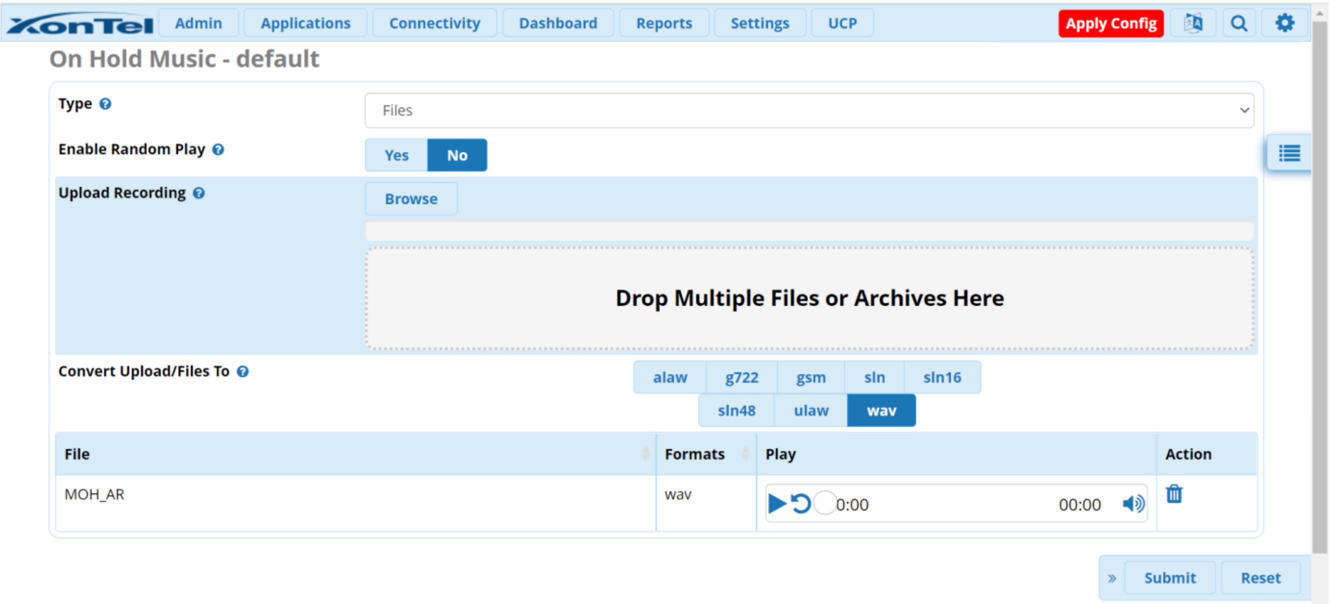
The PBX has one category, which is “default.” You can create additional categories by clicking on the “**Add Category”**





Category Name

Enter a category name, then submit changes to save your new category.



Upload a WAV or MP3 File

Here you can use the file section box to upload a WAV or MP3 from your computer or click **Browse** to upload.

Disable/Enable Random Play

Selecting this option will either toggle random play on or off. Random play picks a random file each time to play when a caller is put in the MoH category. Disabling random play will have each caller start with the first MoH file and work their way down the list in the order in which they are uploaded.

MoH List

A list of MoH files in this category. You can click the red “Delete” button on the right of each file to remove a specific MoH file.

