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1. Overview & Features

Adore IPPBX System User Guide

PBX System – A Private Branch Exchange

Adore Infotech offers an outstanding array of voip solutions. Adore IPPBX System is one such proven voip product that is easy to set up and maintain. It has literally replaced the conventional PBX or phone system, and provides the extension number facility through which one can transfer, conference or make calls in that system.

The voices are sent through the data packets on a data network. It has eliminated the use of traditional phone network in the corporate houses. Adore IP-PBX is the complete PBX software which is easy to install, configure and very economical as well as possess many additional functionality.

Adore IPPBX System : System Specification

AdoreInfotech recommends following Hardware and operating system.
Hardware Requirements: Intel Core i5 Processor /16-32 Gb RAM/1 TB HDD

Software Requirements:

- Linux CENTOS 7.x (complete installation)
- Yum Server

Internet connection:

The use of a 1Gbit Ethernet card is a prerequisite for Adore system with good broadband internet connection .

Pre-installation Considerations:

Here is some information that we think is worth knowing prior to installing the Adore IPPBX System.

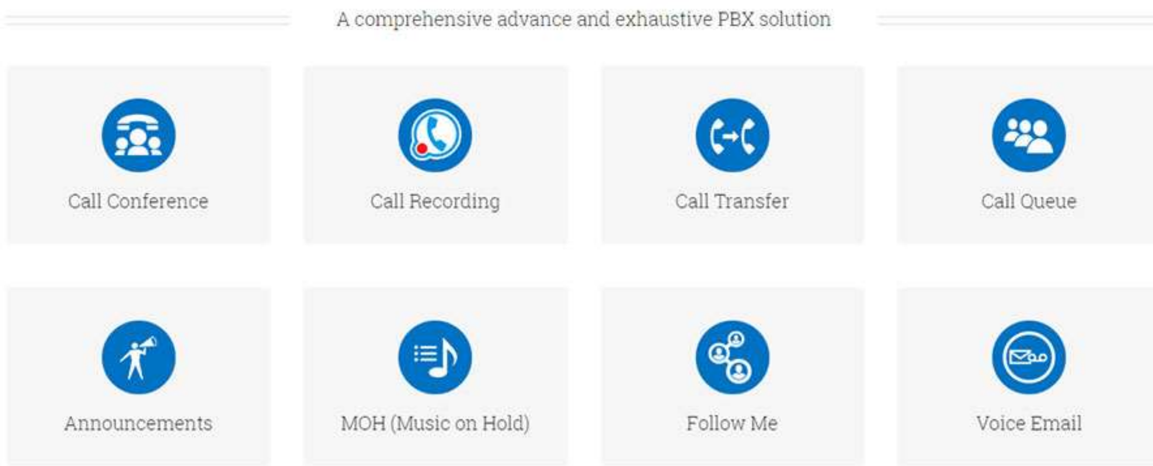
- Linux server should be on public IP.
- SSL Certificate should be install on the Linux server

Carrier:

Carrier which supports SIP calls, g711, g723 and g729 codecs.

Installation:

AdoreInfotech requires remote access of server for installation. The installation will be done by SSH connection on Linux server.Its installation requires internet facility and the time taken for its set up mainly depends on the nature and amount of customization to be done and the nature of service provider’s infrastructure. Our competent and reliable force of engineers renders spectacular services in solving every bit of installation related problem.



Advantages Of IPPBX:

There are various key benefits for utilizing a PBX versus phone service or obtaining an office phone system.

- **Preparatory Expense Savings :** You don't have to make an extensive forthright venture by acquiring an office phone framework and there is no compelling reason to look after it. With a PBX arrangement the workplace phone framework is worked and kept up by you're supplier.
- **Unremitting Expense Savings :** Typically, a PBX execution is fundamentally more financially savvy as time goes on than a conventional PBX. These progressing reserve funds are not with standing the beginning investment funds on capital cost.
- **Reliability :** For little organizations a virtual framework can give them moment validity by giving their clients an expert sounding phone arrangement. Conventional phone frameworks are by and large cost-restrictive for littler associations.
- **Measurability:** Traditional telephone frameworks are constrained by what number of clients they can deal with and additionally what number of lines you have acquired from your neighborhood phone supplier. Virtual frameworks can extend and develop with your business as you need them to.

- **Instant Setup :** A PBX arrangement can have you up and running immediately contrasted with a customary phone framework.
- **Lack of Difficulty for Use :** Customer management of a PBX is significantly more easy to use than is the situation with conventional arrangements. Natural web interfaces imply that anybody can oversee, screen and roll out improvements to the administration whenever, without requiring any particular ability.

Combined with Adore Infotech reasonable & straightway support, this PBX permits customers to convey bother free and dependable IP-based telecom administrations in a fast and consistent way with insignificant interruption for having a low resilience in downtime that adds more indicates its expense viability.

Those Are Just A Few Main Features, There Are Plenty More

User Features

- View User
- Add / Import User

Account Features

- SIP Settings
- Trunk Settings

Call Plan & Dial Plan Features

- Dial Plans
- Out Going Call Rules
- Incoming Call Rules

PBX Features

- Call Conference
- Call Recording
- Call Transfer
- Call Queue
- Announcements
- Ring Groups
- MOH (Music on Hold)
- Follow Me
- IVR (Interactive Voice Response)
- Voice Email
- Time Condition
- Time Groups

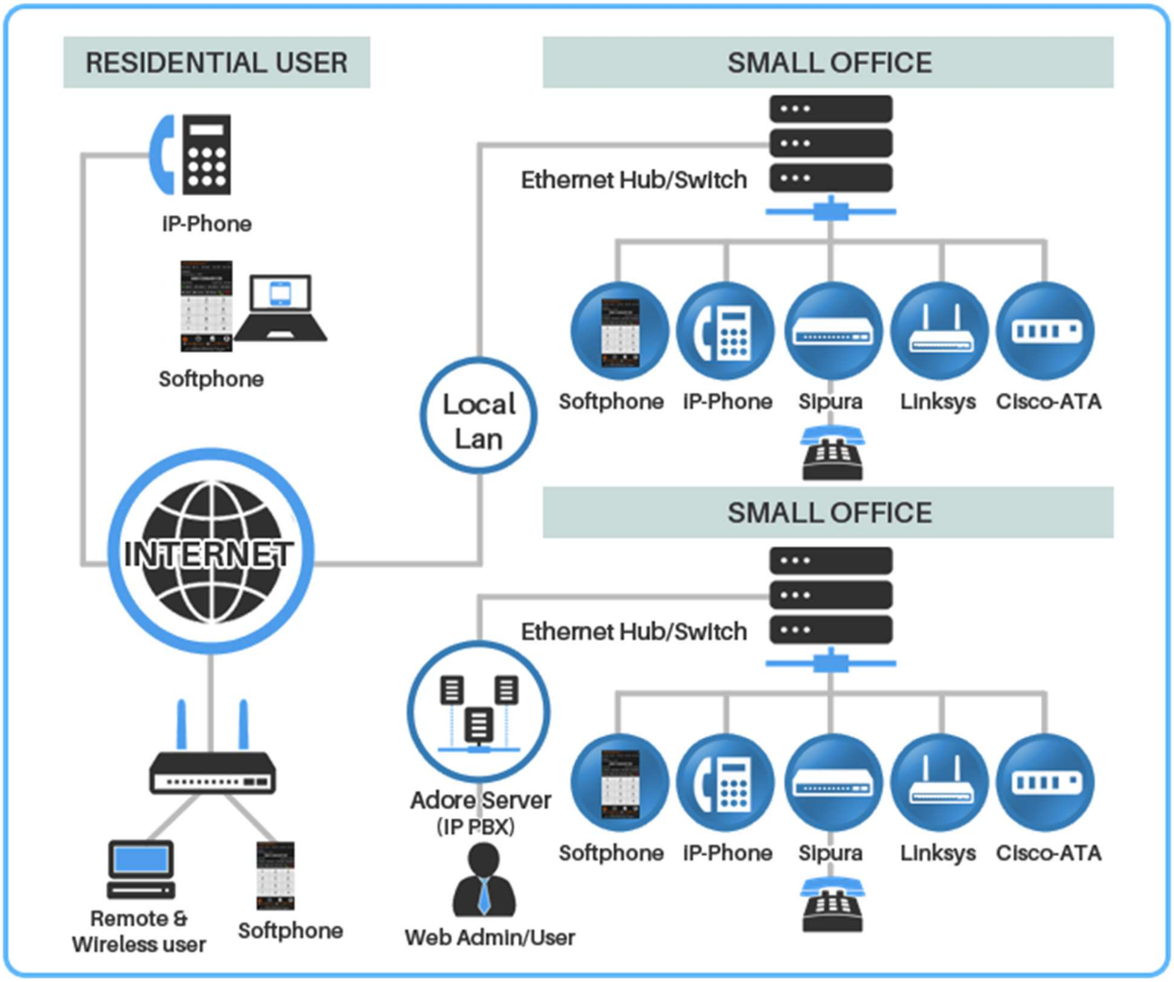
Call Status & Reports

- CDR Reports

System Features

- System Status

Network Diagram of Adore IPPBX System



2. Admin Panel

Adore IPPBX System

PBX (Private Branch Exchange) is a phone exchange or exchanging framework that serves a private association and performs grouping of focal office lines or trunks and gives intercommunication between an expansive numbers of phone stations in the organization. Generally, a mini business PBX is a phone exchanging framework that oversees approaching and active requires an organization's inner clients. A PBX System is associated with general society telephone framework and naturally courses approaching calls to particular expansions. It additionally shares and deals with different lines. A normal miniature business PBX framework incorporates outside and inner telephone lines, a PC server that oversees call exchanging and directing and a console for manual control.

Admin Module

- **How to Login Admin Module**
- **Admin Dashboard**
- **Administrator**
- **Blacklist**
- **Contact Manager**
- **Custom Destination**
- **Custom Extension**
- **Feature Codes**
- **Sound Languages**
- **System Recordings**
- **User Management**

2.1. Login on Admin Module

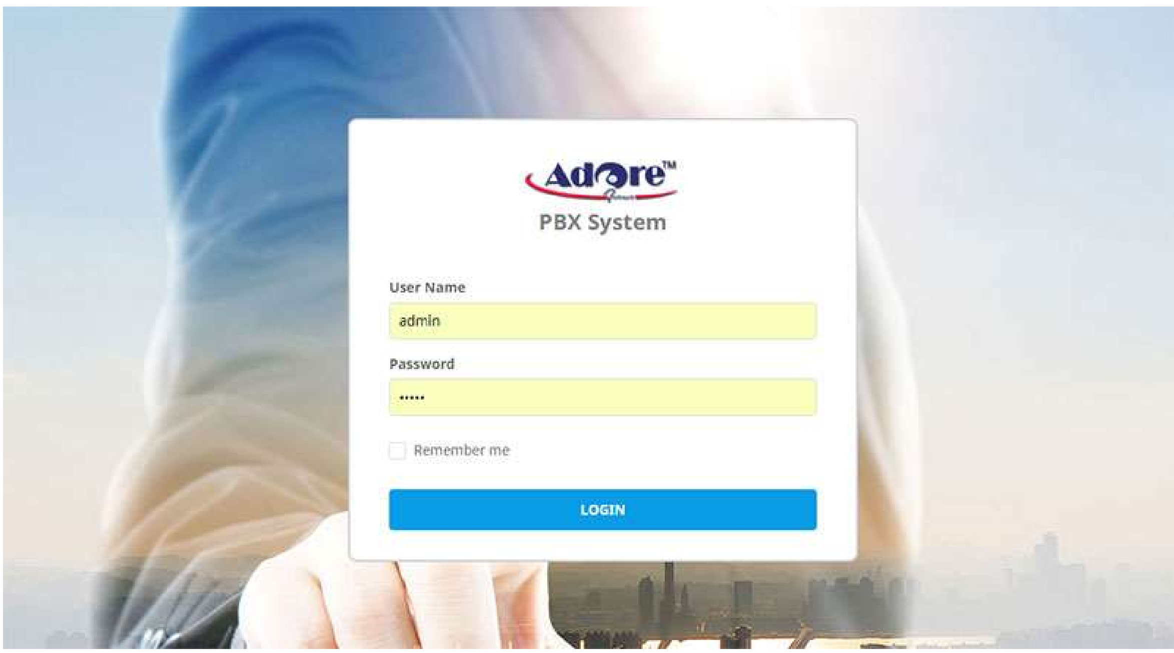
Login on Admin Module

Please visit following URL : **http://pbx.adoreinfotech.co.in/admin**

Enter the user name and password in the appropriate box, and click Login button.

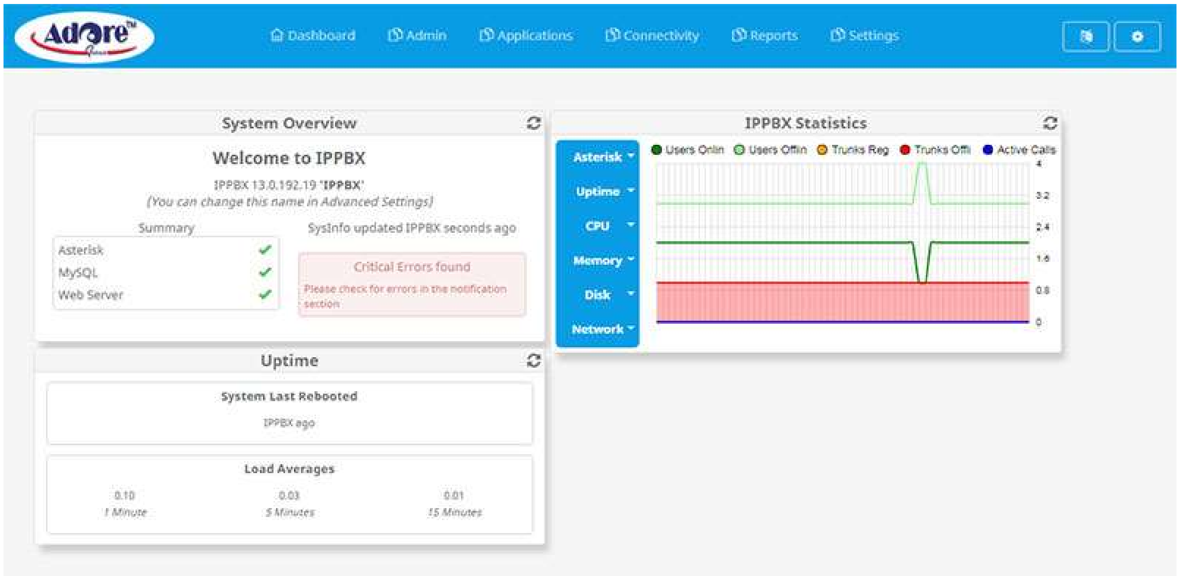
User Name : admin

Password: admin



2.2. Admin Dashboard

Admin Dashboard

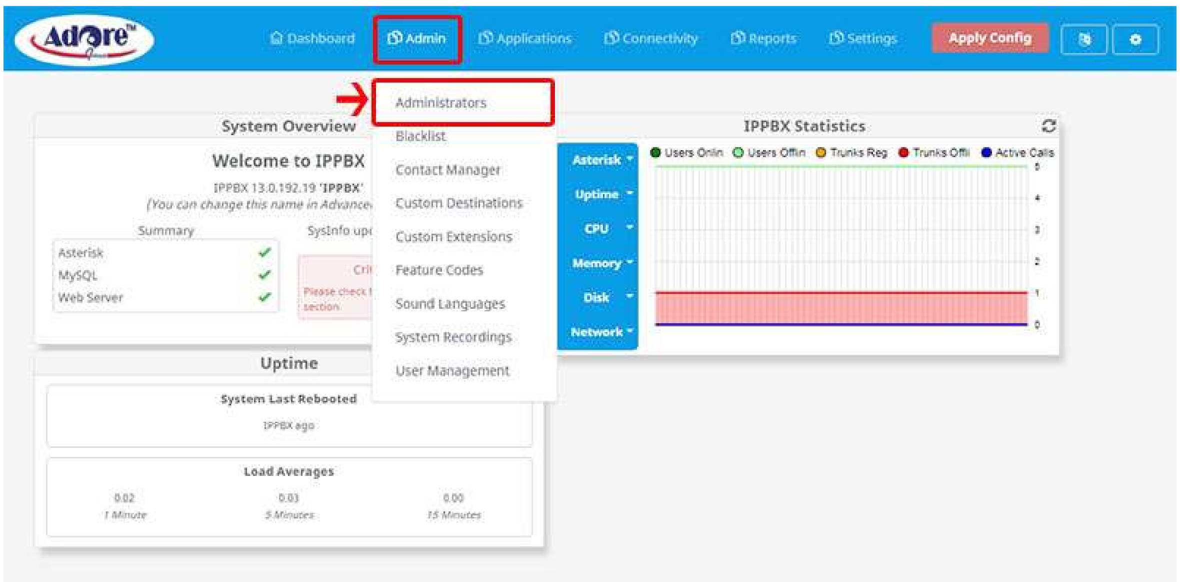


2.3. Administrator

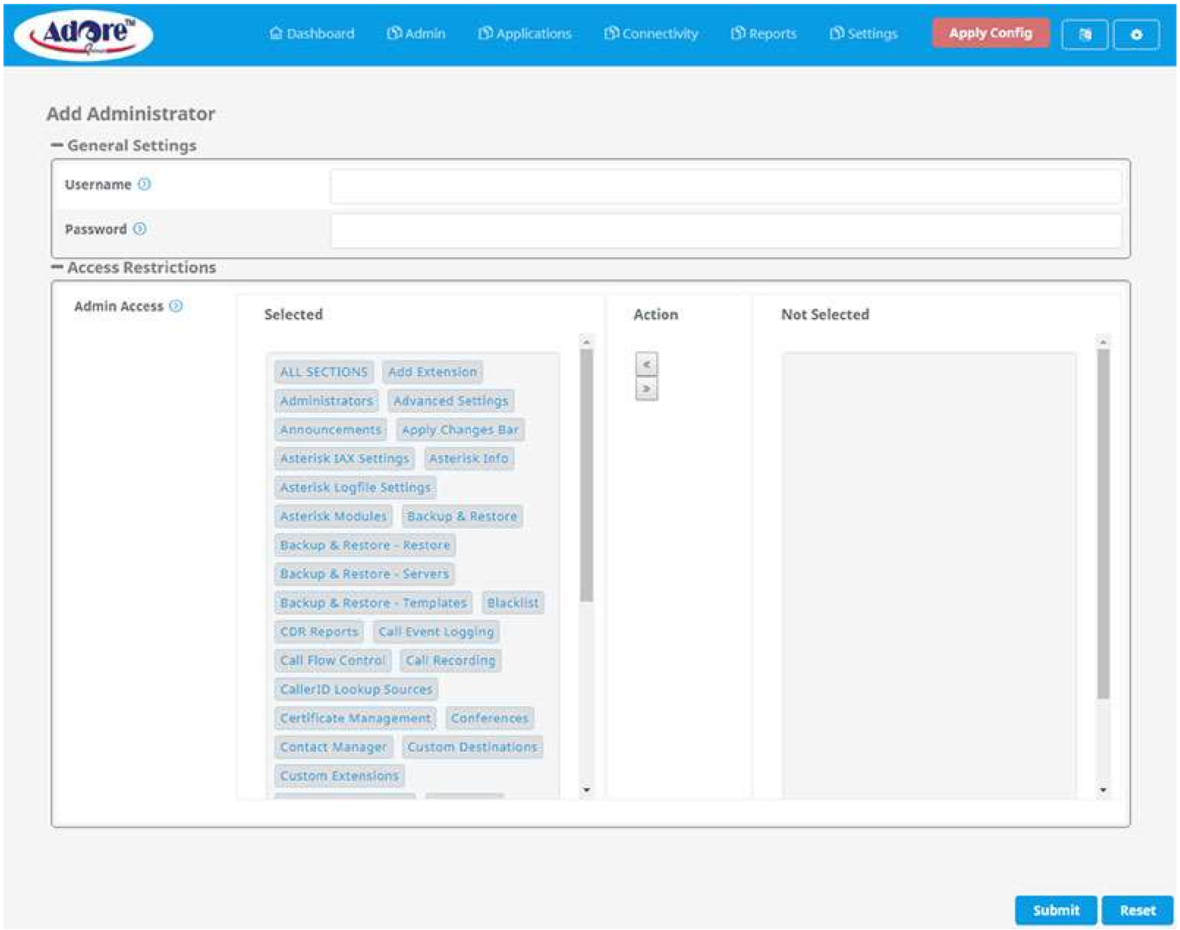
Administrators Module

The Administrator's Module allows you to set up additional users and passwords for the PBX Admin. You can then pick which PBX system modules the additional users can access. Additionally, you can give users access to all modules with a single click.

Go to **Admin - > Administrators**



On Click Administrator following screen will appear.



You can add a user from the landing page seen above. Or, if you are in a current user page, you can click the "Add User" Link at the top of the user list.

Username

This is what the user will log in with. Usernames cannot have spaces and must be unique. Usernames such as "admin," "administrator," and "maint" are pretty common and probably should be avoided. Use something that you will remember. You may try your company name, or company name with "admin" at the end.

Password

Define the password for the user. The password cannot contain spaces. The password should be over 8 characters and contain a mix of upper and lower case letters and numbers. We do not have these minimum requirements programmed in, but you should always use good password practices when creating a password.

Admin Access

Here you can select which modules the user is allowed to see and manage. Drag modules from the **Not Selected** bin to the **Selected** bin.

Save User:

Click the **Submit** button. You may now log out. Then, try to log in as the new user and make sure everything works as expected.

NOTE

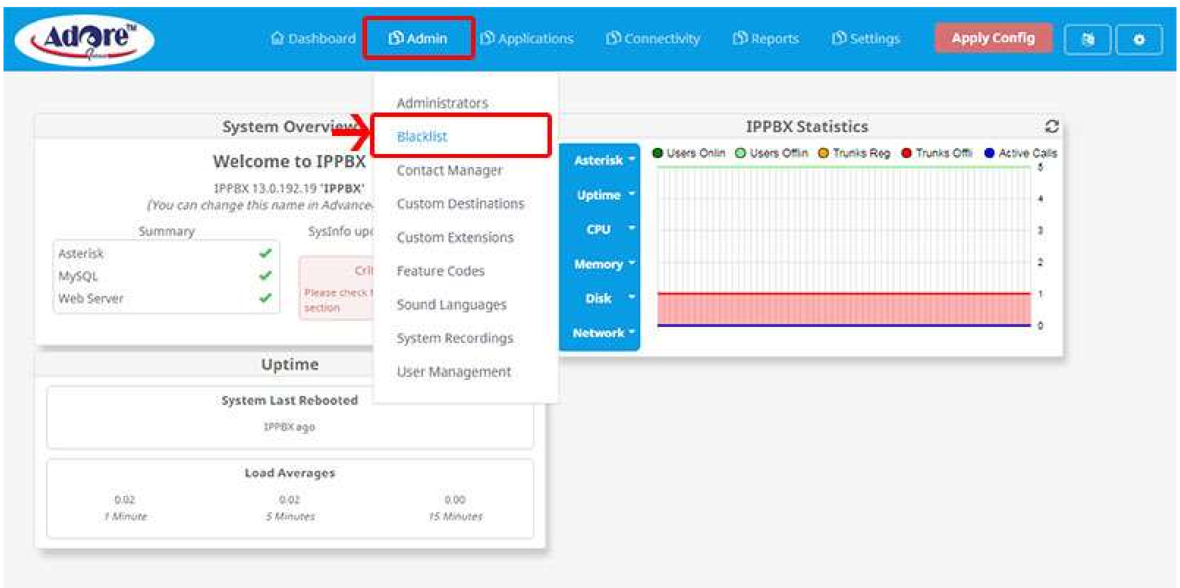
Don't forget to give them access to "Apply Changes Bar," or they won't ever get the Apply Config button after saving changes, and their changes will never take effect.