6.5 PIN Sets

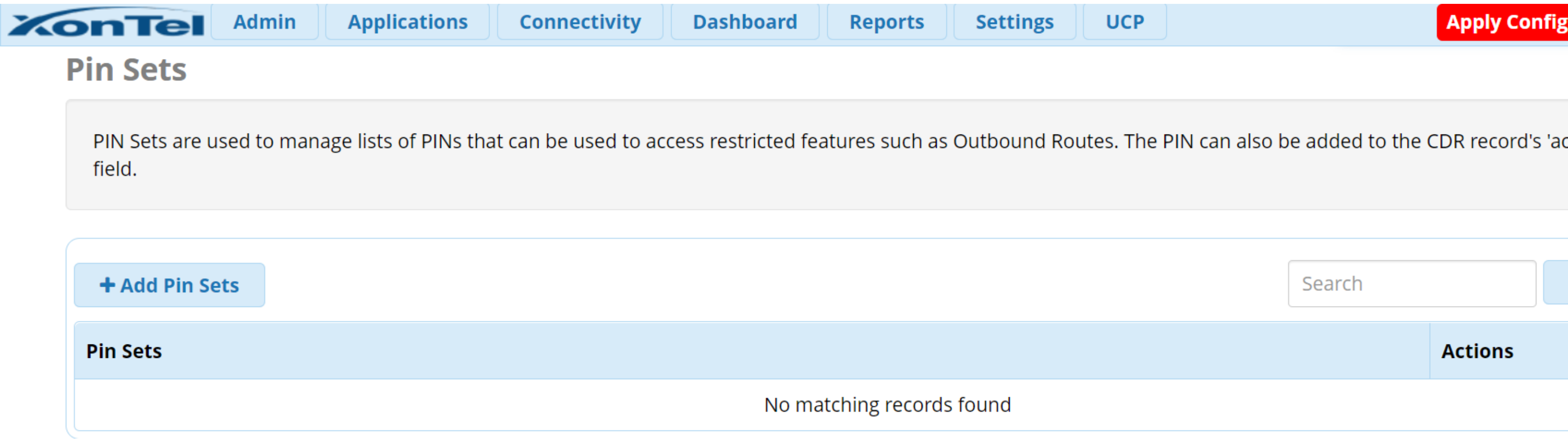
PIN Sets are used to manage lists of PINs (numerical passwords) that can be used to access restricted features such as outbound routes. The PIN can also be added to the CDR record's "accountcode" field.

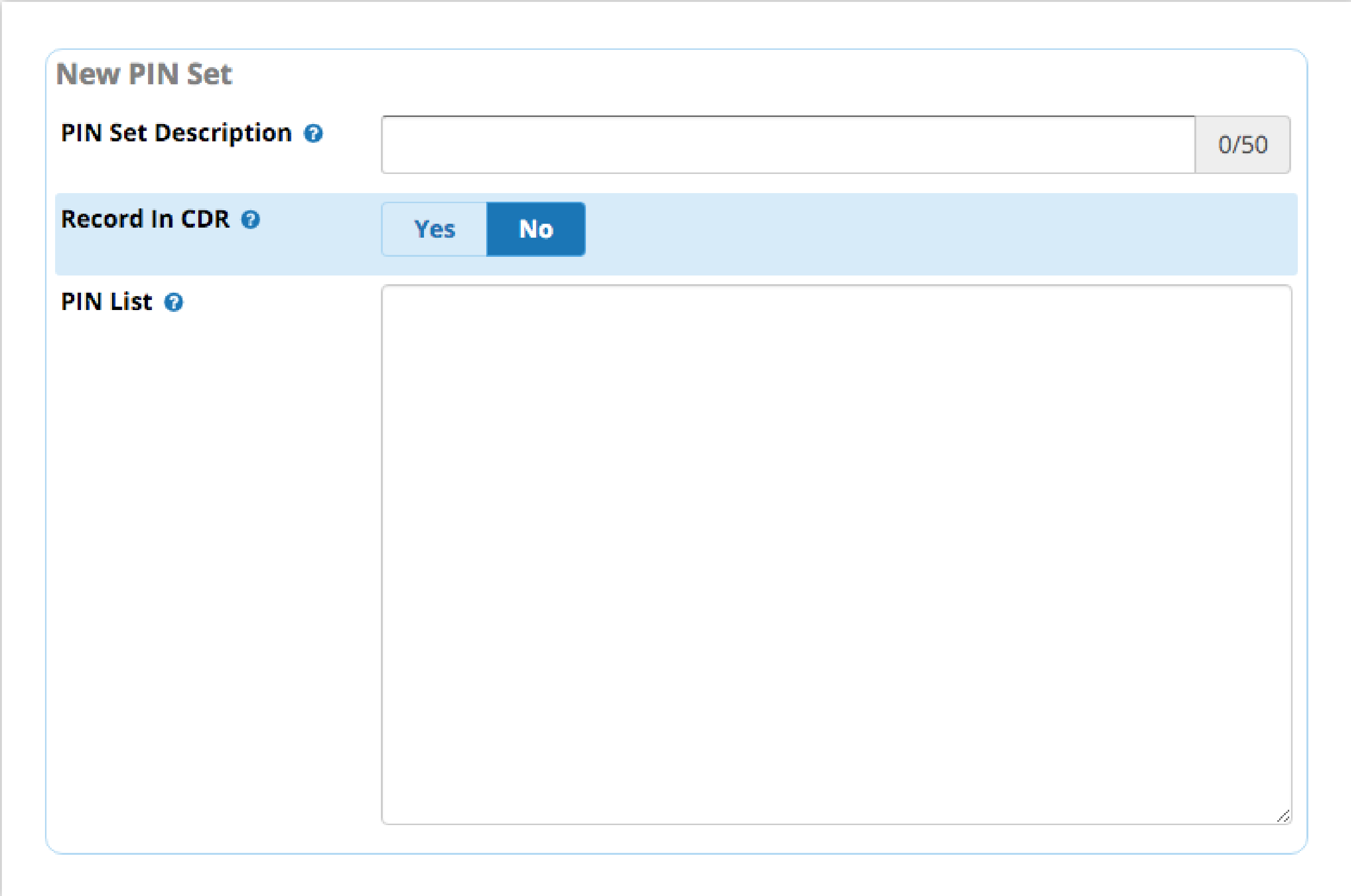
Logging In

- In the top menu go to **Settings**
- In the drop-down menu go to **PIN Sets**

Creating a PIN Set

In the PIN Sets home screen, click the **Add PIN Sets** button.





PIN Set Description

A description of the PIN Set to help you identify it. (Max 50 characters)

Record in CDR

Yes/No: Whether to record the PIN in the call detail records. If **Yes,** PIN numbers dialed will be recorded in CDRs so you can view them in the CDR Reports module.

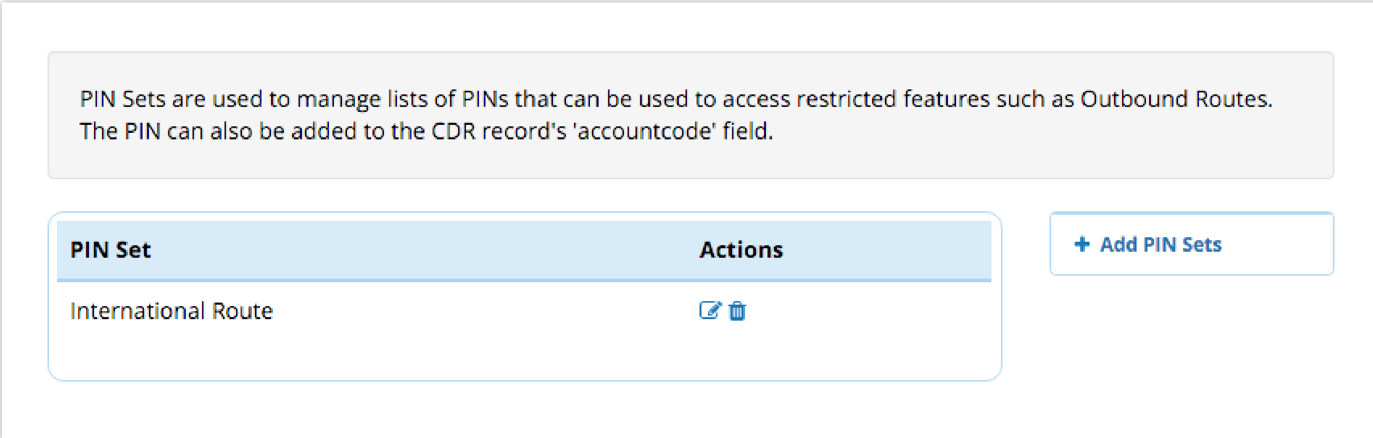
PIN List

A list of one or more PINs. Enter one PIN per line.

Save

Click the **Submit** button, then click the **Apply Config** button.

Your new PIN Set will show up in the list on the module's home screen.



Linking a PIN Set to an Outbound Route

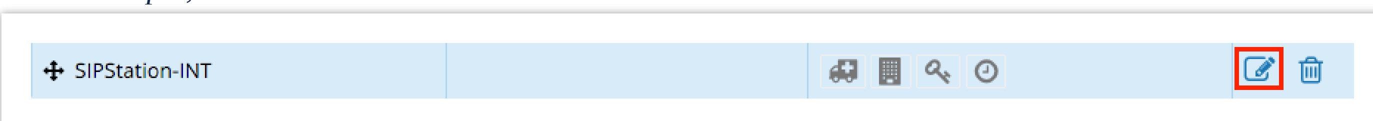
Now that you have created a PIN Set, you can link it to an outbound route.