2.4. Blacklist

Blacklist

The Blacklist module allows you to have a list of numbers that will be blacklisted by the PBX. If a caller calls from one of those phone numbers, they will be sent to a destination you choose. If you do not choose a destination, they will hear the message "The number you have dialed is not in service. Please check the number and try again."

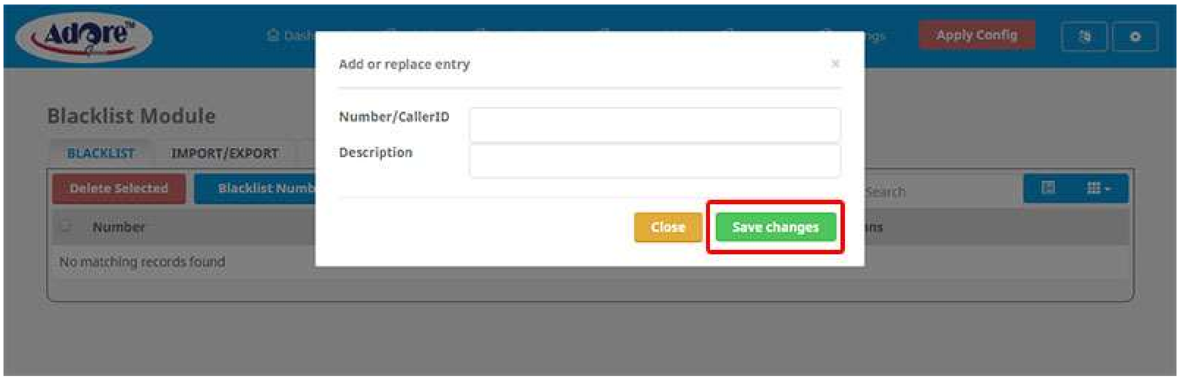
Go to **Admin** -> **Blacklist**



On click **Blacklist** following screen will appear.

Blacklisted a Number

Make sure Blacklist is the active tab. Click the Blacklist Number button. A window will pop up where you can enter a number and description.



To blacklist a number, simply provide the following information:

Number

Define the 10-digit number that you want to blacklist.

Description

Here we give this entry a name such as “Ex Stalker Employee John.”

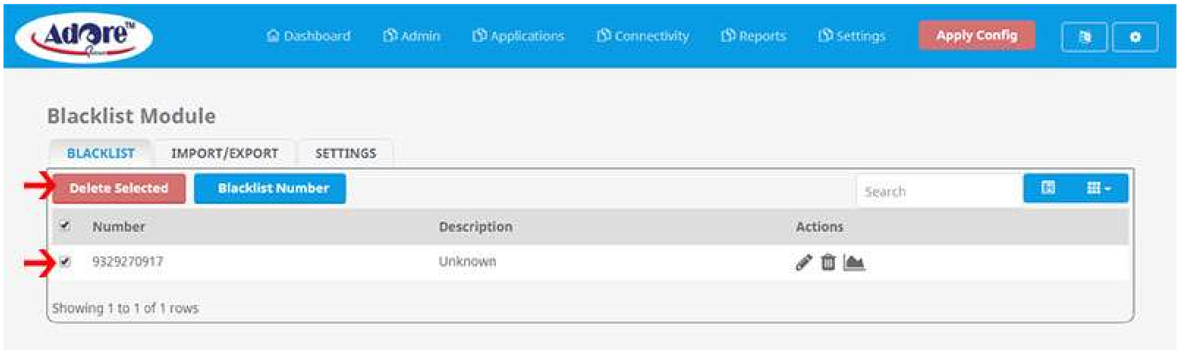
Save Changes

Once complete click the Save Changes button. You'll receive confirmation that the number was added. Then, you can click the Close button.

Removing a Number from the Blacklist

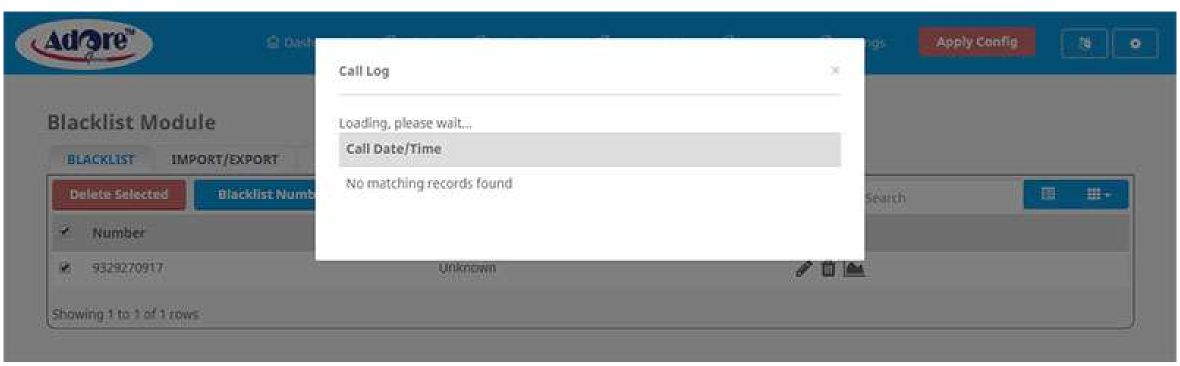
In the Blacklist tab, check the checkboxes for the entries you wish to delete.

Click the Delete Selected button to delete the selected entries. Click OK to confirm the deletion.



Alternatively, you can click the trash icon for an individual entry to delete it. Click OK to confirm the deletion.

Click the statistics button to view a list of inbound calls from the blacklisted Caller ID. Note these are not necessarily all "blocked" calls, since the list will include any calls received prior to when you blacklisted the number.



Importing or Exporting a Blacklist in CSV File Format

Click the **Import/Export** tab.

The required CSV format is shown.

To import a CSV file, click the Import from CSV button, select a file from your computer, and click the Upload button.

To export a CSV file, click the Export to CSV button.

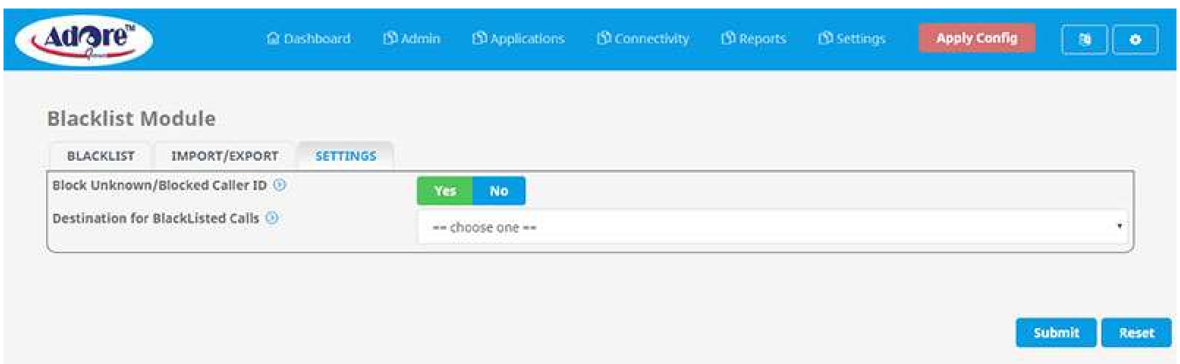
Blocking Unknown Caller ID Calls

You can also optionally decide to block all calls that come to your system with a blocked Caller ID. To do this:

Click on the **Settings** tab within the **Blacklist module**.

Click the **Yes** button next to “**Block Unknown/Blocked Caller ID**.”

Choose a destination. If you do not choose a destination, callers will hear the "**not in service**" message.



Click the **Submit** button.

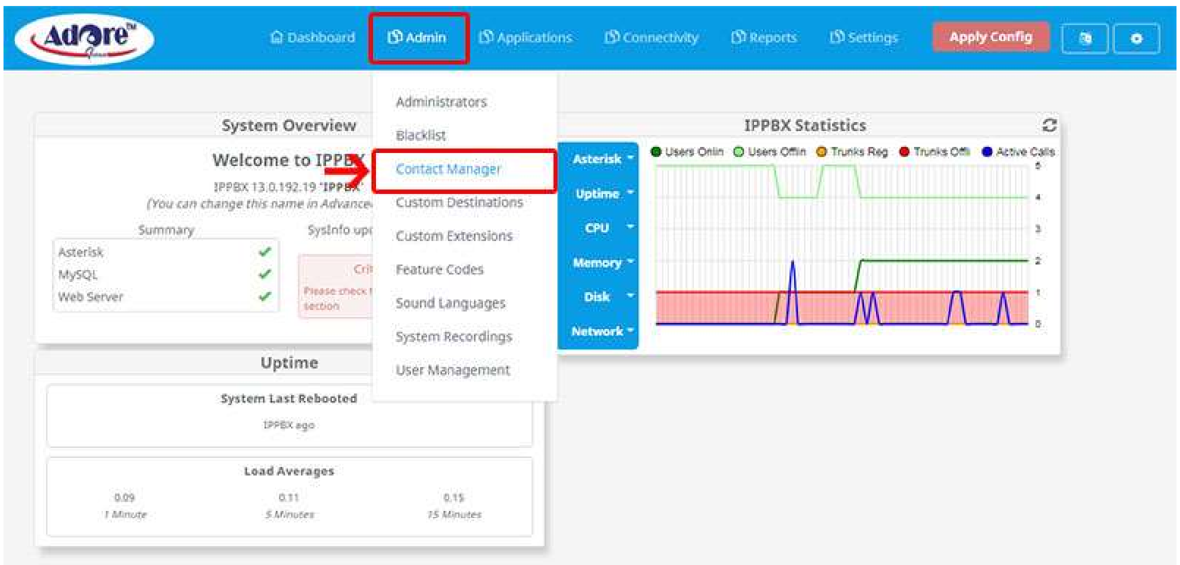
Here you can add Blacklist number manually by using "Blacklist Number".

2.5. Contact Manager

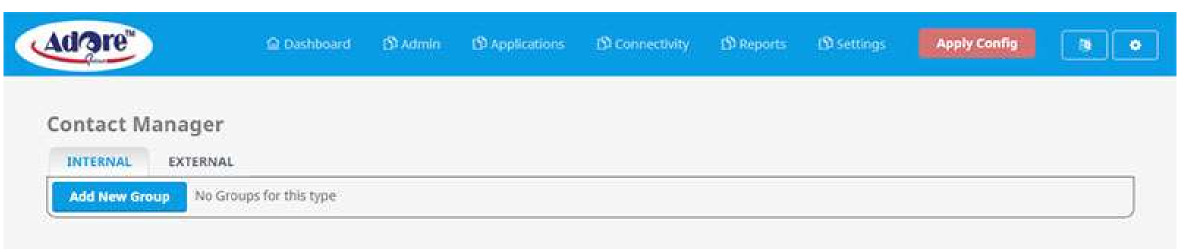
Contact Manager

The Contact Manager module lets you add contacts to groups. Contacts can be grouped under Internal or External categories. A default group called "**User Manager**" contains all users and cannot be deleted.

Go to **Admin** → **Contact Manager** to visit the module.



On click **Contact Manager** following screen will appear

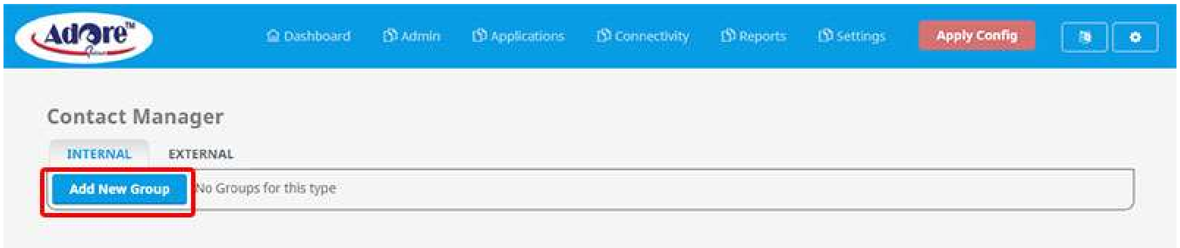


Contact Group Types

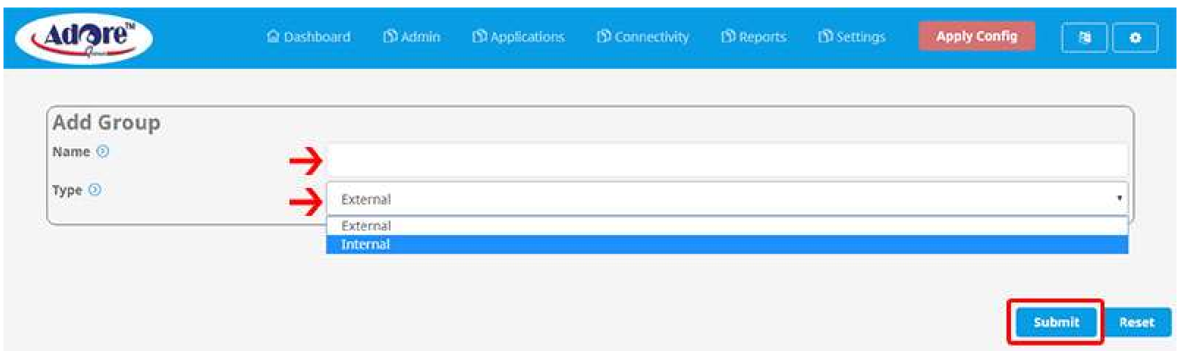
- **Internal Groups**- These are contact groups of internal extensions. The Contact information is pulled from the User Management module.
- **External Groups**- These are contact groups of extension contacts such as customers and vendors.

Adding a New Group

Click **Add New Group** button.



Enter the group a **Name** and select which **Type** of group to create. and click **submit** button.

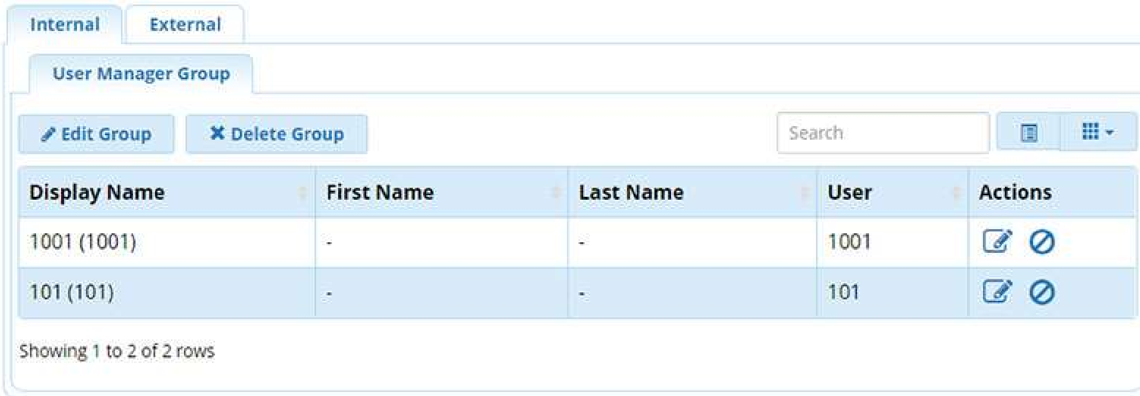


Internal Groups

Under the **Internal Groups** tab, you should see your already-created Internal groups. Each internal group will have its own tab.

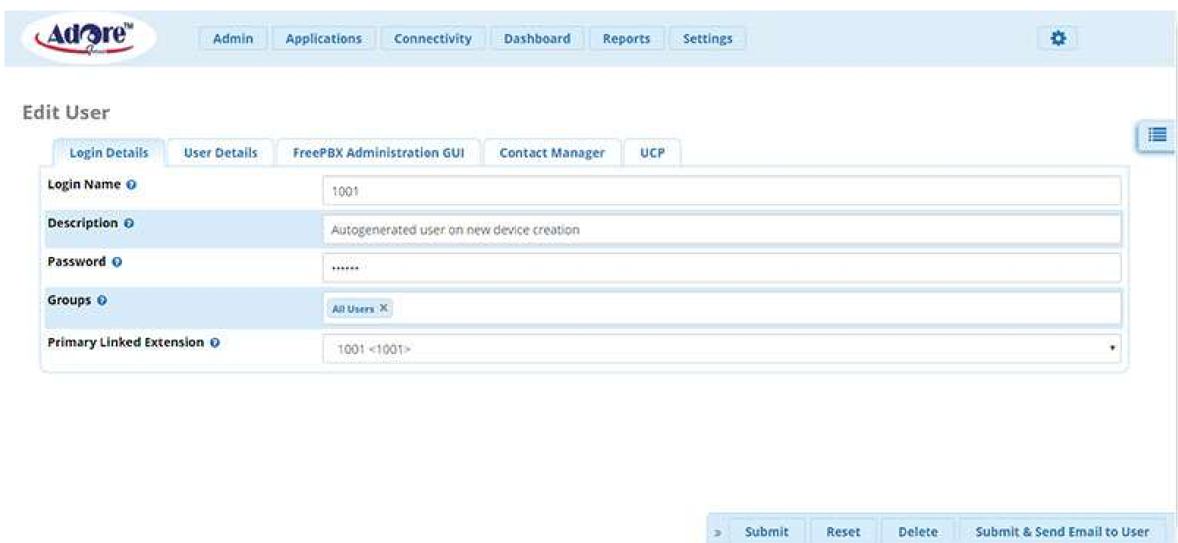
In this example, we have an internal group called "Employees."

From here you can view all users in this group, as well as modify the group name if desired, edit or delete users, and add entries.



Editing an Internal User:

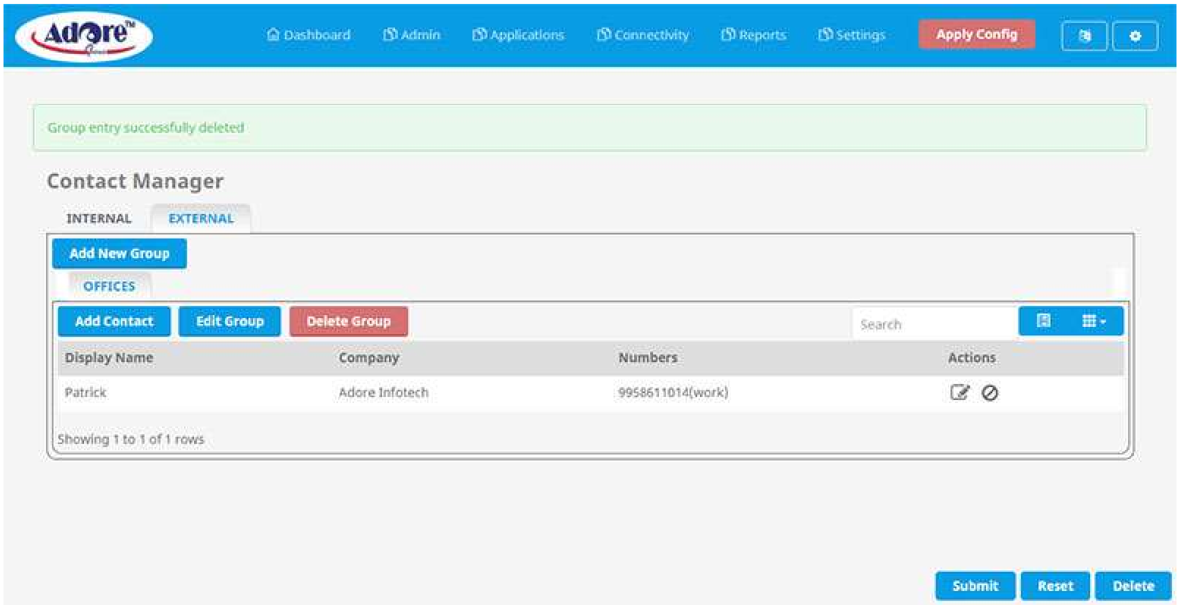
- Click the edit button () next to an entry.
- The default information for an internal user is pulled from the User Management module. This information is shown in light gray text. You can override the default info by typing into the fields. Anything you enter here will show in dark text and override what is in the User Management module (for the purposes of the Contact Manager module only). The User Management module will retain the original information.
- Click the **Submit** button when finished editing.



External Groups

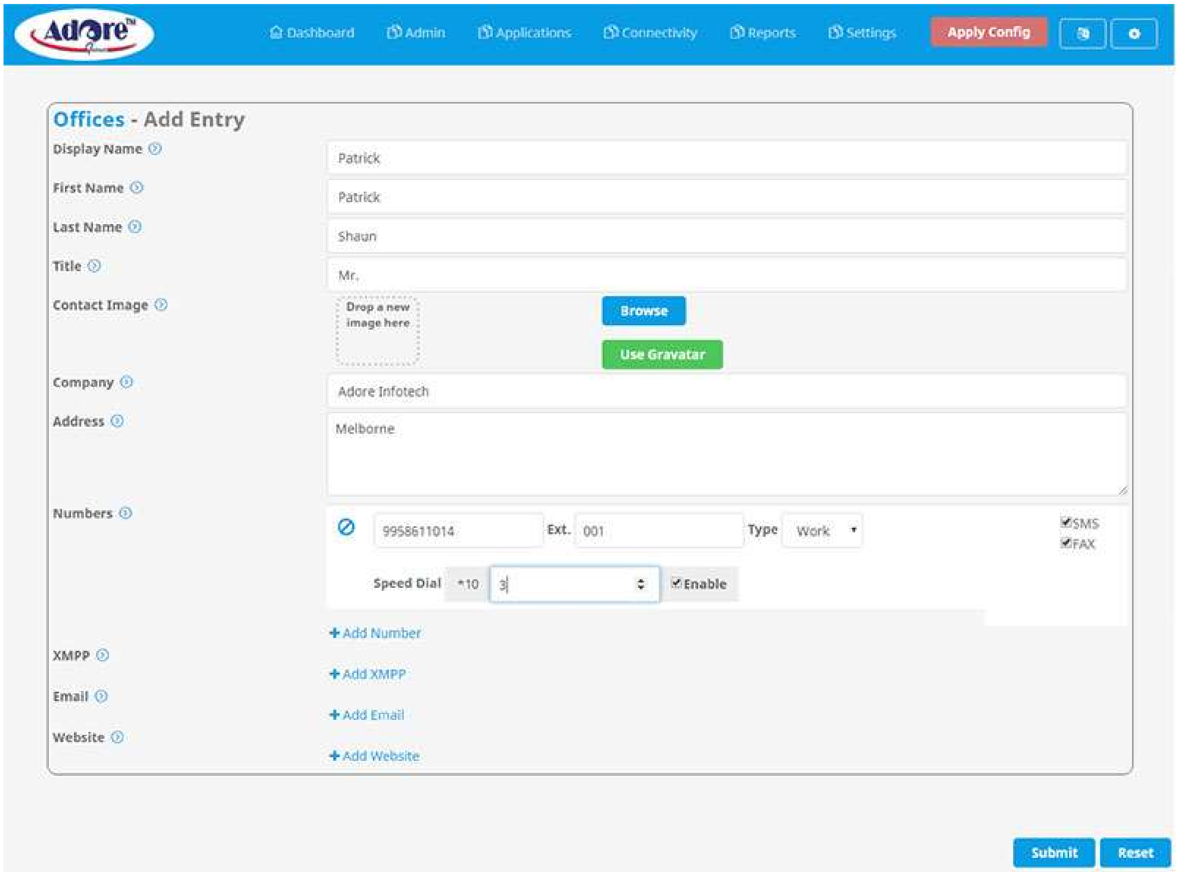
Under the **External Groups** tab, you should see your already-created External groups. Each external group will have its own tab.