Navigate to the Outbound Routes module:

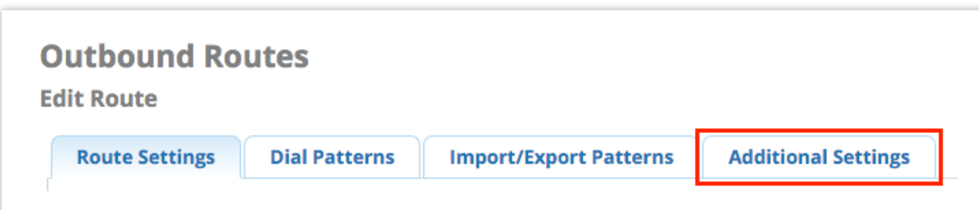
- In the top menu go to **Connectivity**
- In the drop-down menu go to **Outbound Routes**

Click the edit button for the outbound route you wish to edit.

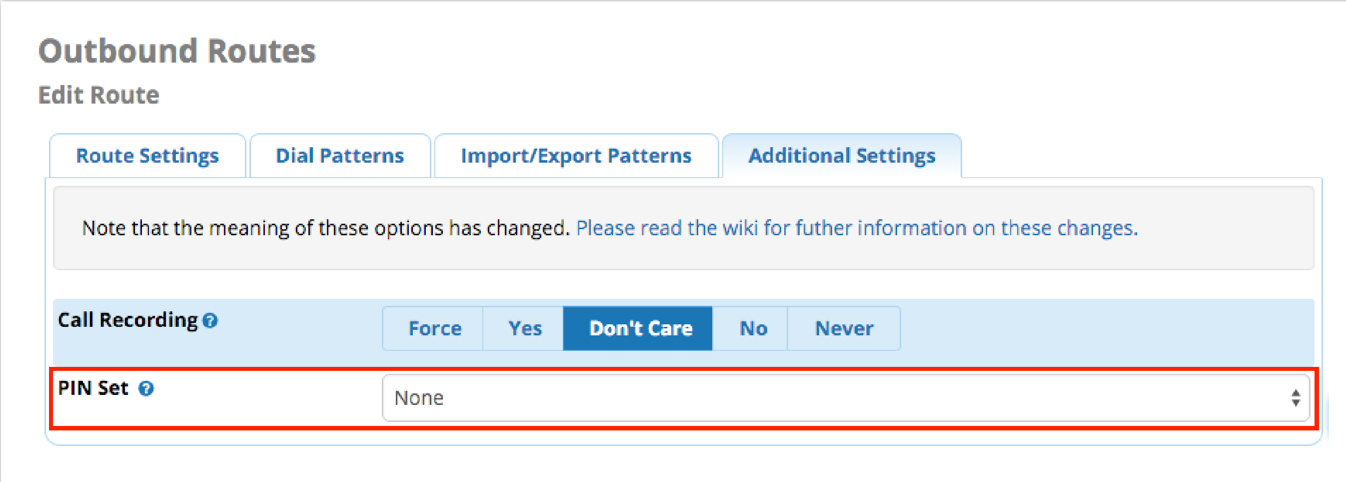
In our example, we will edit our SIPStation International route.



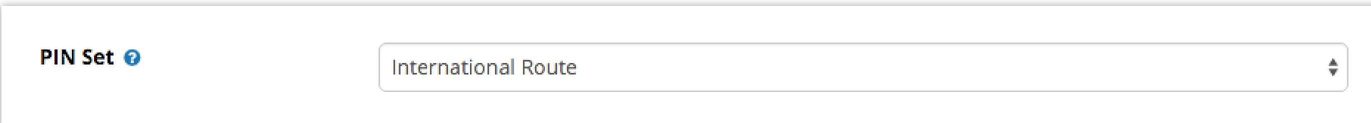
Click the **Additional Settings** tab for the route.



Look for the **PIN Set** option. The default is **None**. Select a PIN Set from the drop-down menu.



In our example, we will choose our International Route PIN Set.



Click the **Submit** button, then click the **Apply Config** button.

Now, any time someone dials out via this outbound route, the system will prompt the user to enter a PIN number. The system will match the entered PIN against this PIN Set. If it does not match, the call will not be completed, and the user will be prompted to enter a PIN again.

Editing a PIN Set

From the module home screen, click the edit button for the PIN Set you wish to edit.

When finished making changes, click the **Submit** button, and then click the **Apply Config** button.

Deleting a PIN Set

From the module home screen, click the trash button for the PIN Set you wish to edit. **Or**, if already viewing the PIN Set, click the **Delete** button.

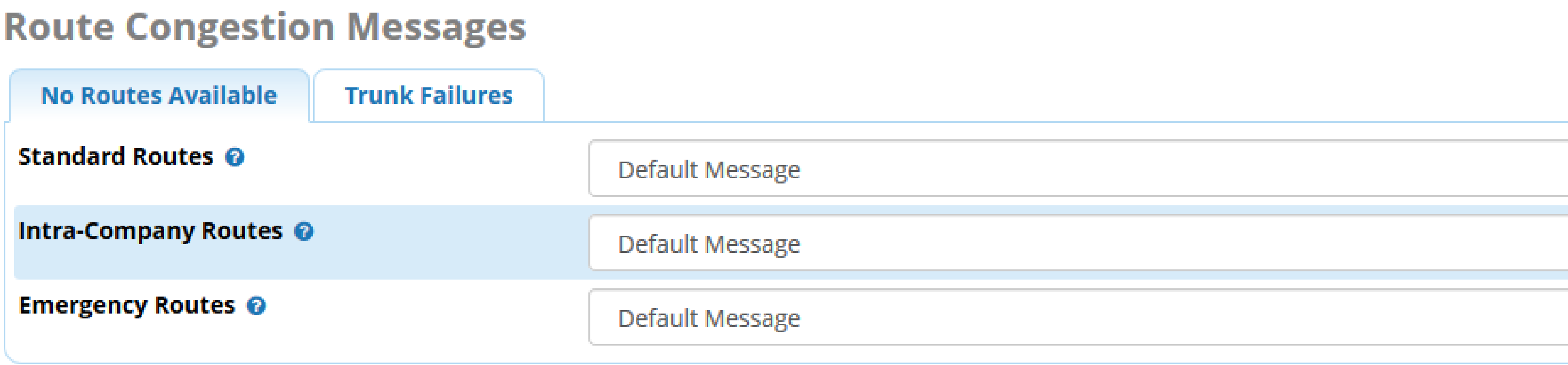
Click the **Apply Config** button.

Note that any outbound routes that were using the deleted PIN Set will no longer use a PIN Set.

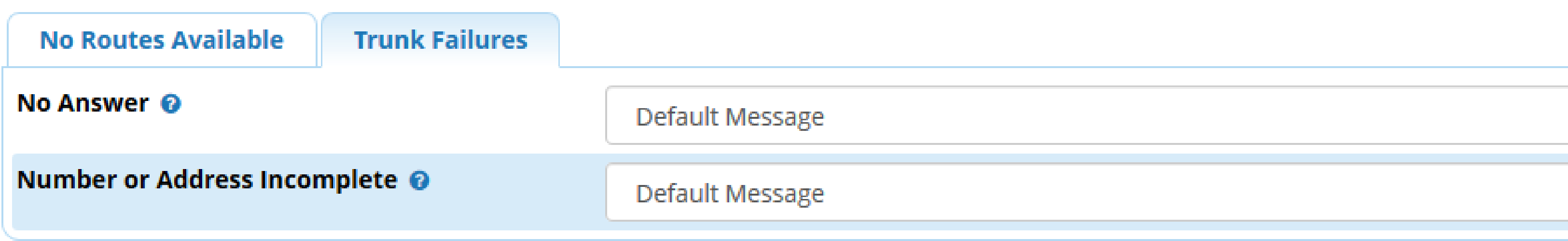
6.6 Route Congestion Messages

Configures message or congestion tones played when all trunks are busy in a route. Allows different messages for Emergency Routes and Intra-Company Routes.

No Routes Available



Trunk Failures



6.7 Voicemail Admin

The "Voicemail Admin Module" allows you to make changes to the default voicemail settings used by the system, and to make changes to voicemail settings for each individual user.

It also allows you to view statistics about the voicemail system on a global or per user basis, and to delete files relating to voicemail on a global or per user basis.

It also allows you to define timezone definitions for use by the voicemail system.

How is the Voicemail Admin Module related to the other Modules?