In this example, we have an external group called "Vendors." From here you can view all users in this group, as well as modify the group name if desired, edit or delete users, and add entries.



Editing an External User:

- Click the edit button () next to an entry.
- The information for an external user is not automatically pre-populated, because this information is not pulled from internal sources. You can edit the **Display Name, First Name, Last Name, Title, Company, Address, Numbers, XMPP, Email,** and Website.
- Click the **Submit** button when done editing.



Adore
Dashboard
Admin
Applications
Connectivity
Reports
Settings
Apply Config
Offices - Add Entry
Display Name
Patrick
First Name
Patrick
Last Name
Shaun
Title
Mr.
Contact Image
Drop a new image here
Browse
Use Gravatar
Company
Adore Infotech
Address
Melborne
Numbers
9958611014
Ext.
001
Type
Work
SMS
FAX
Speed Dial
*10
3
Enable
+ Add Number
XMPP
+ Add XMPP
Email
+ Add Email
Website
+ Add Website
Submit
Reset

2.6. Custom Destination

Custom Destinations

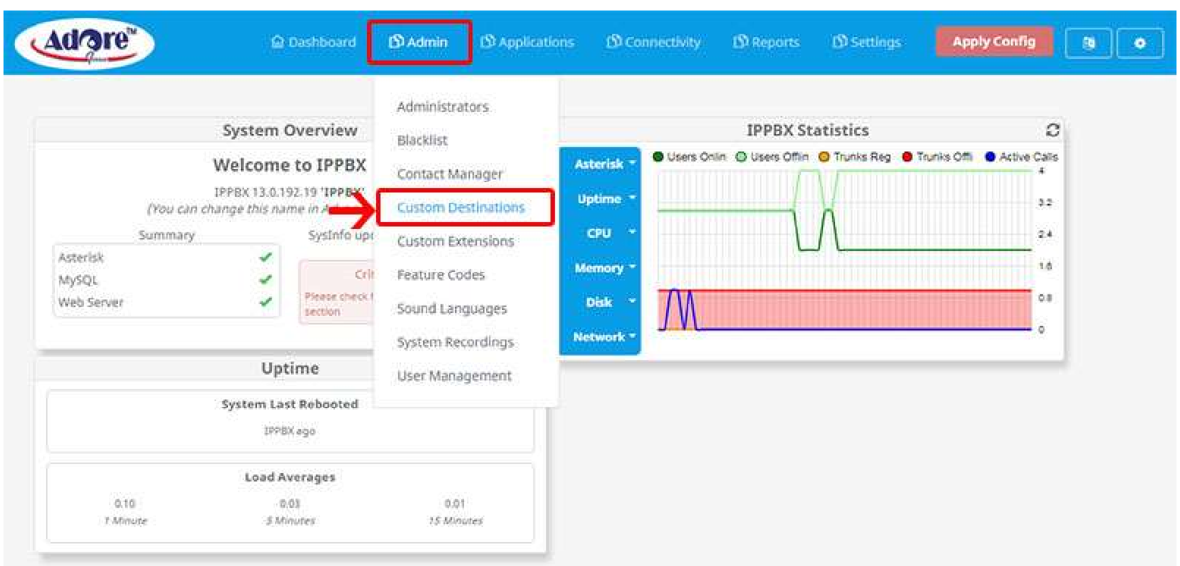
The Custom Destinations module allows you to register your custom destinations that point to custom dial plans and will also 'publish' these destinations as available destinations to other modules. This is an advanced feature that is used to link to custom code on your PBX and should only be used by knowledgeable users.

The custom Destinations module can be used by any module that supports destinations.

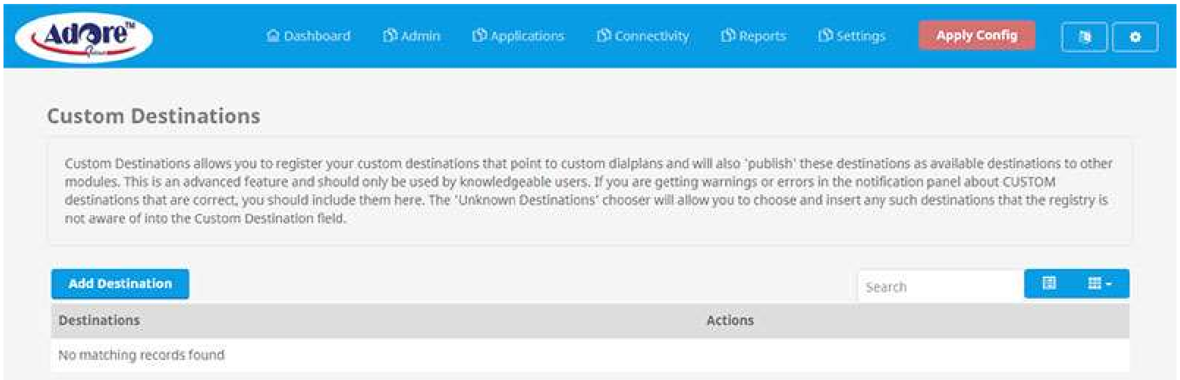
A few examples of modules that use destinations

- Inbound Routes
- Announcements
- Queues
- Extensions

Go to **Admin - Custom Destinations**



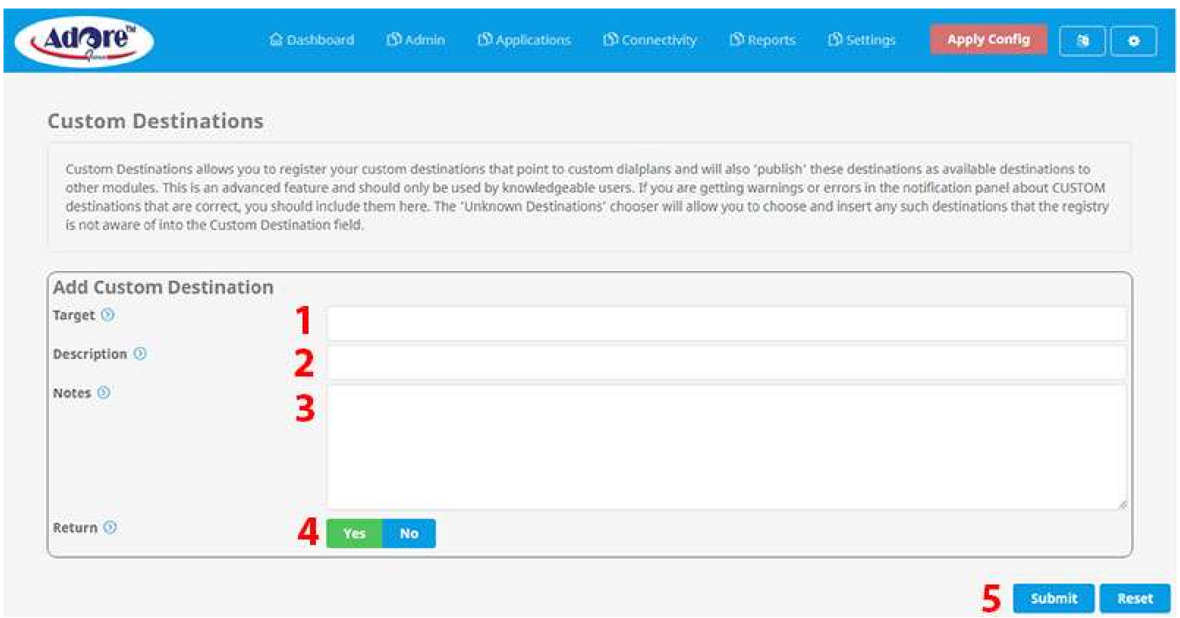
On Click **Custom Destinations** following screen will appear. The Custom Destinations module allows you to register your custom destinations that point to custom dialplans. It will also "publish" these destinations as available destinations to other modules.



Note : This is an advanced feature that is used to link to custom code on your PBX and should only be used by knowledgeable users.

Creating a Custom Destination

For each custom dial plan destination, you can define the following:



1. Target

This is your custom destination. Define the custom dial plan that you want to route the caller to in the format *[context],[exten],[priority]*. Example: “afterhours-pin,501599,1,” which is the start of the custom “After Hours Pin” dialplan we have on this box.

2. Description

Give this custom destination a friendly name. Example: "My Custom Destination"

3. Notes

Here you can define notes on what this custom dialplan or script is used for.

4. Return

Does your custom destination end with 'Return'? If so, you can then select a custom destination after this call flow is complete.

If you select Return: Yes, then you will see a new dropdown menu where you can select the appropriate return destination.

5. Submit

Don't forget to click the **Submit** button and click the red **Apply Config** button at the top when done.

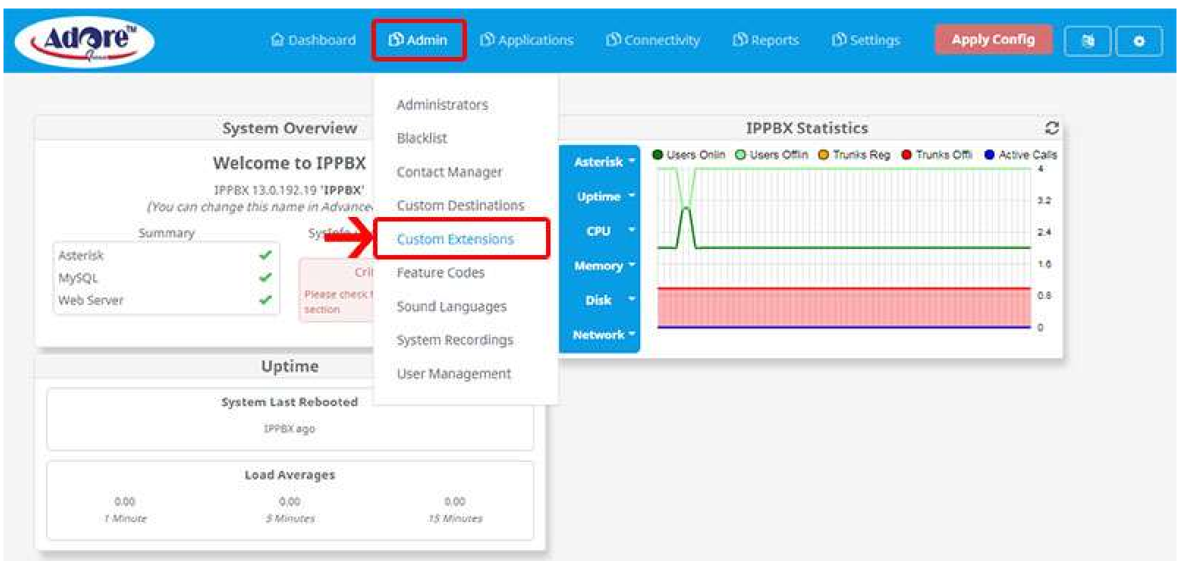
2.7. Custom Extension

Custom Extension

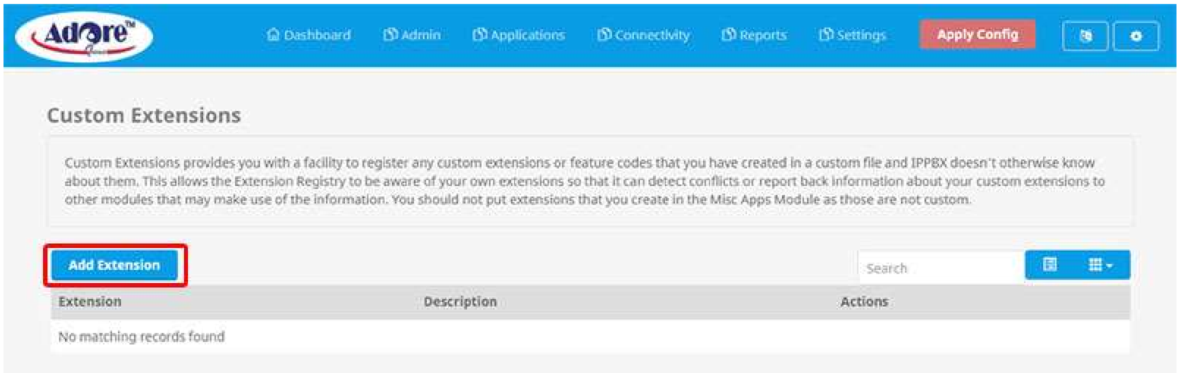
The Custom Extensions provides you with a facility to register any custom extensions or feature codes that you have created in a custom script or dialplan when the PBX doesn't otherwise know about them. This makes the Extension Registry aware of your own extensions so it can detect conflicts or report information about your custom extensions to other modules. Also, this helps prevent you from creating a duplicate extension number somewhere else in the PBX.

The custom extensions registers extensions used in your own custom applications to ensure they don't conflict with other PBX Modules.

Go to **Admin** -> **Custom Extension**

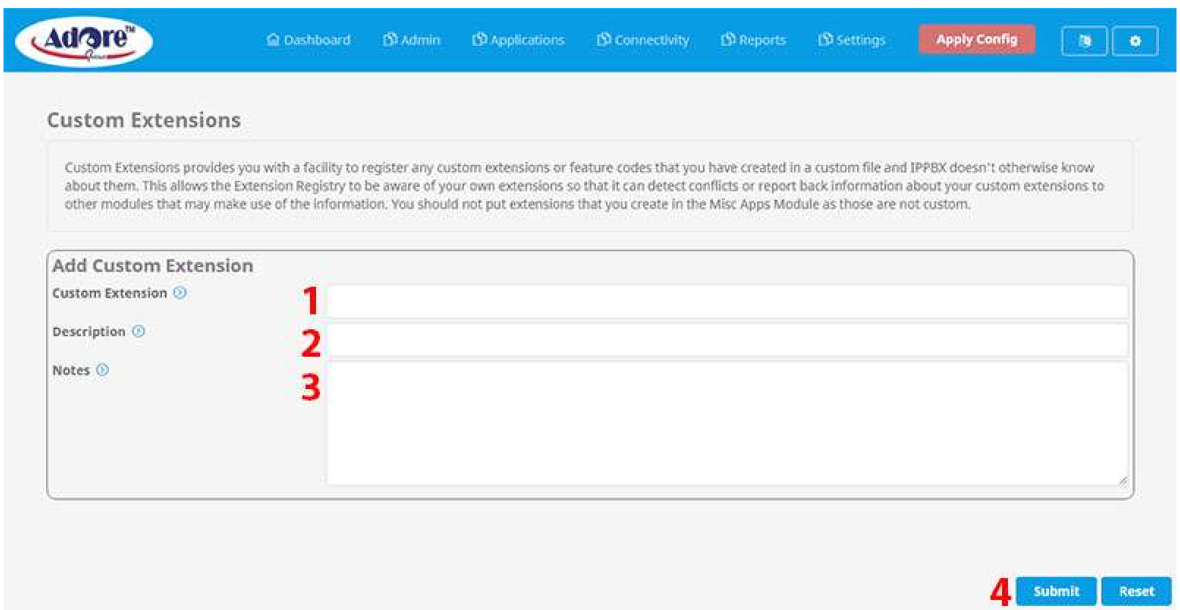


On click **Custom Extensions** following screen will appear. For adding Custom Extension click on "**Add Extension**" button.



Creating a Custom Extension

For each custom extension, you can define the following:



1. Custom Extension

Define the custom extension number that you want the PBX to be aware of. You will not allowed to use the number for something else.

2. Description

Give this custom extension a friendly name.

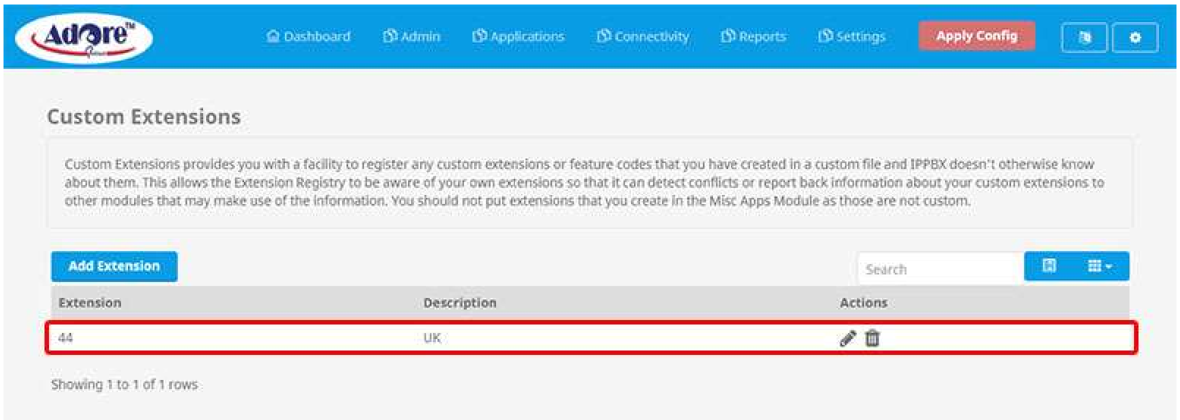
3. Notes

Here you can enter notes on what this custom extension is used for.

4. Submit

When done, don't forget to click the **Submit** button followed by the red **Apply Config** button in order to save and apply your changes.

Now Custom Extension are added.



2.8. Feature Codes

Feature Code

The Feature Codes Module is used to enable and disable certain features available in your PBX and Asterisk, and to set the codes that local users will dial on their phones to use that particular feature.

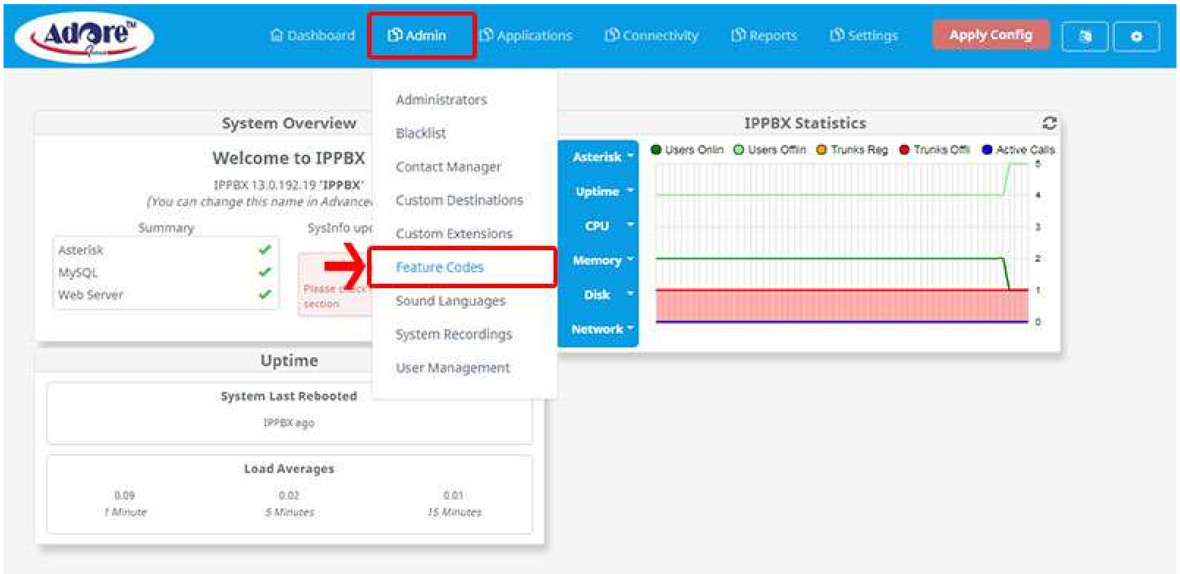
For example, the Feature Codes Module can be used to set the code that a user will dial to activate or deactivate Call Forwarding. It can also be used to set a Code that can be used to enter into an Echo Test, to hear your extension number, or to hear the time of day.

The Feature Codes Module sets the codes used to activate and deactivate features that are configured in various other PBX modules.

In addition, the Feature Codes Module is related to the Advanced Settings Module. In the Advanced Settings Module, when the Enable Custom Device States option is activated, the PBX will create a set of codes, based upon some of the Feature Codes, that can be used to enable phones to display the status of various features on a light. The feature is known as a Busy Lamp Field (BLF). For example, a Busy Lamp Field can be used to show whether a Call Flow Control is enabled or disabled or whether a particular extension has enabled or disabled the Do Not Disturb feature, etc.

You can see a list of codes that can be used to display the status of the features on a light by accessing the Asterisk Info Module, and then clicking on Subscriptions. Each of the codes listed there can be entered into a phone's Busy Lamp Field and will display the status of the related Feature Code.

Go to **Admin** -> **Feature Codes**



On click **Feature Codes** following screen will appear.

The Feature Code module is where you can view all the feature codes that are built into your PBX. You can perform the following actions for each feature code:

- Change a feature code number.
- Enable or disable the feature code.

Customizing a Feature Code

You can customize a feature code by setting it to something different from the system default.

*In our example we will look at the "blacklist a number" feature code. The default is *30. Notice the **Code** field is grayed-out and cannot be edited at this point.*

Select the **Customize** button. The button's color will change to dark blue. The background of the Code field will change to white, indicating that you can now edit it.

Click the **Submit** button followed by the red **Apply Config** button.

Adore Dashboard Admin Applications Connectivity Reports Settings Apply Config

Feature Code Admin

— Blacklist

Description	Code	Actions	
Blacklist a number	*30	Customize	Enabled
Blacklist the last caller	*32	Customize	Enabled
Remove a number from the blacklist	*31	Customize	Enabled

— Callforward

Description	Code	Actions	
Call Forward All Activate	*72	Customize	Enabled
Call Forward All Deactivate	*73	Customize	Enabled
Call Forward All Prompting Activate	*93	Customize	Enabled
Call Forward All Prompting Deactivate	*74	Customize	Enabled
Call Forward Busy Activate	*90	Customize	Enabled
Call Forward Busy Deactivate	*91	Customize	Enabled
Call Forward Busy Prompting Activate	*94	Customize	Enabled
Call Forward Busy Prompting Deactivate	*92	Customize	Enabled
Call Forward No Answer/Unavailable Activate	*52	Customize	Enabled
Call Forward No Answer/Unavailable Deactivate	*53	Customize	Enabled
Call Forward No Answer/Unavailable Prompting Activate	*95	Customize	Enabled
Call Forward Toggle	*96	Customize	Enabled

— Callwaiting

Description	Code	Actions	
Call Waiting - Activate	*70	Customize	Enabled
Call Waiting - Deactivate	*71	Customize	Enabled

— Conferences

Description	Code	Actions	
Conference Status	*87	Customize	Enabled

— Contactmanager

Description	Code	Actions	
Contact Manager Speed Dials	*10	Customize	Enabled

— Core

Description	Code	Actions	
Asterisk General Call Pickup	*8	Customize	Enabled
ChanSpy	555	Customize	Enabled
Directed Call Pickup	**	Customize	Enabled
In-Call Asterisk Attended Transfer	*2	Customize	Enabled
In-Call Asterisk Blind Transfer	##	Customize	Enabled
In-Call Asterisk Disconnect Code	**	Customize	Enabled
In-Call Asterisk Toggle Call Recording	*1	Customize	Enabled
Simulate Incoming Call	7777	Customize	Enabled
ZapBarge	888	Customize	Enabled

— Daynight

Description	Code	Actions	
4: This is a testing purpose.	*284	Customize	Enabled

— Donotdisturb

Description	Code	Actions	
DND Activate	*78	Customize	Enabled
DND Deactivate	*79	Customize	Enabled

Note : Be sure the new feature code is not a duplicate of a code used elsewhere already. (The system will not warn you of duplicates.) Also, the feature code should be absolutely unique. Using *12 and *123 may cause an issue in some versions.

Restoring a Feature Code to the System Default

It is easy to change a customized feature code back to its system default, and you do not have to remember what the original default was. The system will take care of it for you when you de-select the **Customize** option.

Your feature code is in a customized state if the **Customize** button is dark blue and the **Code** field has a white background and is editable.
*For example, here we changed our Blacklist feature code to *9 as shown earlier.*

To restore the default, simply click the **Customize** button again. It will change to light blue, the **Code** field will revert back to the default, and the field will not be editable.

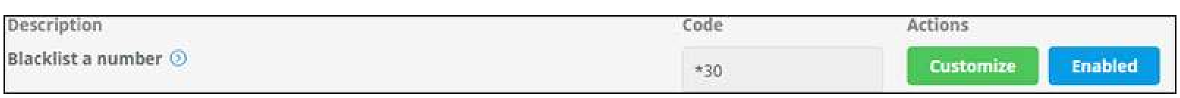
*In this example, the default code for blacklisting a number has been restored to the default of *30.*

Click the **Submit** button followed by the red **Apply Config** button.

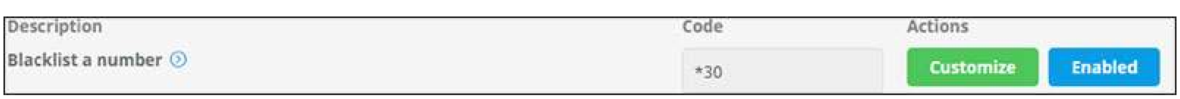
Enabling or Disabling a Feature Code

In the event that you would like to completely disable a feature code from being used, you can toggle the Enable/Disable option for it.

To disable a feature code, click the **Enable** button to turn it to light blue in color, then click the **Submit** button followed by the red **Apply Config** button.



To enable a feature code, click the **Enable** button to turn it to dark blue in color, then click the **Submit** button followed by the red **Apply Config** button.



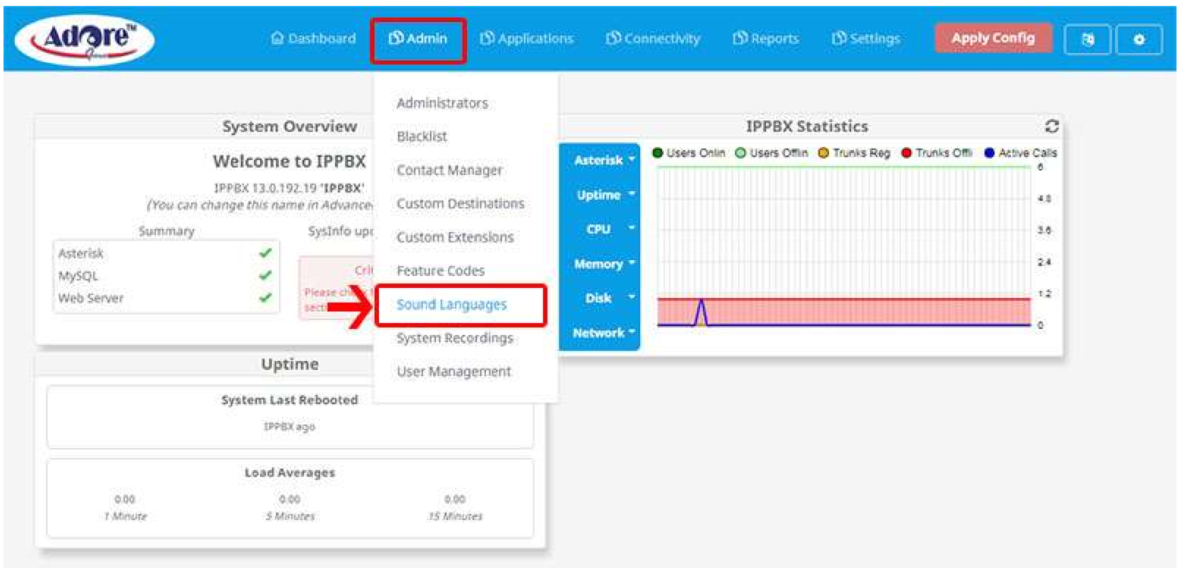
Remember, the **Enabled** button shows whether you have selected an Enabled or Disabled state, but the settings aren't applied until you click **Submit** and **Apply** config. Also, remember dark blue=enabled and light blue=disabled.

2.9. Sound Languages

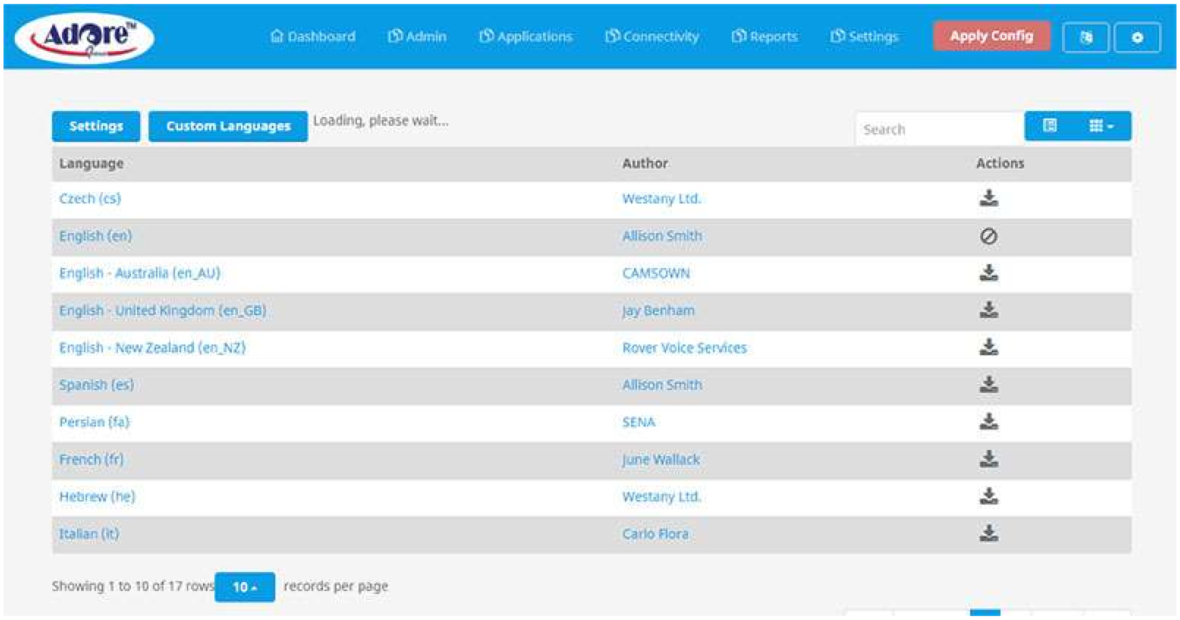
Sound Languages

The Sound Languages module controls which languages are available for sound prompts, for both callers and users. You can define the global default language for your system and make other languages available. This is where you install or remove languages.

Go to **Admin** -> **Sound Languages**



On click **Sound Languages** following screen will appear. Here you can select a language for sound prompts in modules such as Inbound Routes, Conferences, and Extensions to name a few. You can also use the Languages module (not to be confused with Sound Languages) to change the language for sound prompts at any point in the call flow. The available languages will depend upon what you have installed in the Sound Languages module.



The Sound Languages module allows you to set the global default language for sound prompts. The system will use the global language defined here unless the language is changed somewhere in the call flow, such as by an inbound route, extension, or the Language module.

The Sound Languages module also allows you to add or remove additional languages in various codecs. Several language packs are available. You can also define a language code for custom languages, which will allow you to install and use sound files for that custom language.

The system has hundreds of built-in sound prompts for both callers and users. These are *core-sounds* and *extra-sounds*.

Changing the Language in a Call Flow

Detailed instructions are beyond the scope of this wiki, but there are two general ways to change the language in a call flow.

Using the Languages module (not to be confused with the Sound Languages module), you can dynamically change the language used, anywhere in the call flow.

Many other modules have the option to change the language. Look for a "Languages" option, and visit the module-specific wiki for more information. For example, in the inbound routes module, you can set a language for a specific DID. If the French language is installed (confirmed by checking the Sound Languages module), you can choose French for the inbound route. Sound prompts would then be played in French for calls inbound to this DID, and this selection would carry over to other points in the call flow unless it is changed later in the flow.

Available Language Packs

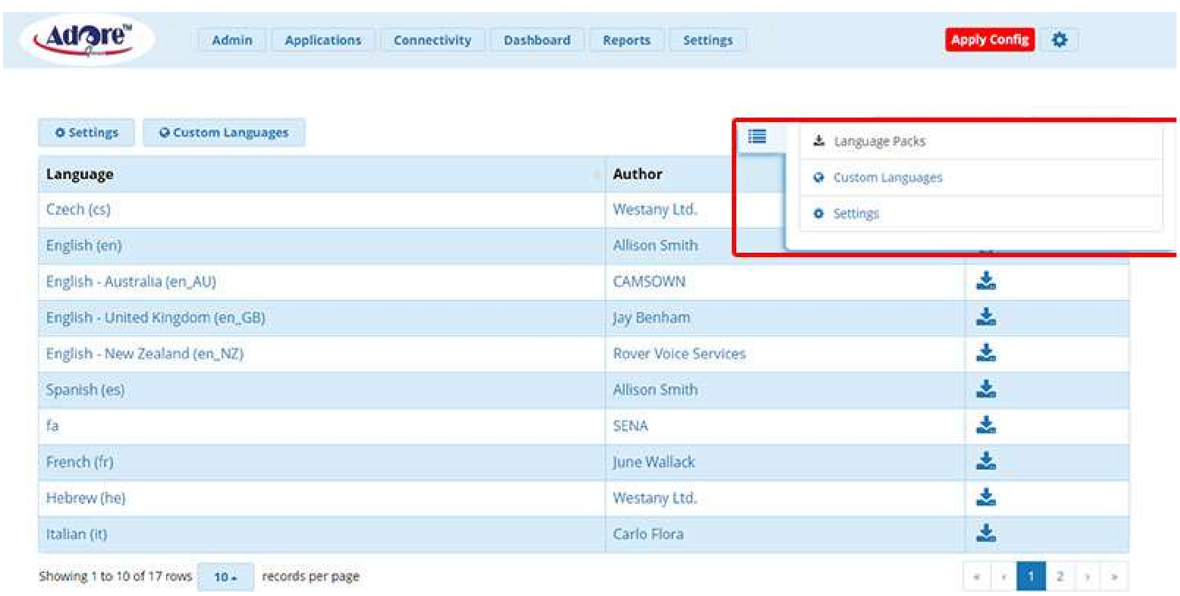
The following languages are available as built-in options at the time of this writing:

- American English
- Australian English
- British English
- Spanish
- French
- Italian

You can also install custom languages.

How to Install and Remove Language Packs

1. When you arrive at the Sound Languages module, the Language Packs section is shown by default.



This screen displays a list of languages. (Note the list may be several pages long). Here is what the columns mean:

a. **Module:** core-sounds or extra-sounds

b. **Language:** The name of the language and its language code (i.e. "en" for English)

c. **Format:** The codec (i.e. alaw, g722, g729, or ulaw)

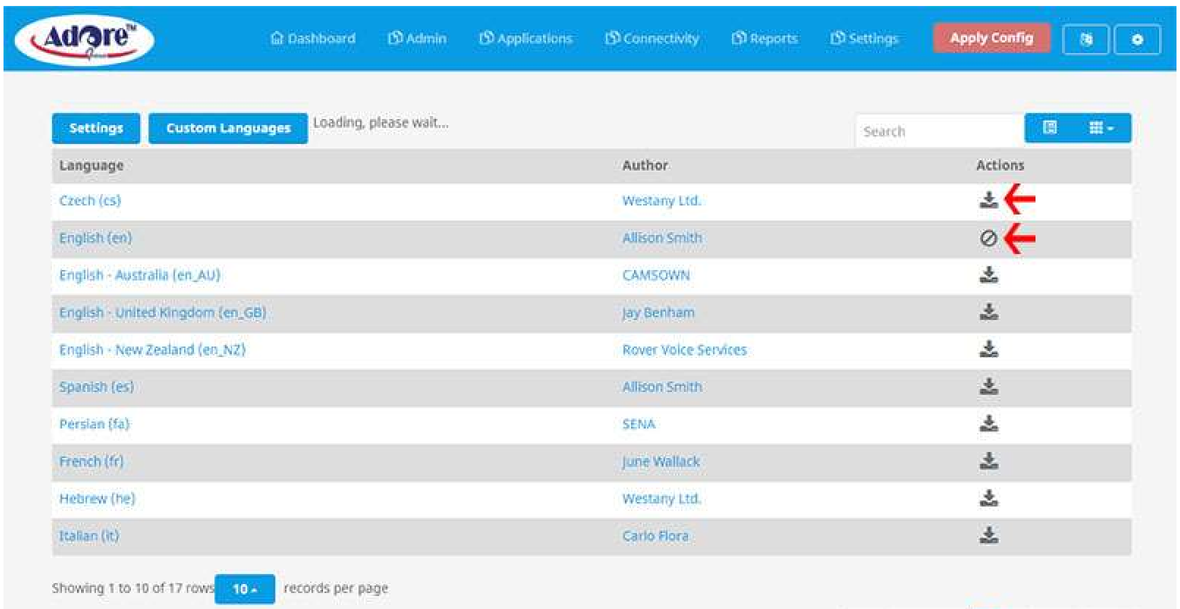
d. **Available:** The latest available verison

e. **Installed:**The version installed on your system, if any

f. **Actions:** A button to install or remove the language

   i. = Download

   ii. = Remove



1. **To Install:** Click the download button:

2. **To Remove:** Click the remove button:

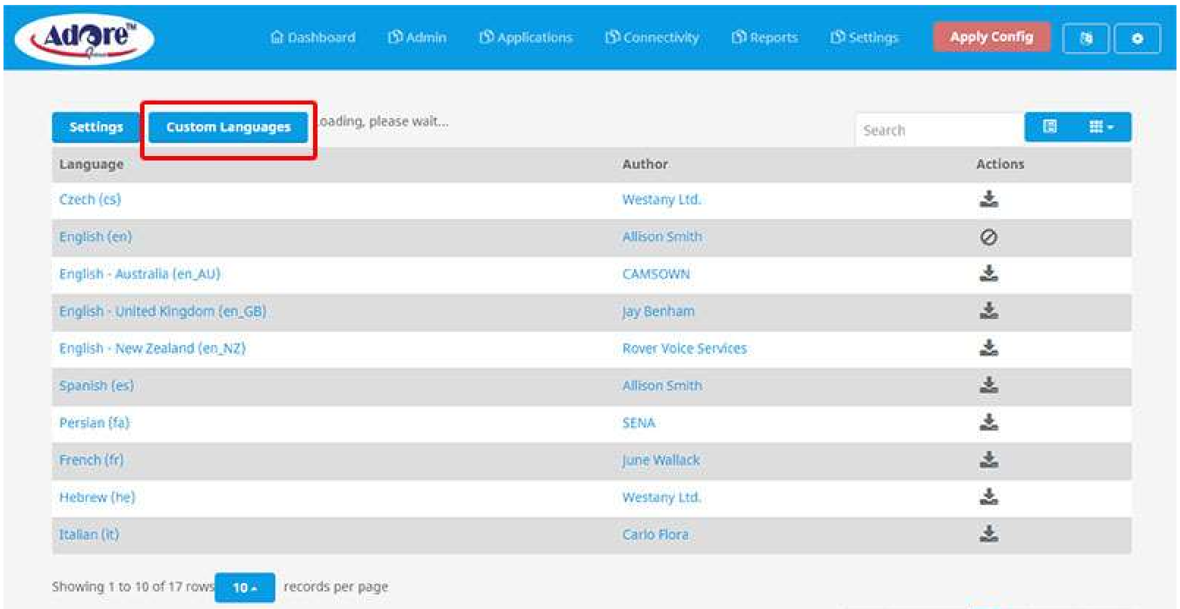
3. When finished, click the **Apply Config** button to apply your changes.

How to Define a Custom Language

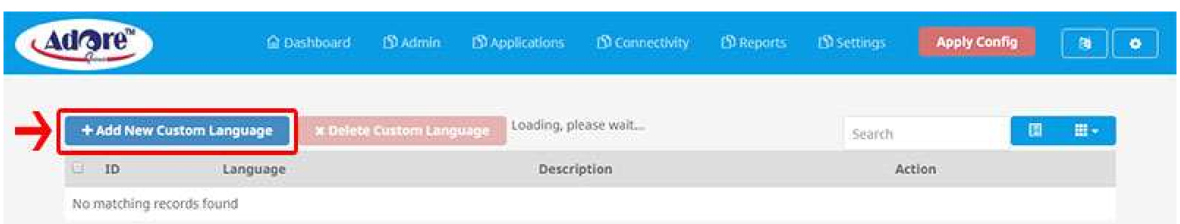
You can add a new language code to your system in preparation for installing your own custom language sound files. This will create a directory in /var/lib/asterisk/sounds with the name of the language code you enter here. It will also make the custom language available for selection in other modules.

After you have defined a custom language, you can upload your own custom language recordings.

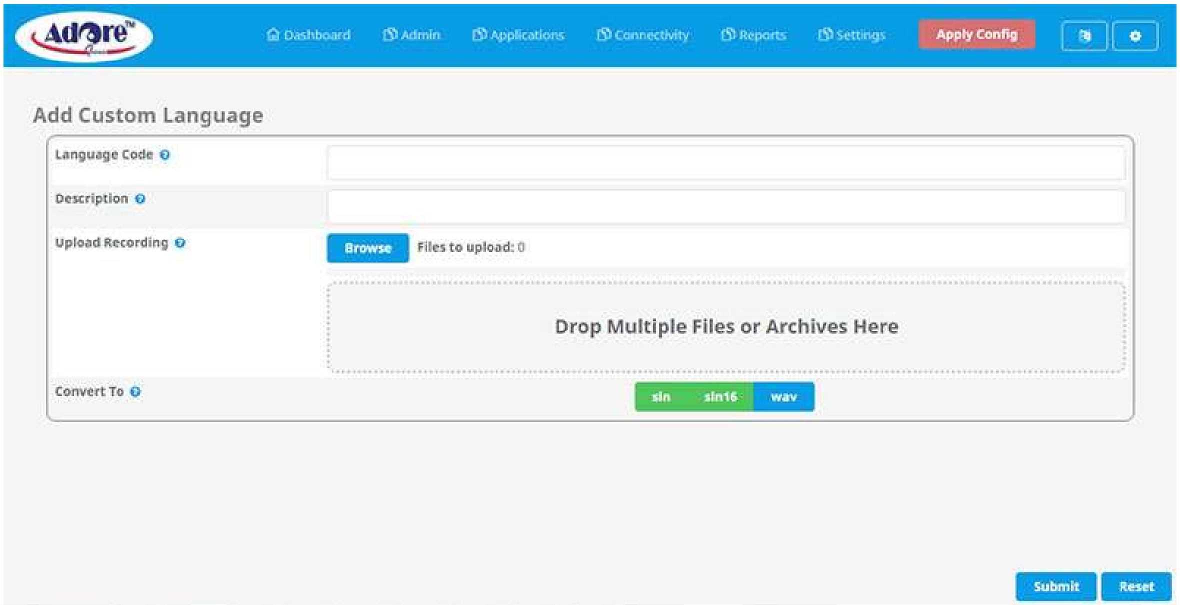
1. From the options at the right, click the **Custom Languages** option to navigate to that section.



2. Click the **+Add New Custom Language** button.



3. This is what the **Add Custom Language** section looks like.



Enter the **Language Code** (for example we're calling this "Hi" for hirate) and a brief **Description** to help you identify this custom language. This name will appear in other modules where you can select a language.

Here you can upload your custom language recordings. Either **Browse** your computer or **Drag and Drop Multiple Files or Archives**. Supported upload formats are: ***WAV, aiff, alaw, flac, g722, gsm, mp3, oga, ogg, sln, sln12, sln16, sln192, sln24, sln32, sln44, sln48, sln96, ulaw, wav, wav16, tgz, gz, tar, zip***. This includes archives (that include multiple files, such as ***tar,gz,zip***) and multiple files.

Once your file(s) is uploaded, the blue progress bar will be full and **File to Upload** will populate with the corresponding number of files.

Lastly check all file formats you would like this system recording to be encoded into.

Click the **Submit** button and a progress bar will let you know when you're finished.

Your new custom language will show up in a list. You can click the pencil button to edit or the trash button to delete.