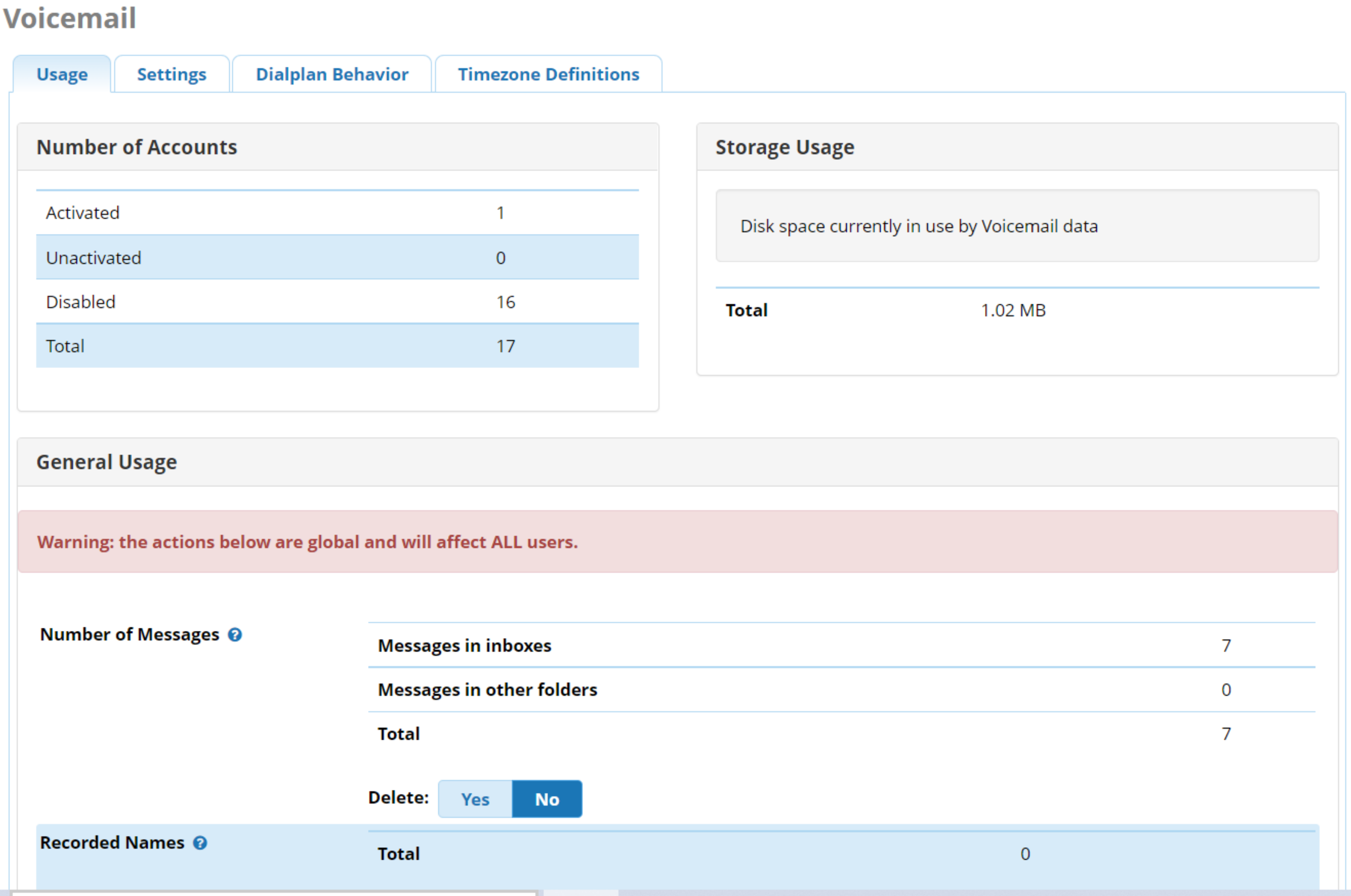
The "Voicemail Admin Module" is related to the Extensions Module, because it sets certain defaults that are used in the Voicemail section of the Extensions Module Module, and because it allows you to set individual settings and view individual statistics for extensions that have been defined on your system in the Extensions Module. If you make a change to a setting for a particular extension in this module, that change will appear in the Voicemail section of the Extensions Module module for that extension, either in the pre-set fields or in the "VM Options" field.

How Do I Get to the Voicemail Admin Module?

- From the top menu click **Settings**
- From the drop down click **Voicemail Admin**