2.10. System Recordings

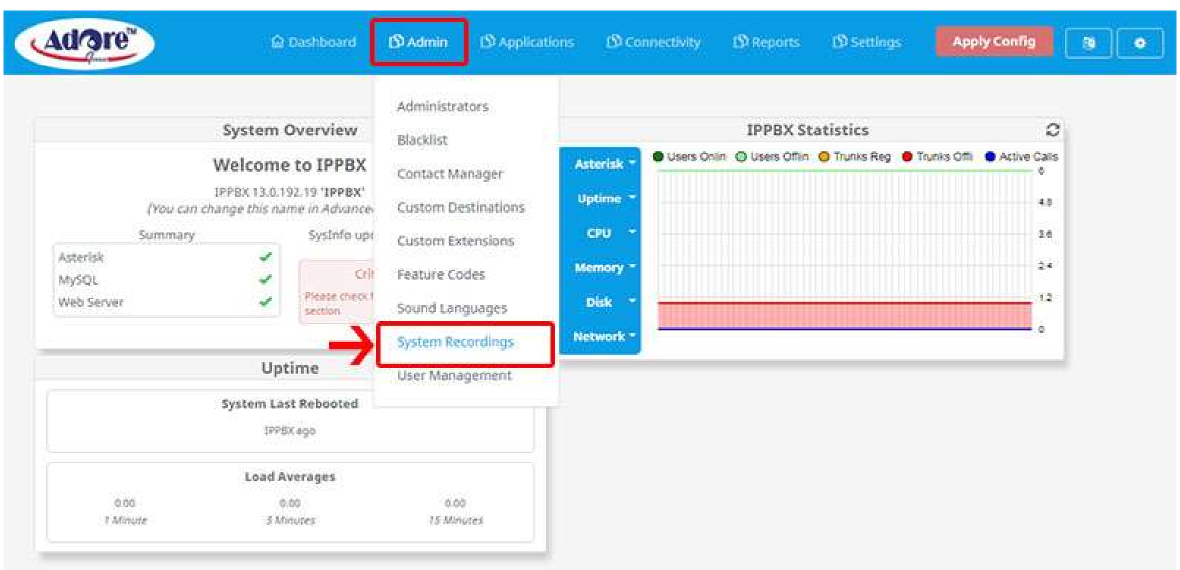
System Recordings

The System Recordings module is used to record or upload messages that can then be played back to callers in other modules. It can also be used to make pre-installed Asterisk recordings available for use in other modules.

For example, you might create a recording called "Main Menu" and then play that message in an IVR before a caller is asked to make a selection. Or, you might record a recording called "Holiday Message" and then use that message in an Announcement. You would then route incoming calls to the Announcement or IVR using the Inbound Routes Module.

The System Recordings module allows you to create and name recordings that can be selected in any module that supports the playing of recordings. Among others, the IVR module, Announcements module, Follow Me module, Queues module, and Ring Groups module contain options to select a recording.

Go to **Admin** -> **System Recordings**



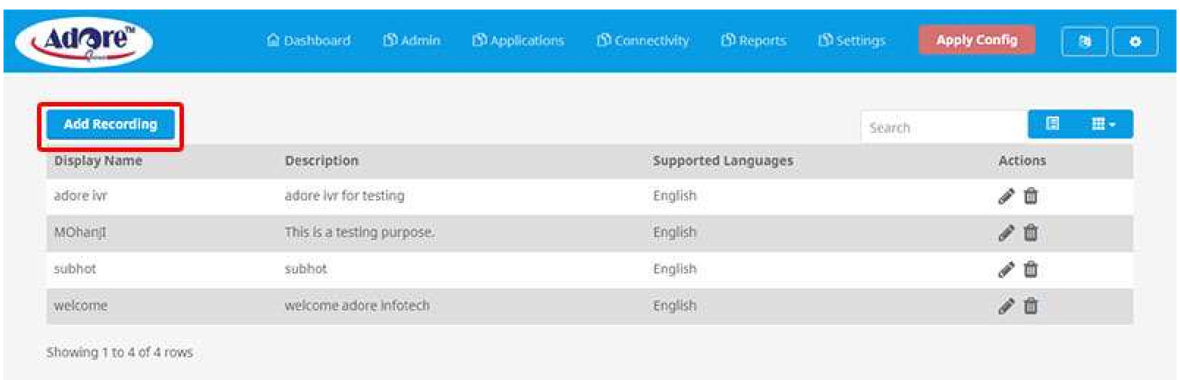
On click **System Recordings** following screen will appear.

The System Recordings module allows for the management of built-in recordings and provides an easy-to-use interface for adding new recordings for IVRs, Announcements, Queues and so on.

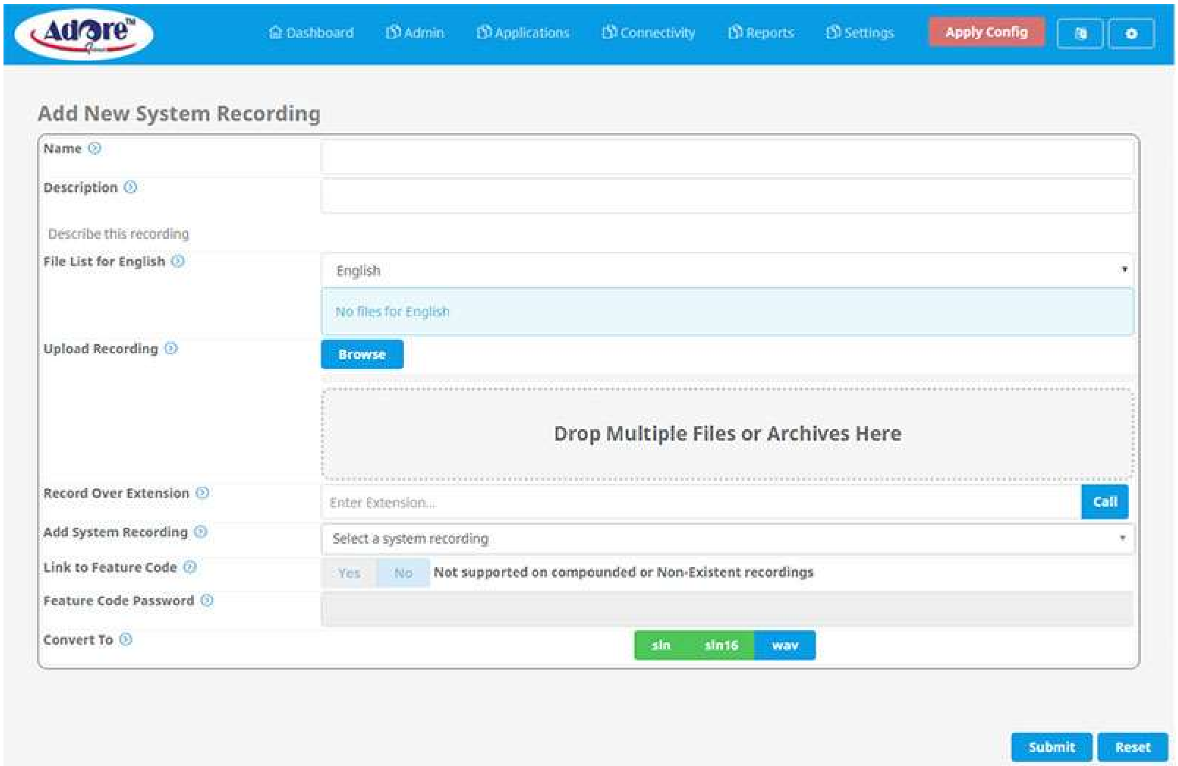
There are three ways to add System Recordings: uploading a file, recording within the browser, and recording over an extension.

Adding a System Recording

Click the **Add Recording** button.



On click **+ Add Recording** following screen will appear.



Name

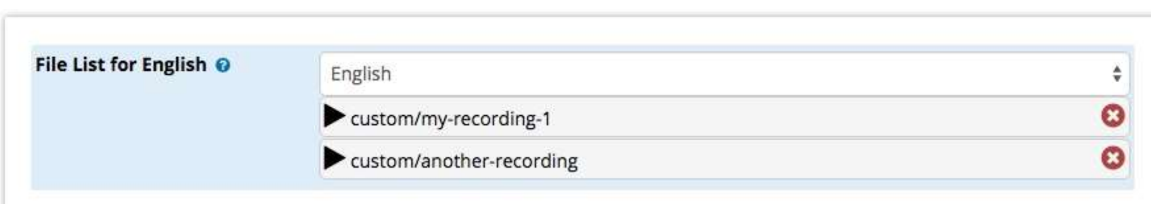
The name of the system recording on the file system. If it conflicts with another file, then this will overwrite it.

Description

A description of this recording to help you identify it.

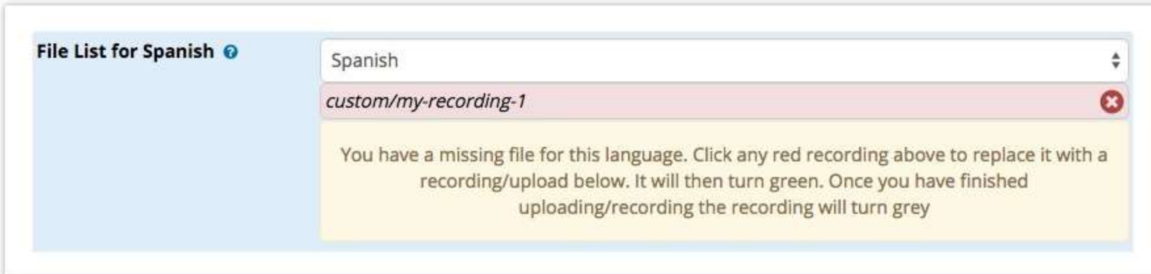
File List for (Language)

A sortable File List / play order. Here, you can string multiple files together into one recording. The playback will be done starting from the top to the bottom.



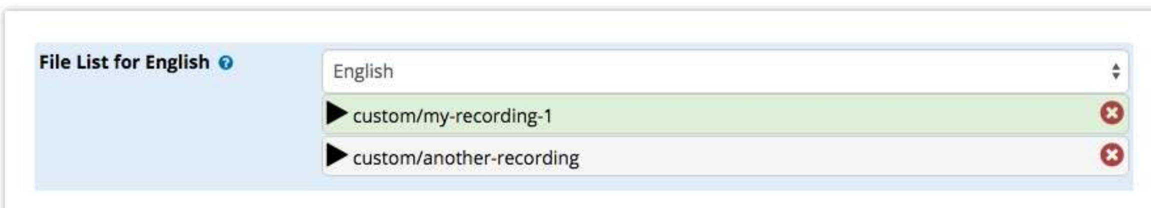
- Click the play icon ▶ to preview a file.
- Click the X icon to remove a file.

- Click on a file name and drag it up or down to a new position to change the playback order.
- Note: if a file is red, it is missing for the selected language.



Replacing a File (Re-Recording)

Files can be replaced by clicking them once, which will turn them green and place them into replace mode. (A dialog box will appear asking you to confirm that this is what you want to do.) The next recording you record or upload will then replace this green file upon save.



Upload Recording

Allows upload of files from your local system. Supported upload formats are: WAV, aiff, alaw, flac, g719, g722, gsm, mp3, oga, ogg, sln, sln12, sln16, sln192, sln24, sln32, sln44, sln48, sln96, ulaw, wav, wav49. This includes multiple files, and archives that contain multiple files.

Click the **Browse** button to select a file from your computer. Or, drag and drop files from your desktop onto the **Drop Multiple Files or Archives Here** box.

Record in Browser

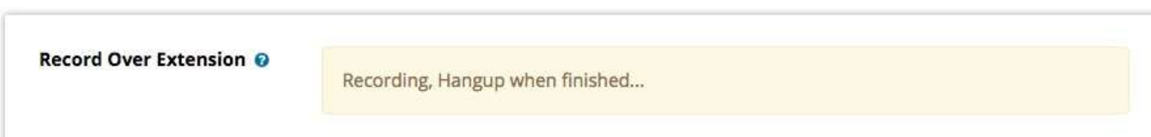
This will initiate a WebRTC request so that you will be able to record from you computer in your browser.

- Click the red record button  to begin recording. Your browser may alert you that the PBX is requesting access to your microphone. This is normal; click "Allow" or a similar option to permit access.
- The record button will flash when recording is in progress.
- Begin speaking, then press the stop button when finished. 
- Click the play button  to review your recording.
- Click the **Save Recording** to save the recording, or the **Delete Recording** button to delete it.

Record Over Extension

The system will call the extension you specify. Upon hangup, you will be able to name the file and it will be placed in the list.

- Enter the extension number the system should call.
- Click the **Call!** button. The system will call the extension.
- Answer the call and begin speaking after the beep. The GUI will display a message stating that recording is in progress. Hang up when finished.



At this point, you can save the recording or delete it. To delete it, click the **Cancel** button. To save it, enter a name in the **Name this file** field and then click the **Save** button.

Add System Recording

This option allows you to add any previously created system recording to the list of files above. To use this option, select a recording from the dropdown menu. It will be added to the file list.

Link to Feature Code

Yes/No: (Options not available on non-existent or compound recordings.) Whether to allow users to re-record the recording by dialing a feature code. Select **Yes** to create a feature code that will allow this recording to be changed directly by a user. This allows users to make their own changes without needing to contact an administrator.

Feature Code Password

(Optional) If the **Link to Feature Code** option is set to **Yes**, this password can prevent unauthorized users from accessing the feature code for the recording. Users will need to enter this password before they can re-record the recording via the feature code. Enter digits only. If the **Link to Feature Code** option is set to **No**, this field will be grayed-out.

Convert To

The file formats you would like this system recording to be encoded into. Options include **alaw**, **g719**, **g722**, **gsm**, **sln**, **ulaw**, **wav**, and **wav49**. Select one or more file formats.

Save

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

2.11. User Management

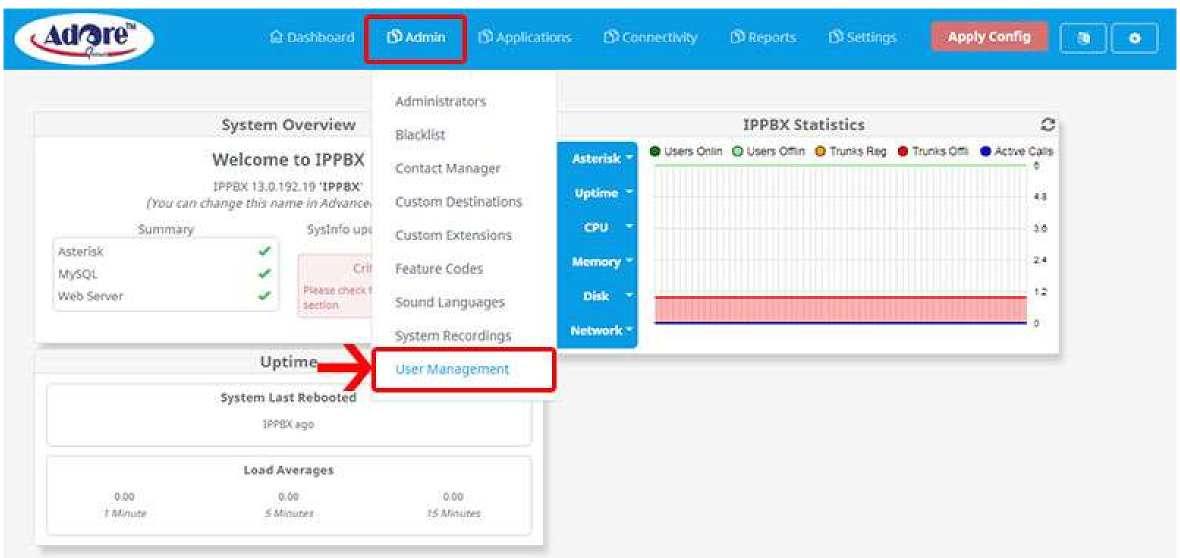
User Management

The User Management (userman) module controls and manages users and administrators for your PBX. Including UCP, Zulu, Admin, iSymphony and more.

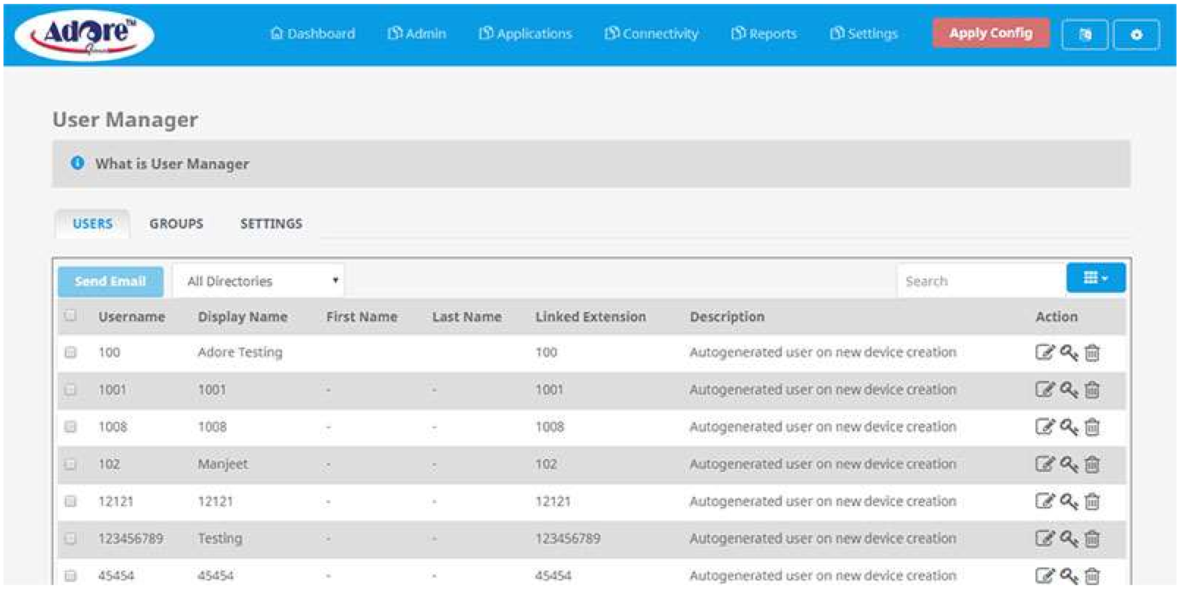
In the User Management module, you can create users who have access to extensions and the settings associated with those devices.

For example, a user could be allowed to log into the User Control Panel and access the voicemail of three other accounts besides their own primary extensions.

Go to **Admin** -> **User Management**

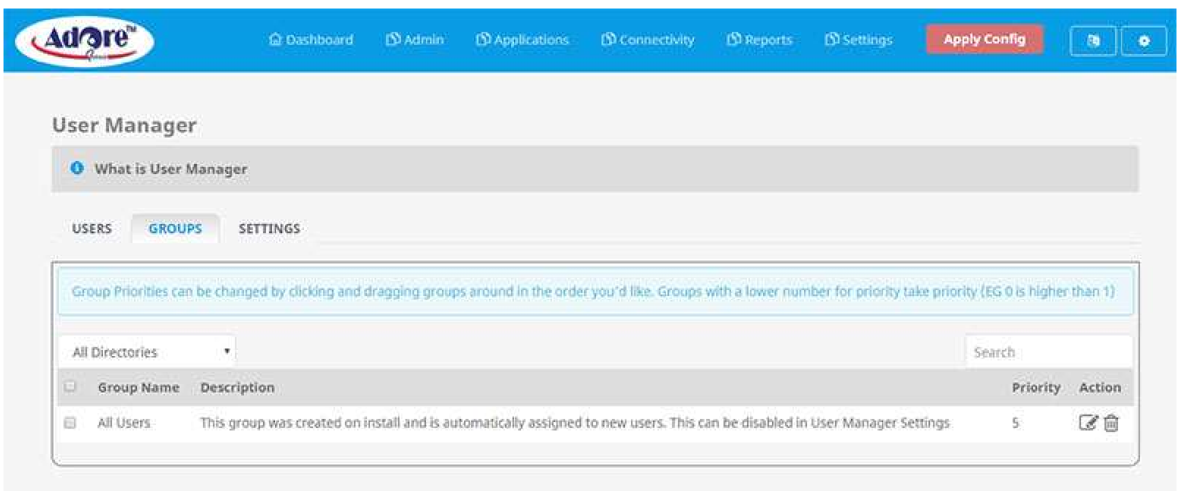


On click **User management** following screen will appear. The **Users** tab should be active, and a list of all current users should be displayed.



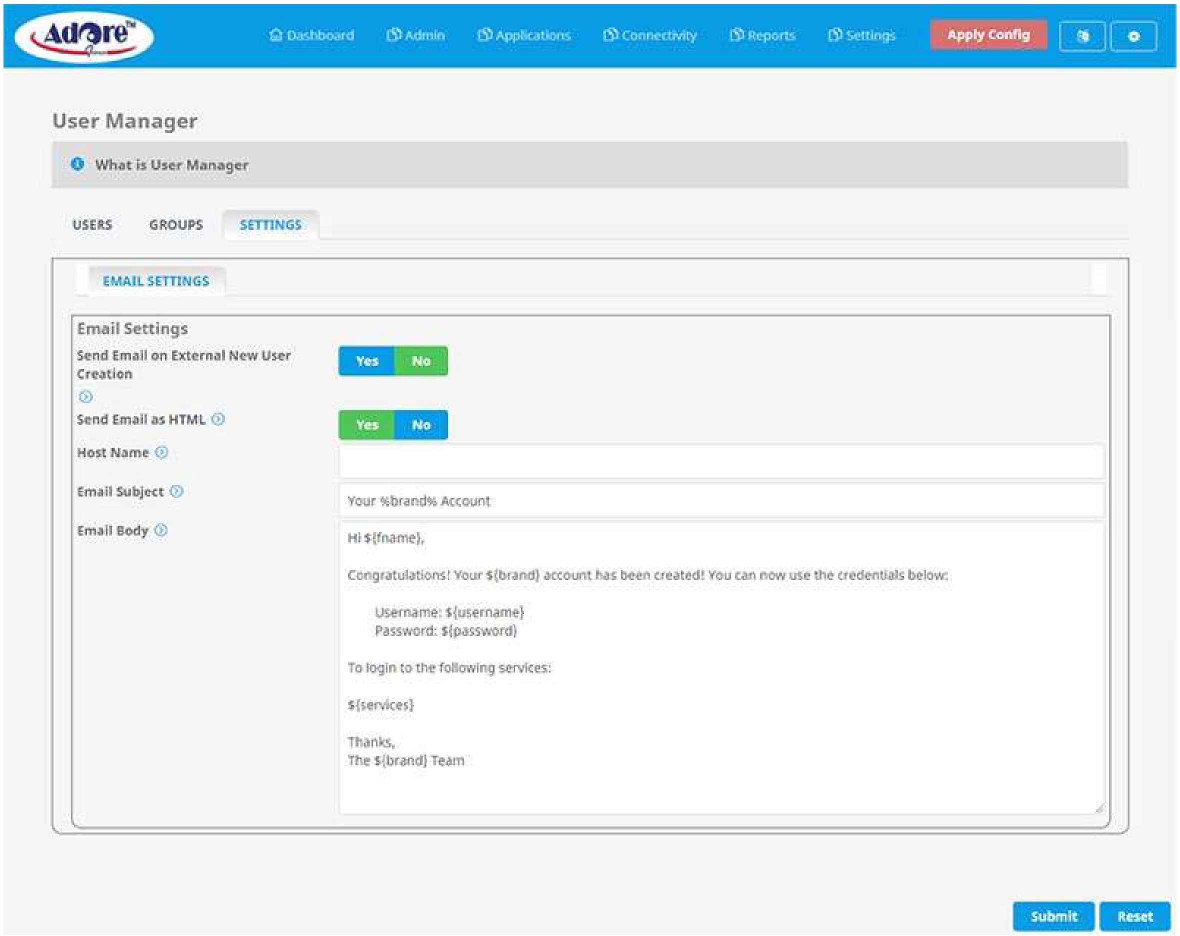
Groups

Upon clicking the '**Groups**' tab in the User Management module you will see the listing of all of your groups on the system.