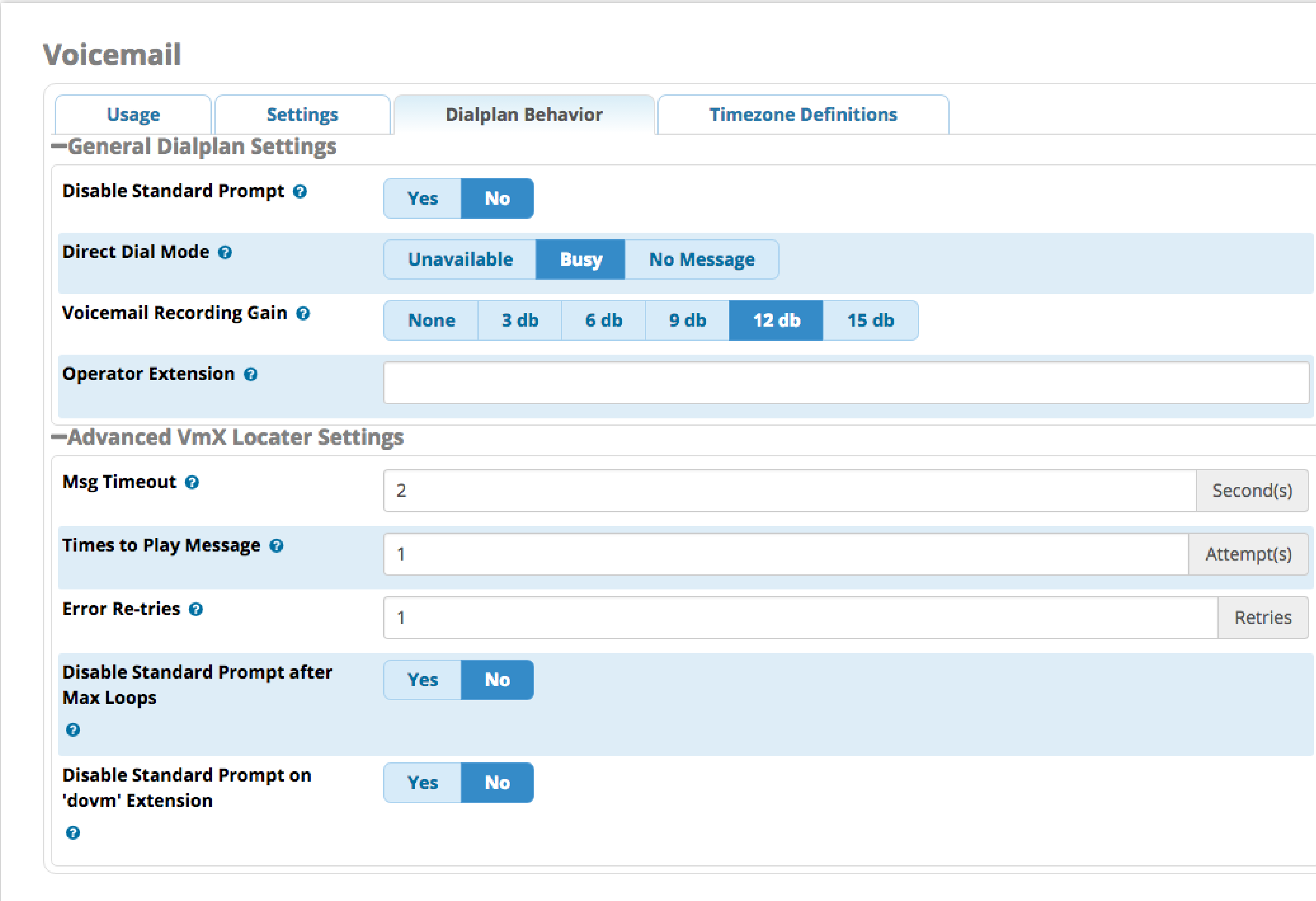
1. **Usage**

From here you check voicemail usage in the MS PBX



Number of Messages	**Messages in inboxes**	7
	Messages in other folders	0
	Total	7
	Delete: Yes No	
Recorded Names	**Total**	0
	Delete: Yes No	
Unavailable Greetings	**Total**	0
	Delete: Yes No	
Busy Greetings	**Total**	0
	Delete: Yes No	
Temporary Greetings	**Total**	0
	Delete: Yes No	
Abandoned Greetings	**Total**	0
	Delete: Yes No	

2. Dialplan Behavior



General Dialplan Settings

Disable Standard Prompt

Yes/No - Whether to disable the standard voicemail instructions that follow the user-recorded message. These standard instructions tell the caller to leave a message after the beep. This can be individually controlled for users who have VMX locater enabled.

Direct Dial Mode

Unavailable/Busy/No Message - Whether to play the busy message, the unavailable message, or no message when direct dialing voicemail.

Voicemail Recording Gain

The amount of gain to amplify a voicemail message when being recorded. You may need to increase this if users are complaining about messages on your system being hard to hear, which is often caused by very quiet analog lines. The gain is in Decibels, which doubles the volume for every 3 db.

Operator Extension

Default number to dial when a voicemail user "zeros out" (if enabled) - in other words, the user will be sent to this destination when they press "0" from within the voicemail system. This destination can be any number, including an external number. There is NO VALIDATION, so it should be tested after configuration. Note: This default will be overridden by any VMX Locater "0" option you set for an extension. The VMX Locater option is used even when VMX Locater is not enabled.

Advanced VmX Locater Settings

Msg Timeout

Time to wait after message has played to timeout and/or repeat the message if no entry pressed. Default = 2 seconds.

Times to Play Message

Number of times to play the recorded message if the caller does not press any options and it times out. One attempt means we won't repeat it, and it will be treated as a timeout. A timeout would be the normal behavior. It is fairly common to leave this at zero and play a message telling callers to press an option, letting them know that they will otherwise go to voicemail.

Error Re-tries

Number of times to play invalid options and repeat the message upon receiving an undefined option. One retry means it will repeat at one time after the intial failure.

Disable Standard Prompt after Max Loops

If the Max Loops are reached and the call goes to voicemail, setting this to **Yes** will disable the standard voicemail prompt that follows the user's recorded greeting. This default can be overriden with a unique *..vmx/vmxopts/loops AstDB* entry for the given mode (busy/unavail) and user.

Disable Standard Prompt on 'dovm' Extension

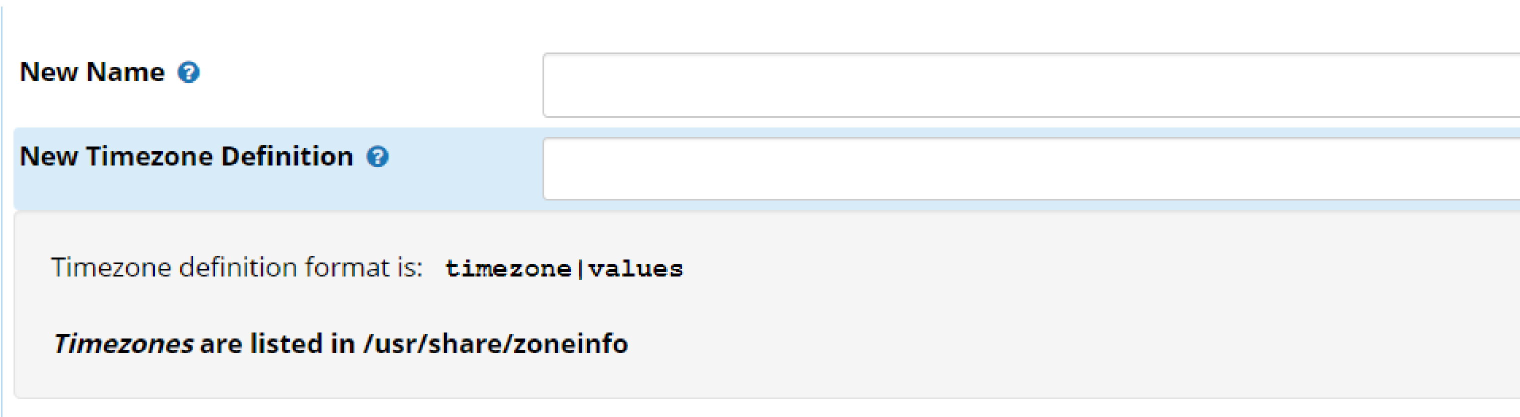
If the special advanced extension of 'dovm' is used, setting this to **Yes** will disable the standard voicemail prompt prompt that follows the user's recorded greeting. This default can be overridden with a unique *..vmx/vmxopts/dovm AstDB* entry for the given mode (busy/unavail) and user.

3. Timezone Definitions

A timezone definition specifies how the Voicemail system announces the time. For example, the time a message was left will be announced according to the user's timezone on message playback.

Entries below will be written to Voicemail configuration as-is.

Please be sure to follow the format for timezone definitions described below.



New Name: Timezone definition name

New Timezone Definition: Time announcement for message playback

The *values* supported in the timezone definition string include:	
'filename'	The name of a sound file (the file name must be single-quoted)
variable	A variable to be substituted (see below for supported variable values)
Supported *variables*:	
A or a	Day of week (Saturday, Sunday, ...)
B or b or h	Month name (January, February, ...)
d or e	numeric day of month (first, second, ..., thirty-first)
Y	Year
I or l	Hour, 12 hour clock
H	Hour, 24 hour clock (single digit hours preceded by "oh")
k	Hour, 24 hour clock (single digit hours NOT preceded by "oh")
M	Minute, with 00 pronounced as "o'clock"
N	Minute, with 00 pronounced as "hundred" (US military time)
P or p	AM or PM
Q	"today", "yesterday" or ABdY
q	"" (for today), "yesterday", weekday, or ABdY
R	24 hour time, including minute