Settings

The settings tab lets you define global User Manager settings.



- **Send Email on External New User Creation:** Whether to send an email (using the 'Email Subject' and 'Email Body') to new users when they are created externally (not directly through User Manager)
- **Send Email as HTML:** Whether Email Body will send as HTML or plain text to the user
- **Host Name:** The hostname used for email. If left blank the default value of the address in your browser will be used
- **Email Subject:** Text to be used for the subject of the welcome email.
  - Useable variables are:
    - fname: First name
    - lname: Last name
    - brand: Your Company Name
    - title: title
    - username: Username
    - password: Password
- **Email Body:** Text to be used for the body of the welcome email
  - Useable variables are:

- fname: First name
- lname: Last name
- brand: Your Company Name
- title: title
- username: Username
- password: Password

3. Applications

Applications Module

Under Applications module you can handle all the features.

- **Announcements**
- **Call Flow Control**
- **Call Recording**
- **Conferences**
- **Directory**
- **Extensions**
- **Follow Me**
- **IVR**
- **Paging and Intercom**
- **Parking**

3.1. Announcements

Announcements

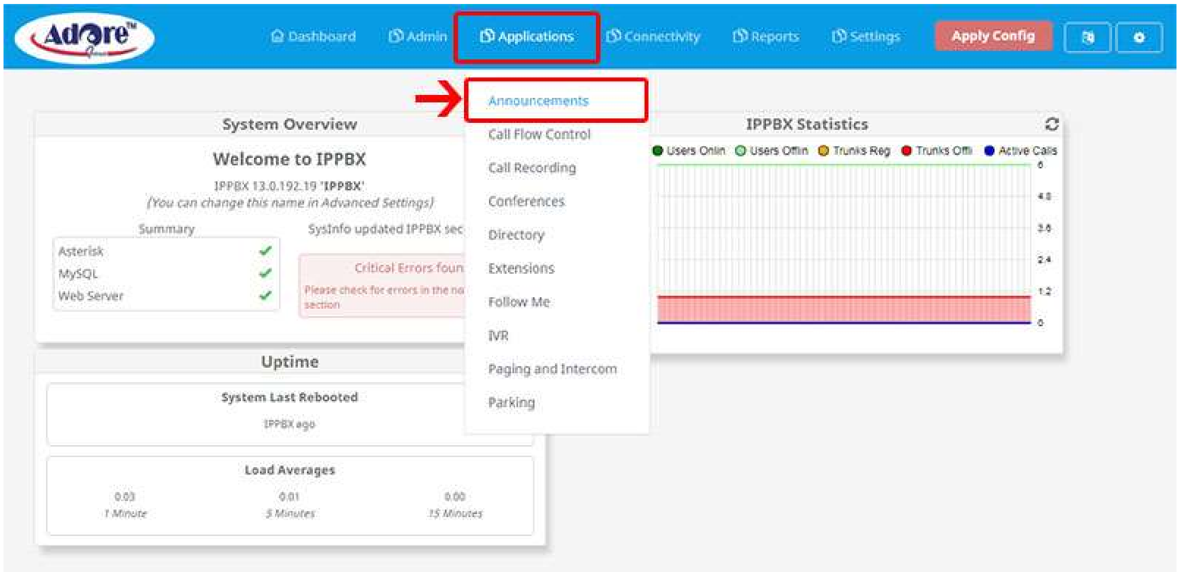
The Announcements Module is used to create a destination that will play an informational message to a caller. After the message is played, the call will proceed to another destination.

For example, you might create an Announcement that plays the address, fax number, and the web-site of your business. A caller could reach that message by pressing the number 2 from the company's main menu. After hearing the message, the call might be routed back to the company's main menu and allowed to make another selection.

The Announcements Module is related to any Module that has a field to set a destination for a call to be routed. There are a number of PBX Modules that are used to route calls to a destination. For example, the IVR Module Module permits you to set up a menu system where a caller can use his touchtone telephone to select from various options. If you configure an Announcement on the Announcements Module, that Announcement will appear as one of the possible destinations in the IVR Module. The Announcement will also appear as a potential destination in any Module that supports destinations, including the Inbound Routes Module Module, the Ring Groups Module Module, the Queues Module Module, the Call Flow Control Module Module, the Time Conditions Module Module, and the Misc Applications Module Module.

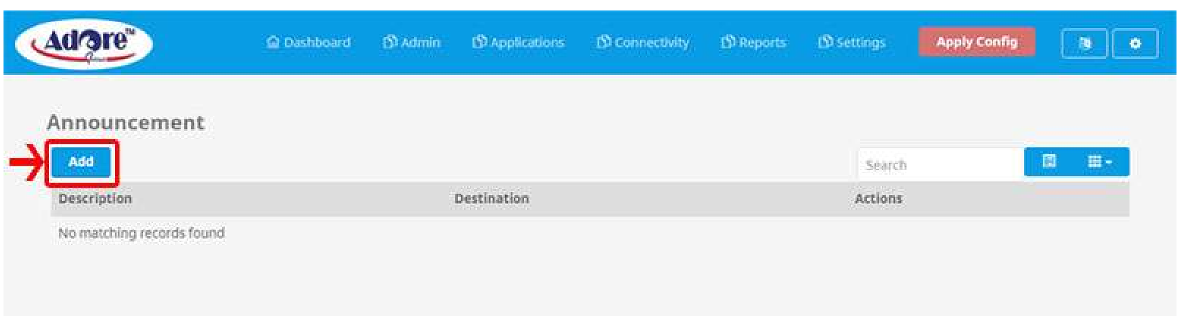
The Announcements Module is also related to the System Recordings Module Module. The System Recordings Module allows you to use your phones to make audio recordings that can be selected for playback in the Announcements Module.

Go to **Applications** -> **Announcements**

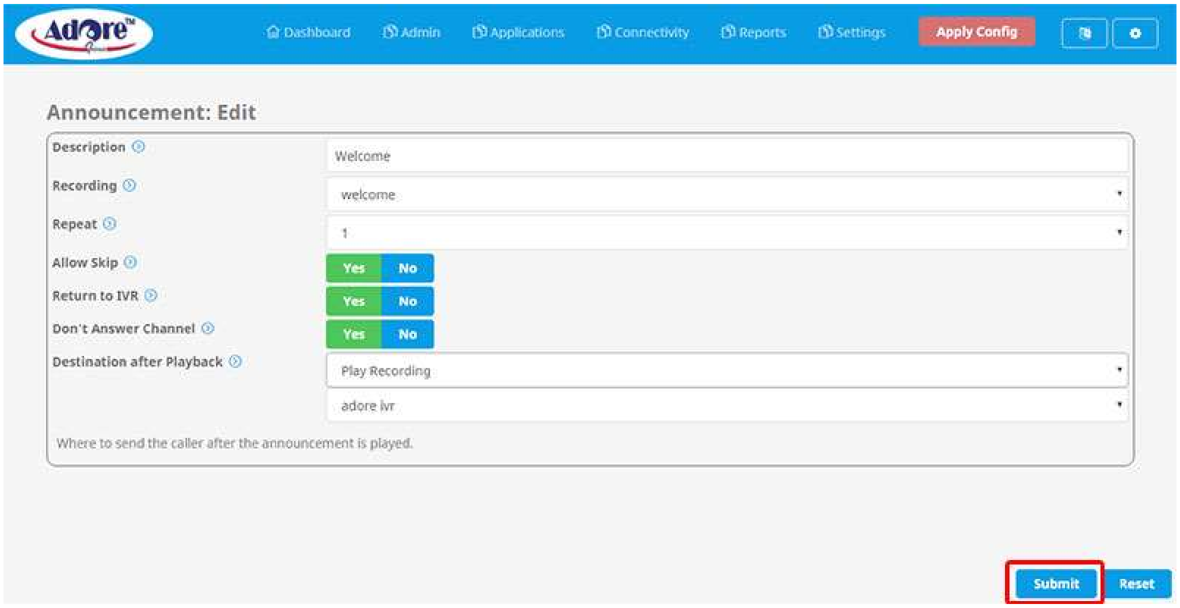


On click **Announcements** following screen will appear. The Announcements module is used to play a recording to callers and then send them to a different destination once the announcement has been played. Do not confuse Announcements with the System Recordings module. The System Recording module is 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.

Click on **+ Add** button to Add Announcements in your PBX System



You will be taken to a new windows 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.

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. Call Flow Control

Call Flow Control

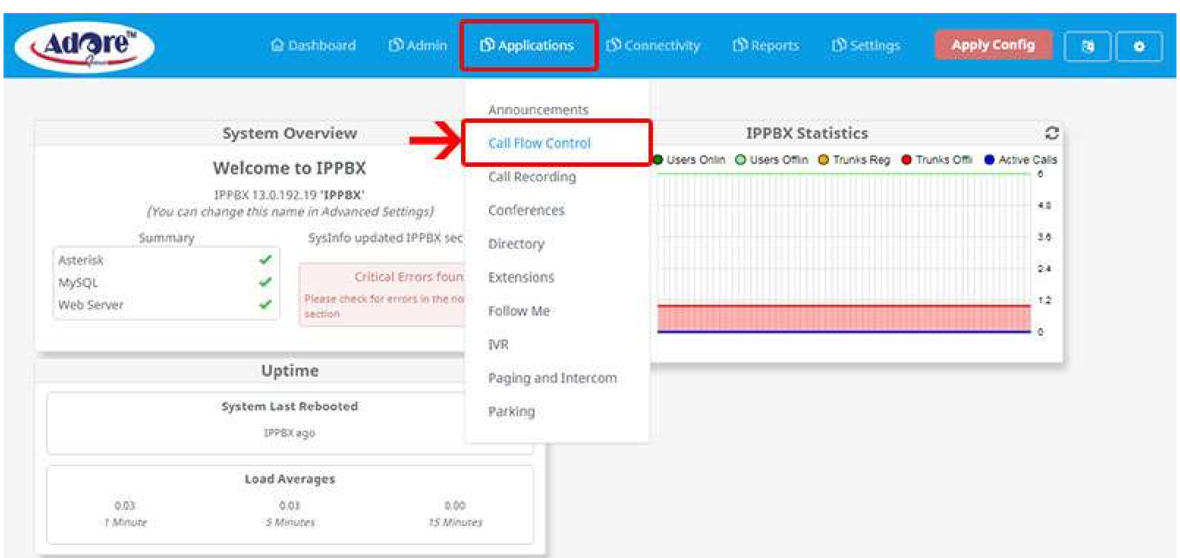
The Call Flow Control module is used to create a single destination that can act as a switch that can be toggled by anyone who has access to a local phone. It is commonly used to allow phone system users to manually switch between "Daytime Mode" and "Nighttime Mode."

Call Flow Control should not be confused with Time Conditions. While both of these modules relate to call flow, Call Flow Control is designed to be a *manual* switch, while a Time Condition is designed to be a *scheduled, automatic* switch.

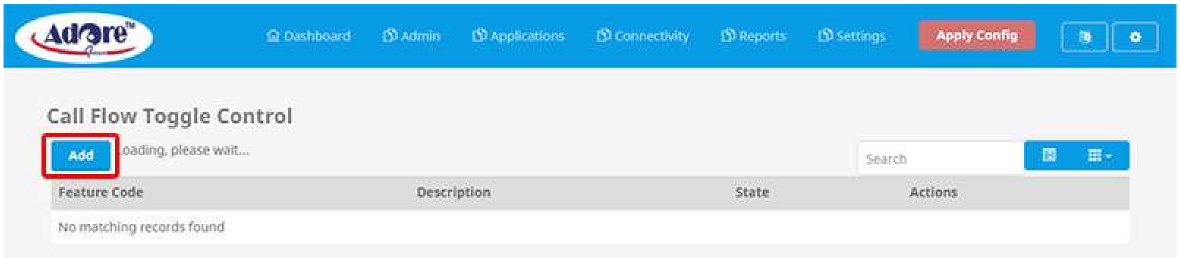
As a common usage example, you might create a Call Flow Control that is activated by dialing *280. When a system phone dials *280, the switch will toggle between **green/off** mode and **red/on** mode. In **green/off** mode, calls might route to all phones in the office for 30 seconds before going to voicemail. In **red/on** mode, the calls might ring to only the front desk, and only for ten seconds, before going to Voicemail.

The code that is used to configure the Call Flow Control (usually *280 through *289) can also be programmed to act as a Busy Lamp Field on your system phones. When configured this way, your phones will show your users whether the switch is On or Off.

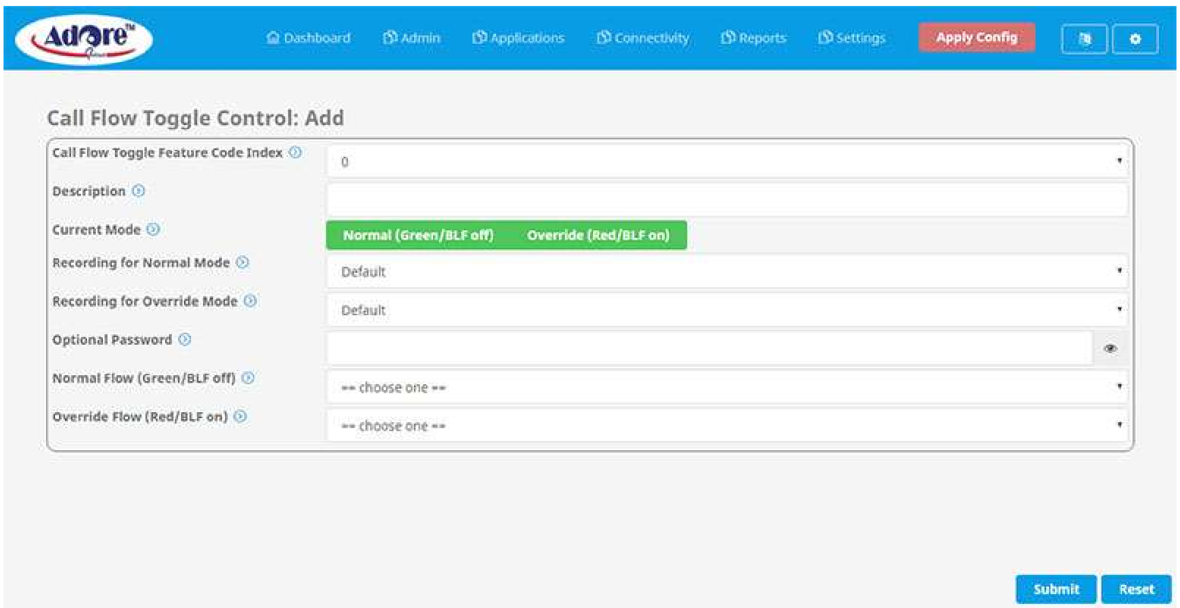
Go to **Application**-> **Call Flow Control**



On click Call Flow Control following screen will appear.



Fill in the information as described below.



Call Flow Toggle Feature Code Index

Each Call Flow control toggle feature code starts with *28. The index is the last part of the feature code, which can be 0 through 99. For example, if you choose "0" here, you would dial *280 to toggle the feature code. If you choose "99," you would dial *2899.

Description

Define a name for this toggle so you can easily identify it in the list.

Current Mode

Select the **Normal (Green/BLF off)** or **Override (Red/BLF on)** button to set the initial state for your new Call Flow Control. Later, these buttons can be used (in addition to the feature code) to change the mode.

-
    - **Normal (Green/BLF off) -** This is the normal destination that calls go to. If you have a BLF button on your phone programmed to this feature code, the light on the phone will either be off or green.
    - **Override (Red/BLF on) -** Override mode means you are not going to the normal destination. If you have a BLF button on your phone programmed to this feature code, the light on the phone will be lit (red).

Recording for Normal Mode

The default recording played when toggling into normal mode is to beep and say "feature code deactivated." You can record your own announcement in the System Recordings module and pick that recording within the Call Flow Control module to override the default recording.

Recording for Override Mode

The default recording played when toggling into override mode is to beep and say "feature code activated." You can record your own announcement in the System Recordings module and pick that recording within the Call Flow Control module to override the default recording.

Optional Password

You can optionally set a password that the user who is toggling the call flow will have to enter on their phone before they can toggle this call flow.

Normal Flow Destination

This is the destination to route the call to when in Normal (Green/BLF off) mode of the toggle. This destination can be any other module on your PBX such as an extension, announcement, or queue.

Override Flow

This is the destination to route the call to when in Override (Red/BLF on) mode of the toggle. This destination can be any other module on your PBX such as an extension, announcement, or queue.

Save

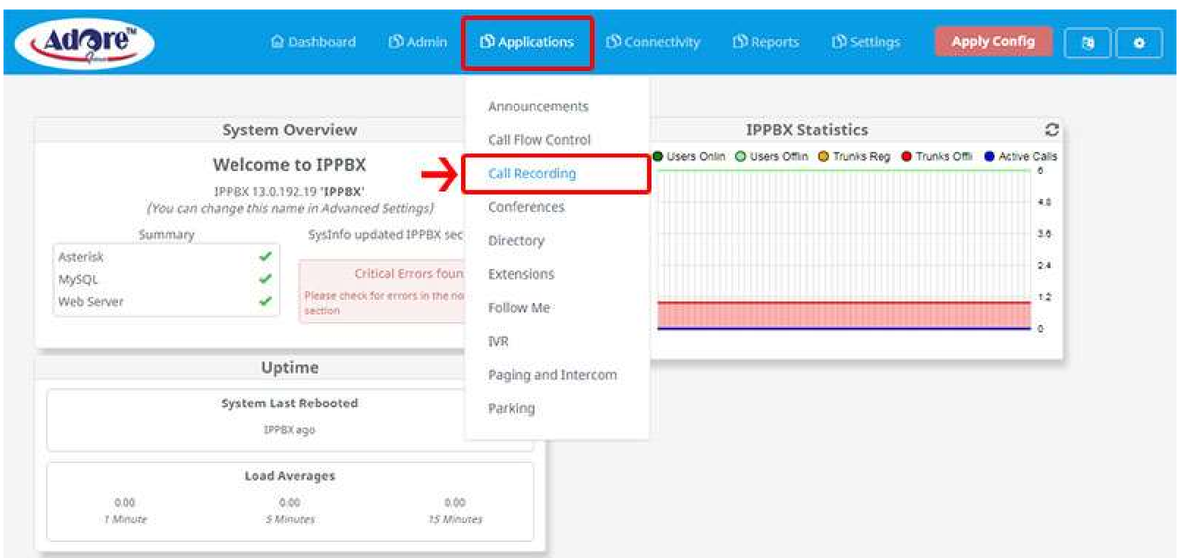
- Click the **Submit** button
- Click the **Apply Config** button

3.3. Call Recording

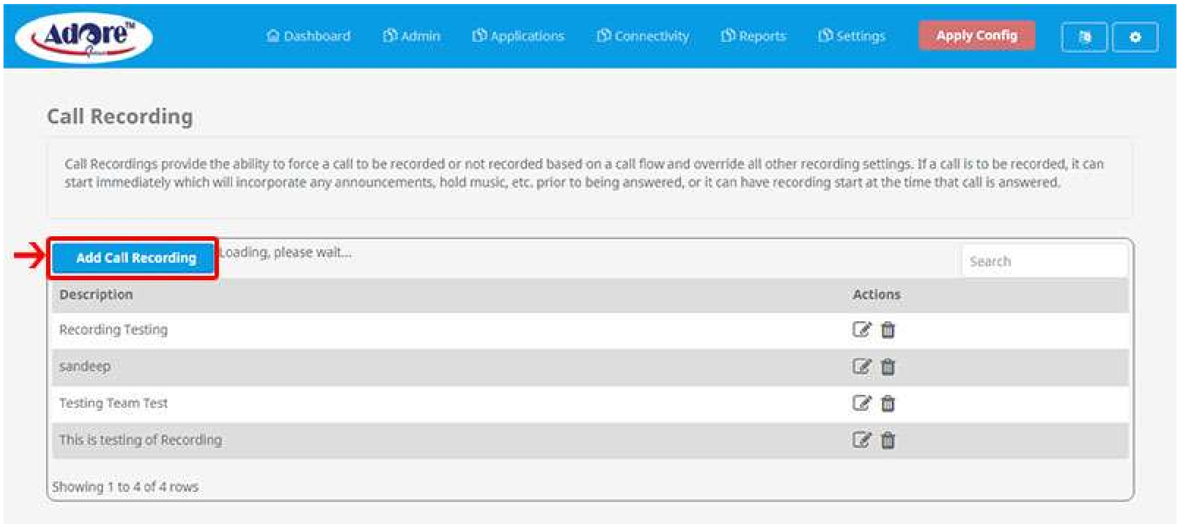
Call Recording

The Call Recording module provides the ability to force a call to be recorded or not recorded based on a call flow and override all other recording settings. If a call is to be recorded, it can start immediately. This will incorporate any announcements, hold music, etc. prior to being answered. It can also have the recording start at the time that call is answered. Several modules inside your PBX admin allow you to control call recordings directly, such as the Queues, Inbound Routes, Ring Groups and Extensions modules. The Call Recording module is designed to allow you to force a call to start recording prior to going to a specific destination that does not allow call recordings to be set, such as a page group or a specific IVR. Through various configuration levels, the Call Recording module controls permissions related to audio recording of various activities.

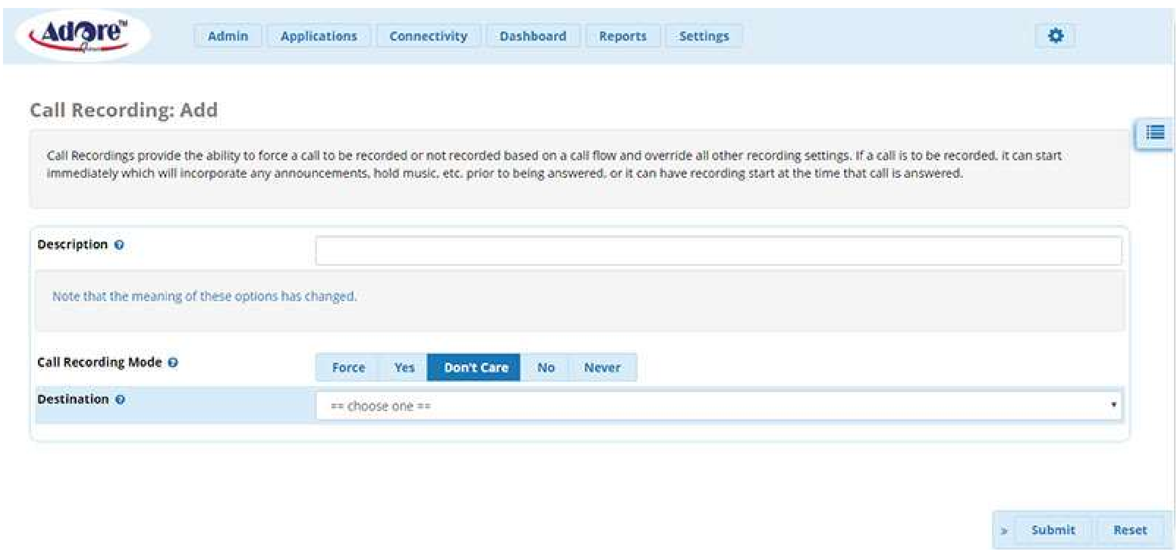
Go to **Applications** -> **Call Recording**



On Click **Call Recording** following screen will appear. Click the **Add Call Recording** button.



Enter information into the form as described below.



Description

The descriptive name of this call recording instance. For example "Support IVR Recording.”

Call Recording Mode

Controls or overrides the call recording behavior for calls continuing through this call flow.

Force

An override with a higher priority than "Yes." Changes what was already set. User cannot stop the recording unless they have an override permission. A later setting of "Never" would override a "Force."

Yes

Equal priority with "No"; does not change a "No" that was previously set. Can be overridden with Force or Never; cannot be overridden by "No."

Don't Care

Honors whatever was set earlier in the call flow, and does not affect downstream settings.

No

Equal priority with "Yes"; does not change a "Yes" that was previously set. Can be overridden with Force or Never; cannot be overridden by "Yes."

Never

An override with a higher priority than "No." Changes what was already set. User cannot start the recording unless they have an override permission. A later setting of "Force" would override a "Never."

Destination

Select the destination to send the call to after it passes through this Call Recording instance.

Save

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

3.4. Conferences

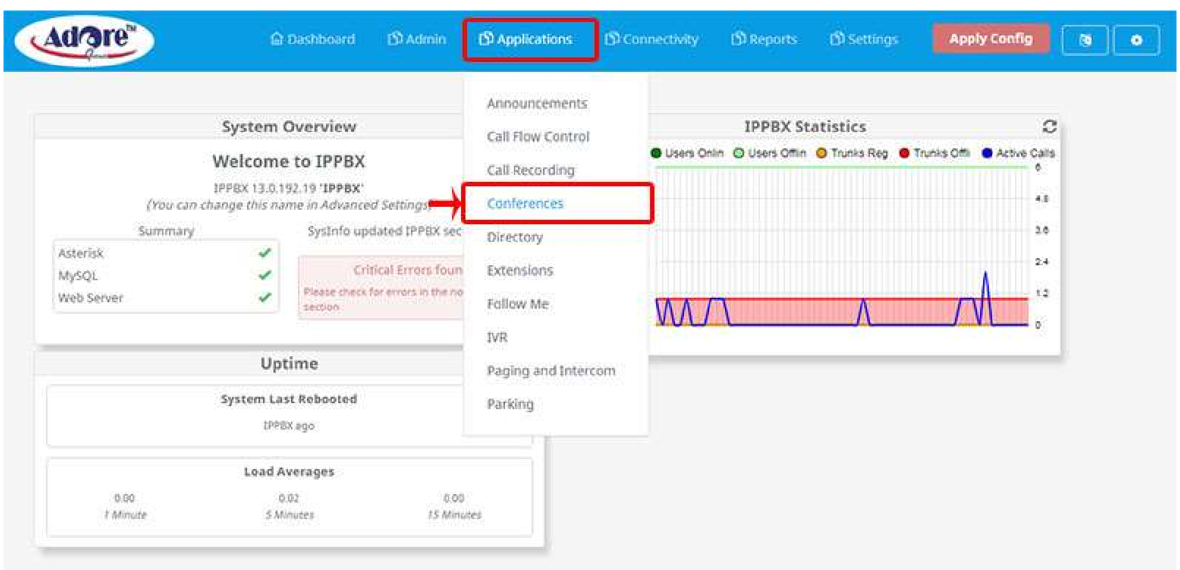
Conferences

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.

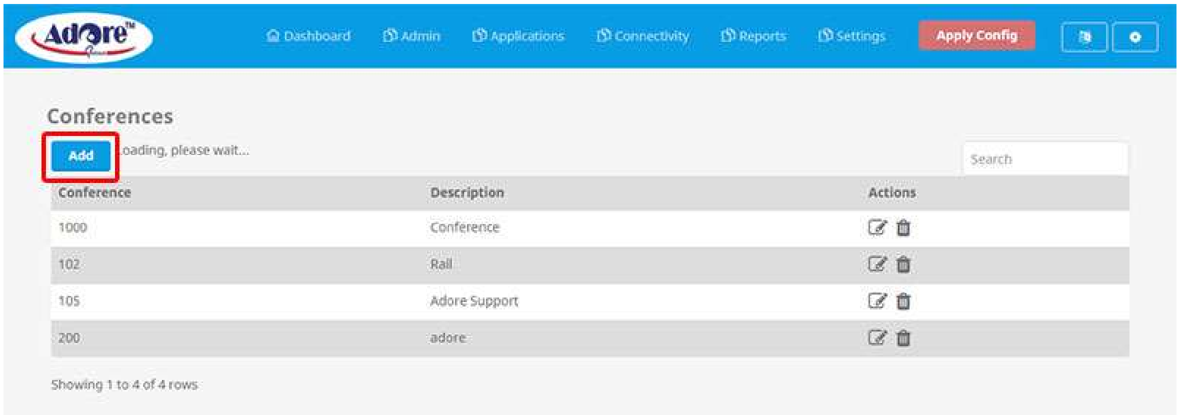
For example, you could create a Conference that will allow your local phones to dial 800, and then enter into a conference call.

Conferences Module creates a destination to which you can route calls, the Conferences Module is related to the any module that can route calls to a destination, including the Inbound Routes Module, the IVR Module, the Time Conditions Module, etc. The conference module can be enhanced through other modules such as VQ Plus.

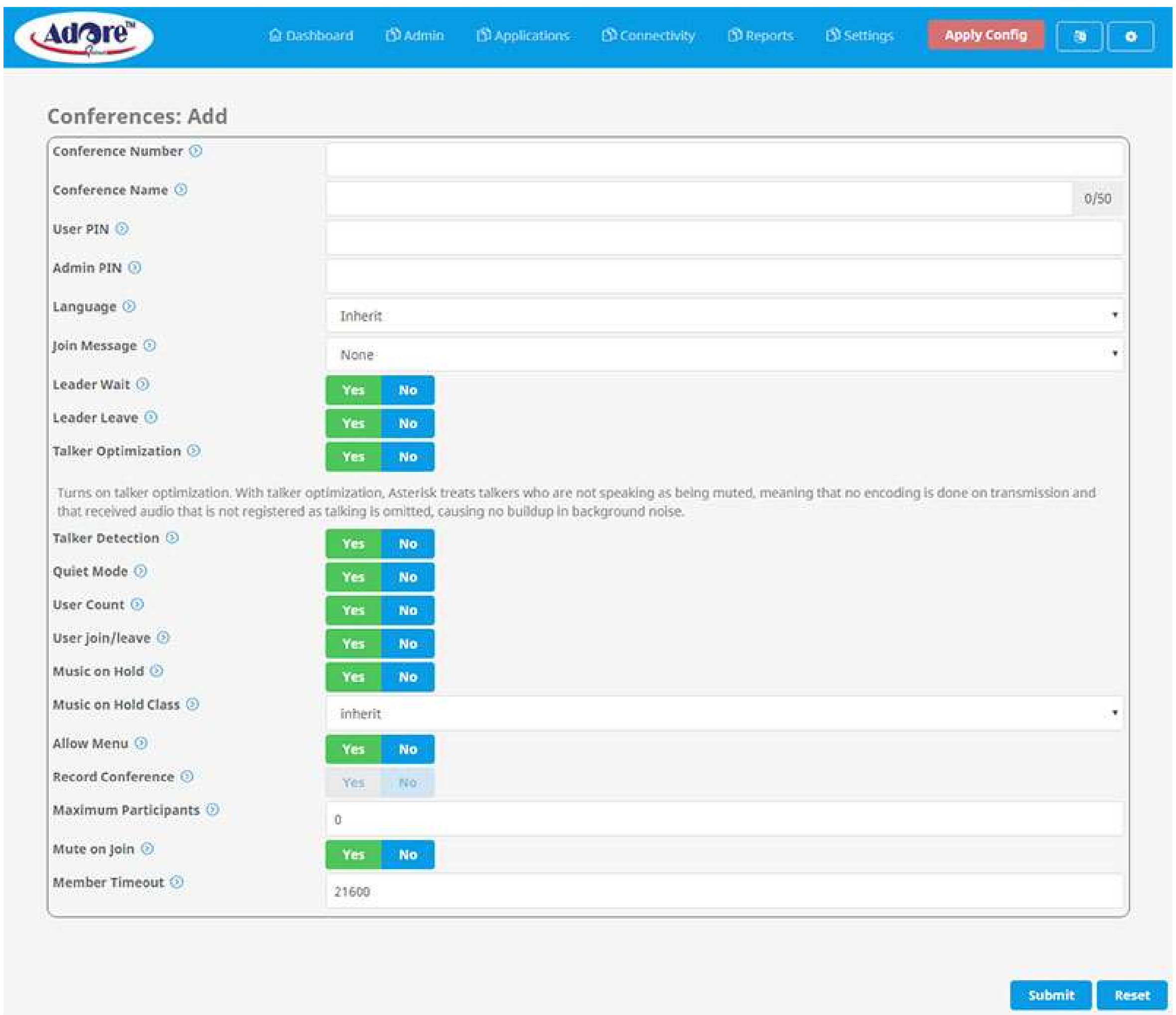
Go to **Applications** -> **Conferences**



On click **Conferences** following screen will appear. 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

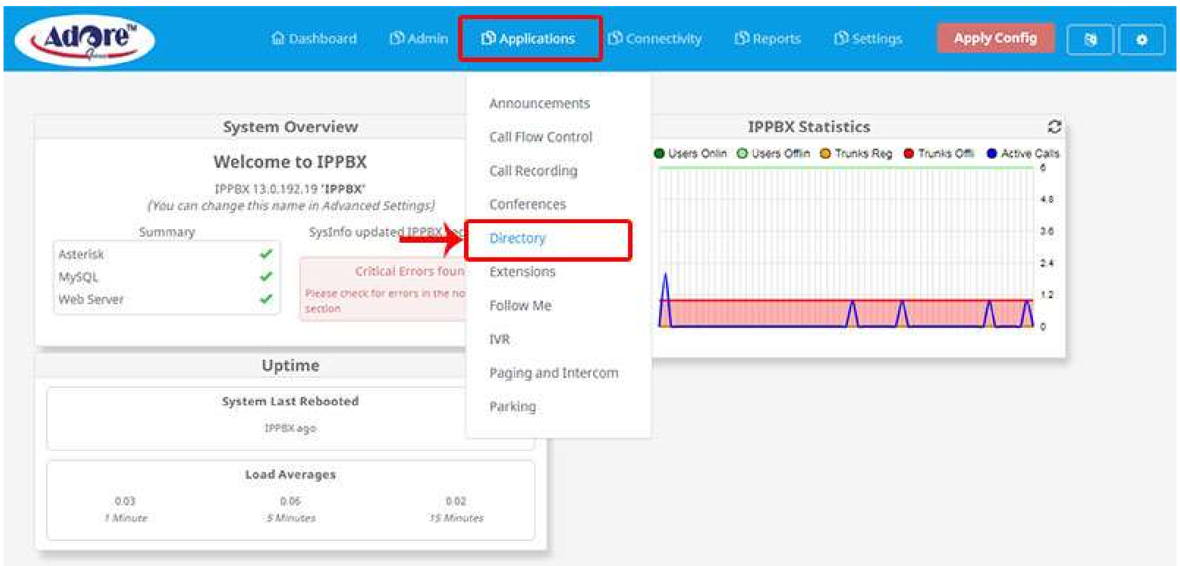
- Click the **Submit** button.
- Click the **Apply Config** button.

3.5. Directory

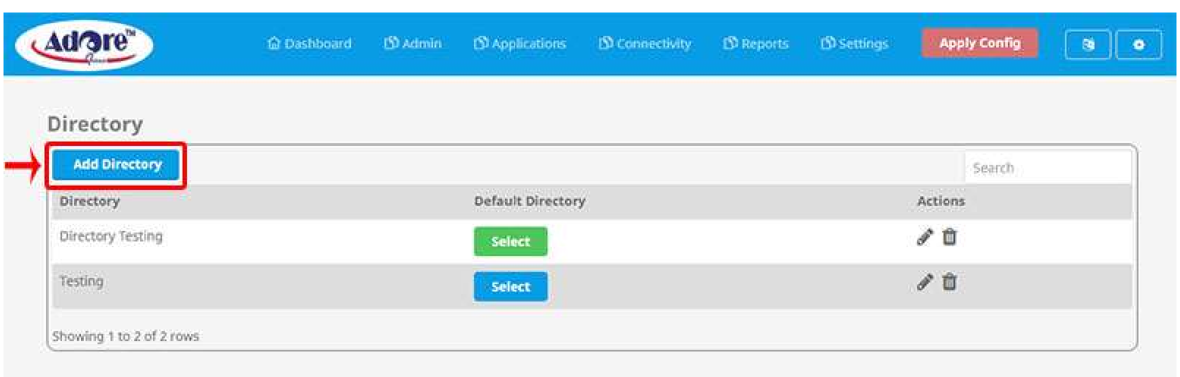
Directory

The directory module allows you to create directories of users that can be accessed by callers through modules like the IVR. You can create a company wide directory or create a directory for an individual department. You may also specify where the user goes when they choose a directory member. For example callers may look up the CEO but it dials the CEO's assistant.

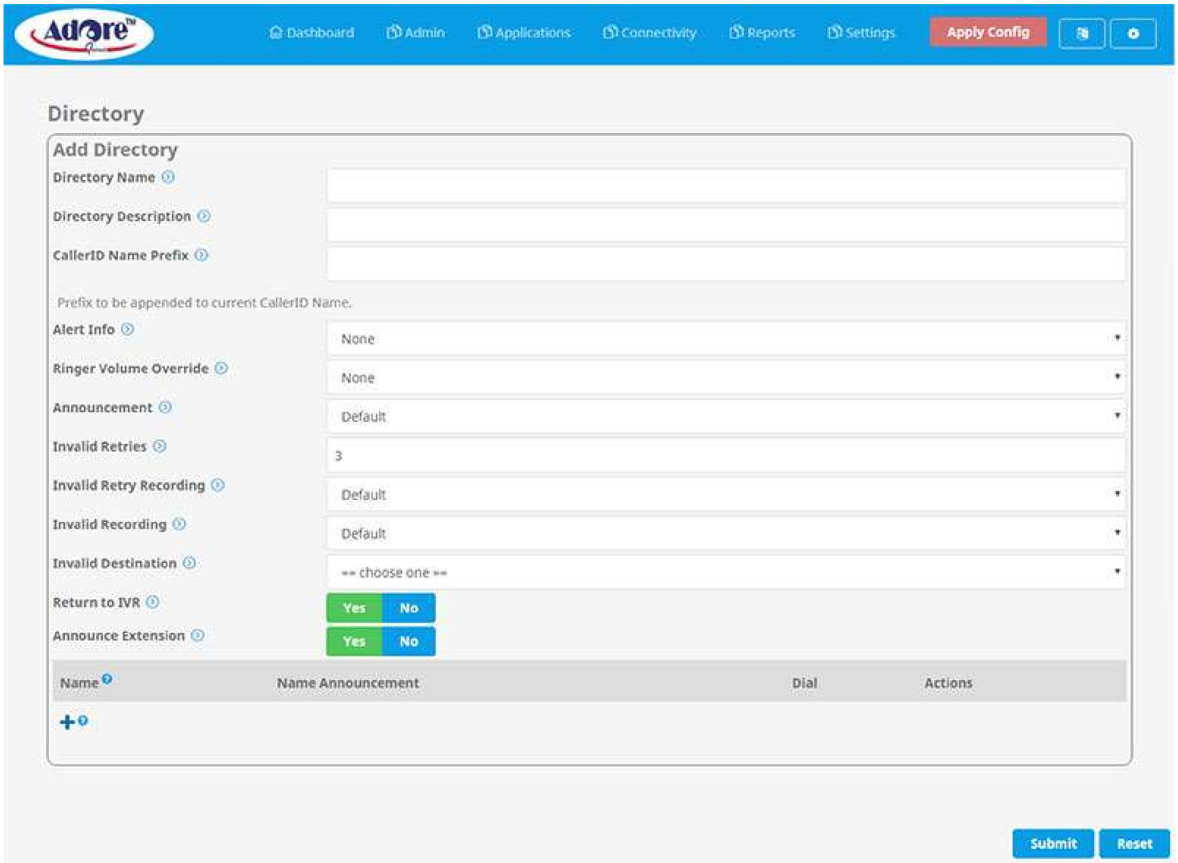
Go to **Applications** -> **Directory**



On click Directory following screen will appear. Click on "**Add Directory**" button.



Fill in the form as described below.



Directory Name

A friendly name for the directory, such as "Sales" or "Support" for example.

Directory Description

Optional - A description of the directory.

CallerID Name Prefix

Optional - A prefix added to the Caller ID name of the caller when they call through the directory. For example, you could enter "DIR-" to help users identify callers who have come from the directory.

Alert Info

Optional - ALERT_INFO to be sent to SIP devices called from this directory. Can be used for distinctive ring for SIP devices when supported.

Announcement

The greeting played to callers when they enter the directory. The default built-in announcement is "Welcome to the directory. Please enter the first three or more letters of the party's first or last name, using your touch tone keypad. Use the 1 key for punctuation. Press 0 to exit the directory and obtain further assistance." If desired, you can select a different announcement from your list of available System Recordings.

Invalid Retries

Number of times to retry when receiving an invalid/unmatched input from the caller.

Invalid Retry Recording

Prompt to be played when an invalid/unmatched response is received. The default message is "I'm sorry, there's no entry matching the keys you have entered." If desired, you can select a different announcement from your list of available System Recordings.

Invalid Recording

Message played to caller when they press 0 or when they have run out of retries, prior to sending them to the invalid destination. The default message is "We are now transferring you out of the directory. Please hold on for further assistance." If desired, you can select a different announcement from your list of available System Recordings.

Invalid Destination

Where to send callers when they run out of retries or press 0. May be overridden by the "Return to IVR" option below.

Return to IVR

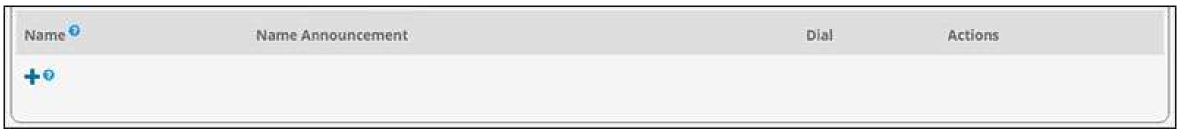
Yes/No - Whether to send the caller back to the originating IVR (if the caller came from an IVR), upon reaching the invalid retry limit or pressing 0. If the caller came from an IVR, a "Yes" setting here overrides the "Invalid Destination" set above.

Announce Extension

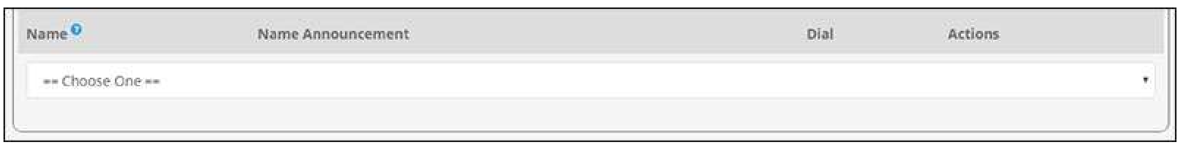
Yes/No - Whether to announce the destination's extension number to the caller prior to the transfer. This can be useful if you want to let callers know how to direct-dial an extension in the future.

Adding Directory Entries

- Choose the type of entry from the drop-down menu. **(All Users**, **Custom Entry**, or an **Extension** as described below.) To add additional entries, click the plus sign



On click Plus Sign following selection option will appear



All Users

This will add an entry for every extension.

Custom Entry

This will allow you to specify a full dial string. This is useful if you wanted to add a cell phone or external resource.

Extension

Select an individual extension to add to the list.

Save

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

3.6. Extensions

Extensions

The Extensions Module is used to set up each extension on your system. In the Extensions module, you will set up the extension number, the name of the extension, the password, voicemail settings for the extension, and other options.

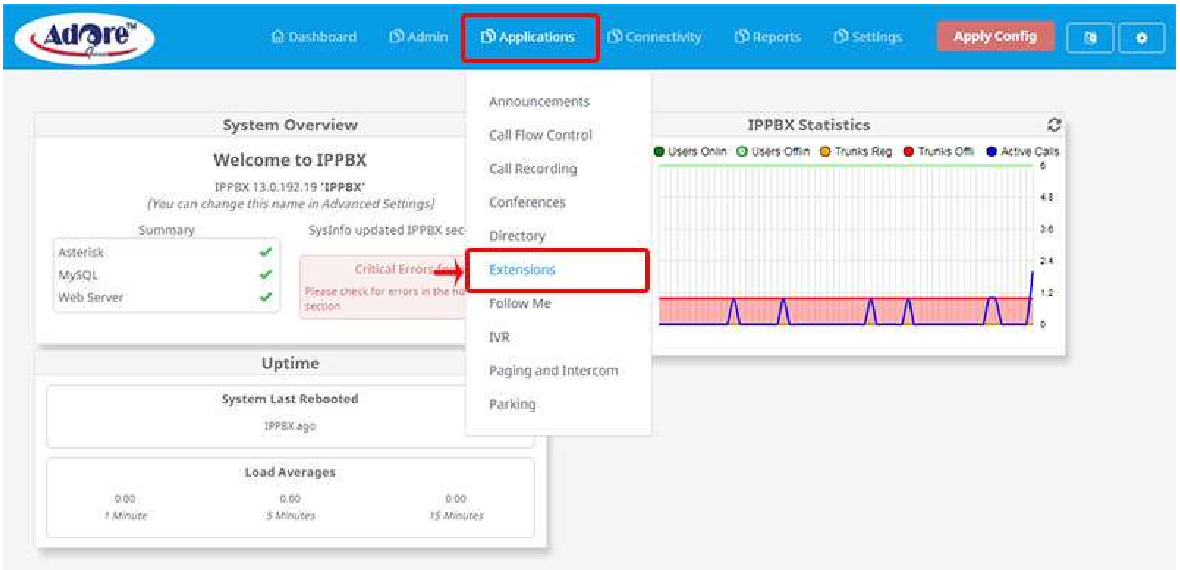
Normally, each physical phone will be assigned to one extension. If you have a phone that has more than one "line" button, you would normally make each line button register to the same extension number, and then use the line buttons to manage multiple calls to and from the same line. However, you could also create two or more extensions and assign each extension to a different line button.

The Extensions Module works together with any module that can route a phone call, including the Inbound Routes Module, the Ring Groups Module, the Queues Module, and the Paging Module. The Extensions Module also works together with the Follow Me Module, because each extension can have its own Follow Me options.

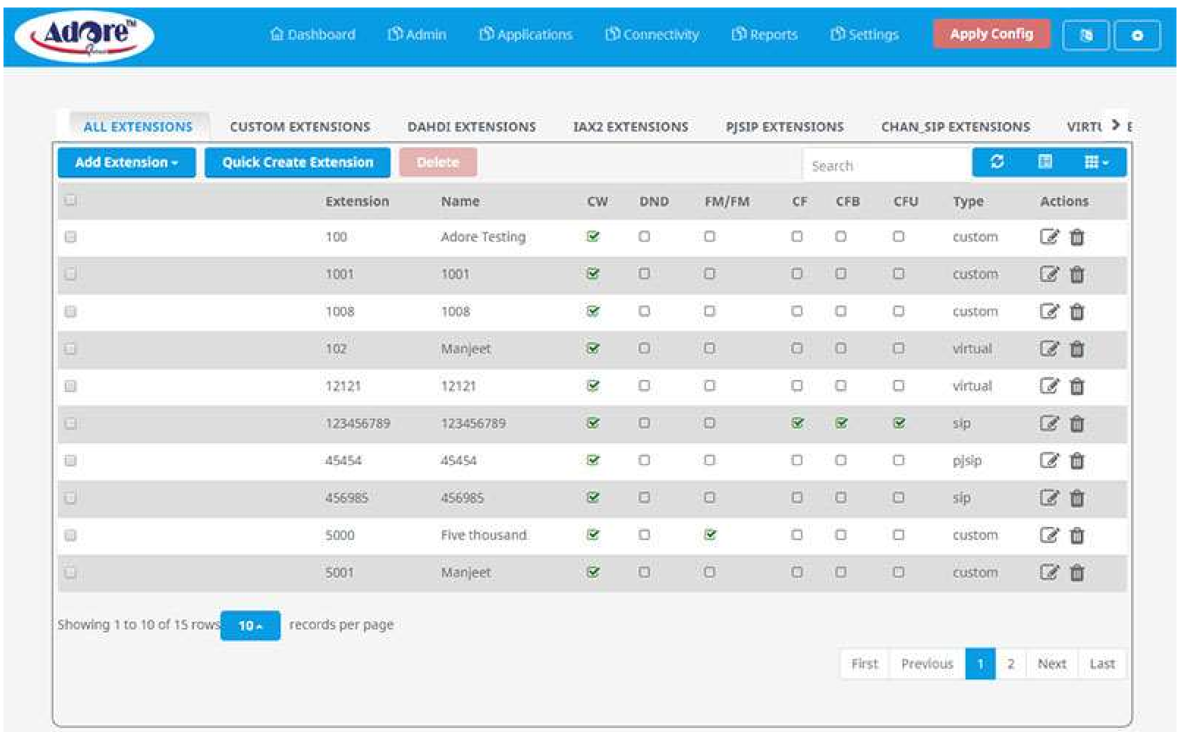
The Extensions Module is also related to the Advanced Settings Module. In the Device Settings section of the Advanced Settings Module, you can change a number of the default settings that will apply when you create a new extension. In addition, the Advanced Settings Module can be used to enable Device and User Mode. When Device and User Mode is enabled, the Extensions Module will disappear and be replaced with two separate modules called "Devices" and "Users."

The Extensions Module is related to the User Management Module. In the User Management Module, a user may have a "primary linked extension."

Go to **Applications** -> **Extensions**



On click **Extensions** following screen will appear.



Viewing/Editing an Extension

- Click on the pencil icon for the extension you wish to view.
- Make the desired changes. For explanations of each type of extension, please view the appropriate wiki (Custom, DAHDI, IAX, Chan_SIP, or Virtual).
- Click the **Submit** button.

- Click the **Apply Config** button.

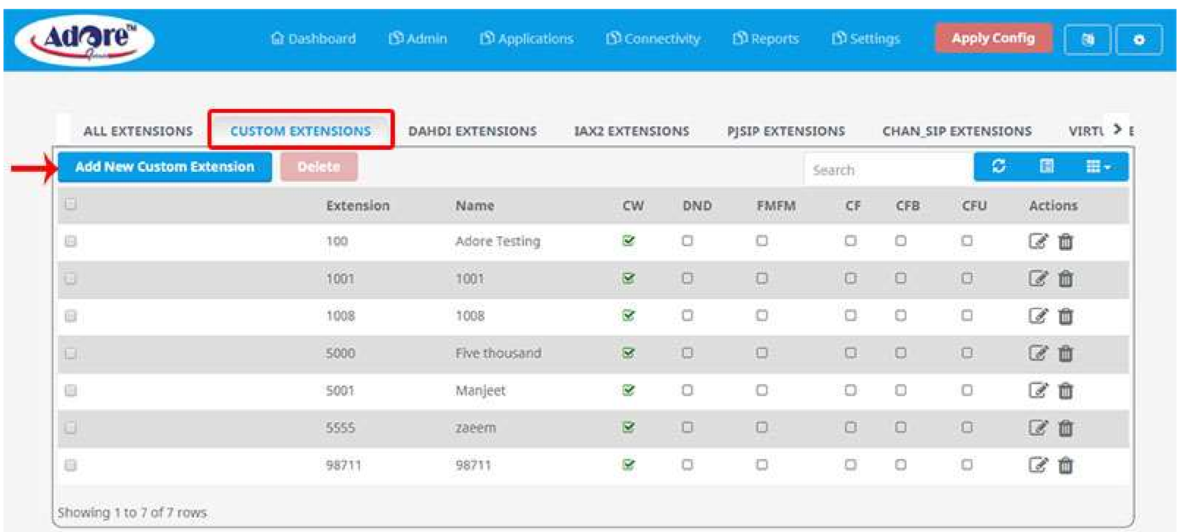
Deleting an Extension

From the extensions list:

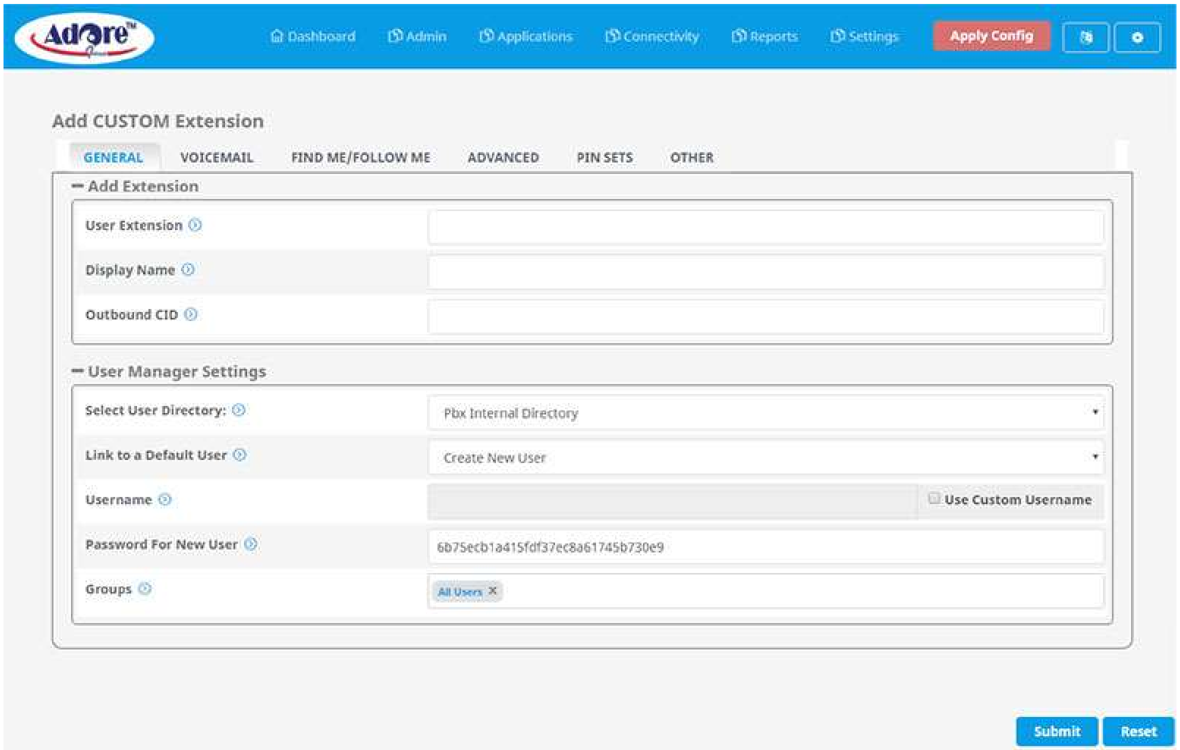
- Click the trash icon for the extension you wish to delete.
- In the alert window that appears, click the **OK** button to confirm the deletion.
- Click the **Apply Config** button.

CUSTOM EXTENSIONS

From the Extensions landing page click on the **Add New Custom 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.

User Manager Settings

Link to a Default User

Select a user that this extension should be linked to in the User Management module. Default = Create New User. An extension may only be linked to one user, and a user may only be linked to one extension.

-
  - **Create New User**: Creates a new user in User Manager that will be linked to this extension.
  - **None:** The extension will not be associated with a user in User Manager
  - **Existing User**: If you have any existing users in User Manager who are not already linked to an extension, you can select one here in order to link the user to this extension.

Username

If **Create New User** is selected above, this will be the new user's login username. If you leave the Username field is blank (grayed-out), the username will be the same as the extension number. To customize the username, check the **Use Custom Username** box and enter a username.

Password for New User

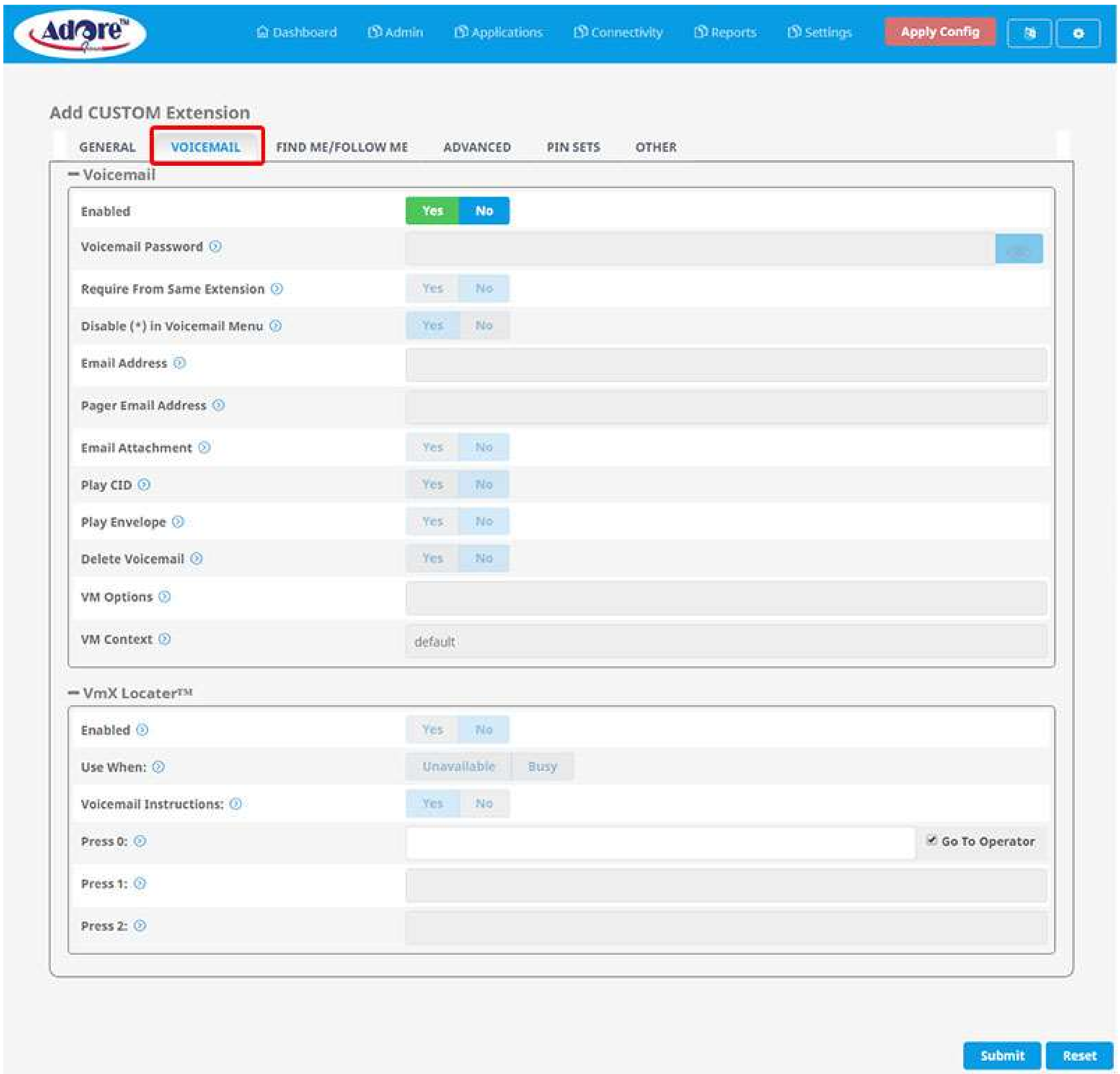
If **Create New User** is selected above, this will be the auto-generated user's new password. A password is automatically generated, but you can edit it here.

Groups

If **Create New User** is selected above, or if you are linking this extension to an existing user, you can add the user to one or more groups. Groups are defined in the User Management module, so if you haven't created any groups, none will show up here. To add this user to a group, click inside the field, and available groups will show up in a menu. You can start typing to quickly find a group. Click on a group name to add it to the field. Repeat the process if you with to enter multiple groups.

VoiceMail

- Click on the **Voicemail** tab.
- If you wish to enable voicemail, click the **Yes** button next to **Enabled** in order to allow editing the options below.



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

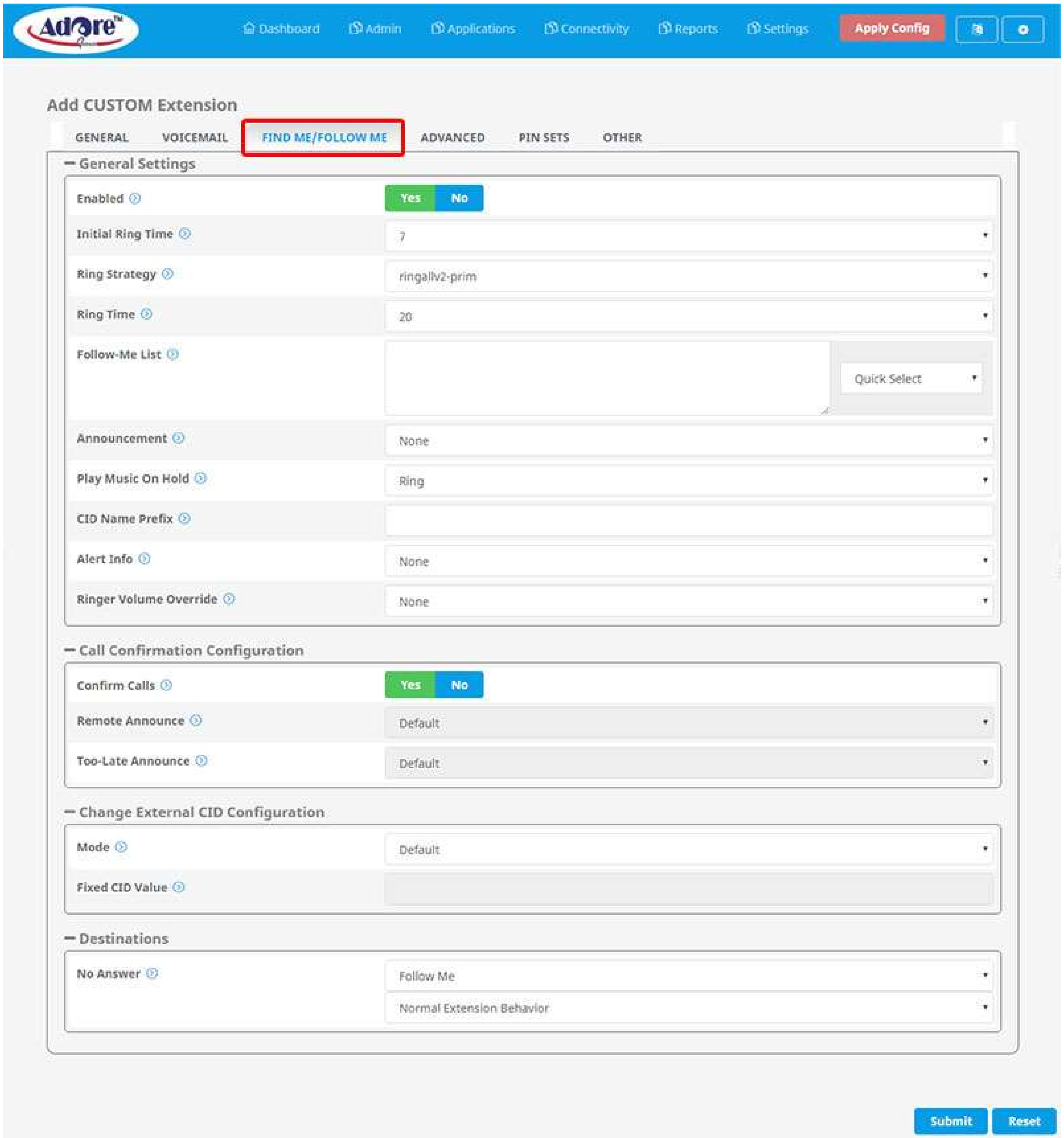
Optional - Enter a destination to send the caller to when they press 1. This can be an internal extension, ring group, queue, or external number such as a cell phone number.

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 do-not-disturb (DND) mode, it won't be rung. If the primary is 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 (see Outbound Routing).

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. To add additional recordings, please use the System Recordings module.

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

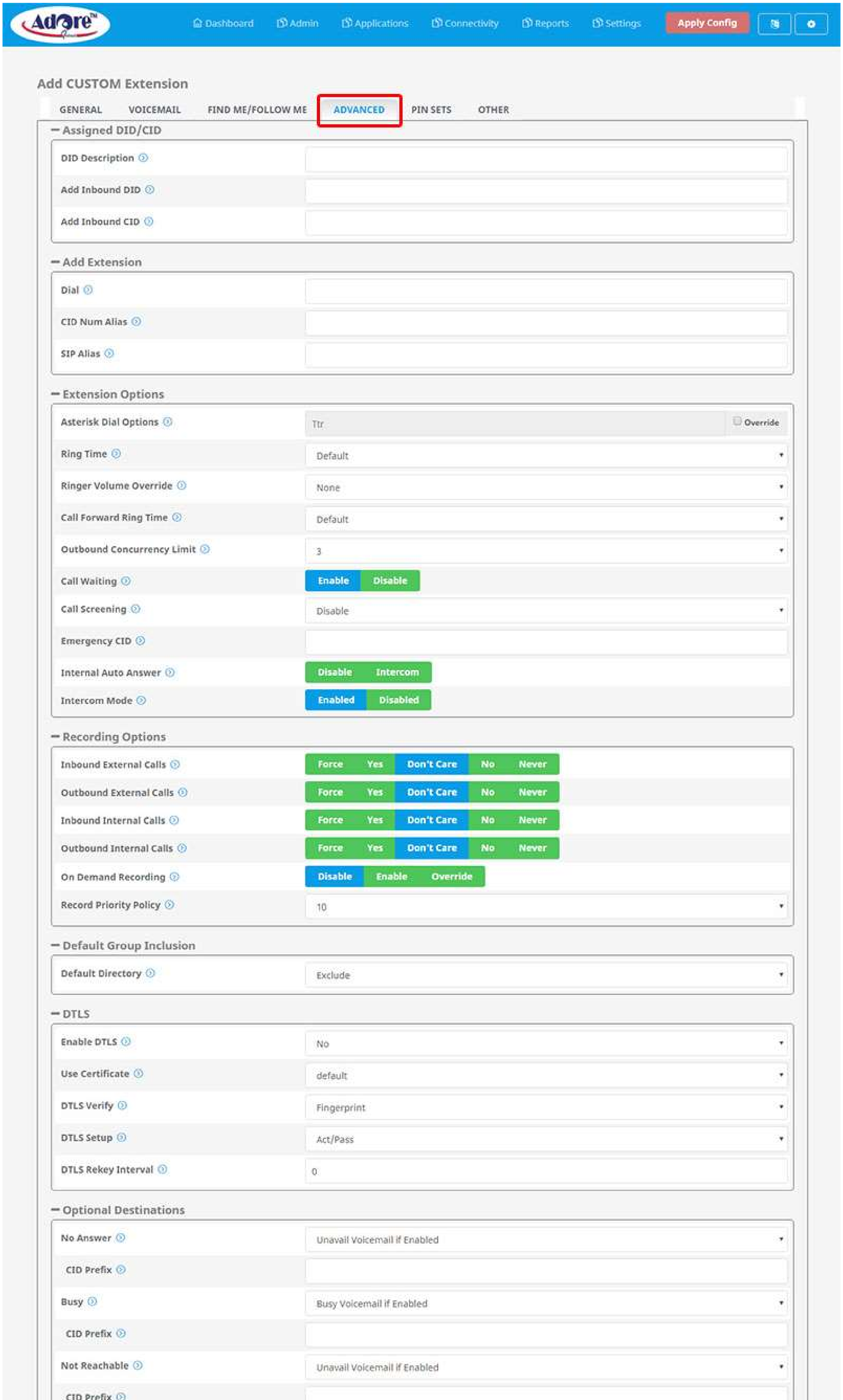
No Answer

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.


Adore
Dashboard
Admin
Applications
Connectivity
Reports
Settings
Apply Config
Add CUSTOM Extension
GENERAL
VOICEMAIL
FIND ME/FOLLOW ME
ADVANCED
PIN SETS
OTHER
Assigned DID/CID
DID Description
Add Inbound DID
Add Inbound CID
Add Extension
Dial
CID Num Alias
SIP Alias
Extension Options
Asterisk Dial Options
Ttr
Override
Ring Time
Default
Ringer Volume Override
None
Call Forward Ring Time
Default
Outbound Concurrency Limit
3
Call Waiting
Enable
Disable
Call Screening
Disable
Emergency CID
Internal Auto Answer
Disable
Intercom
Intercom Mode
Enabled
Disabled
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
Default Group Inclusion
Default Directory
Exclude
DTLS
Enable DTLS
No
Use Certificate
default
DTLS Verify
Fingerprint
DTLS Setup
Act/Pass
DTLS Rekey Interval
0
Optional Destinations
No Answer
Unavail Voicemail if Enabled
CID Prefix
Busy
Busy Voicemail if Enabled
CID Prefix
Not Reachable
Unavail Voicemail if Enabled
CID Prefix

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

Dial

How to dial this device. This will be device specific. For example, a custom device which is really a remote SIP URI might be configured such as SIP/joe@somedomain.com.

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 ringgroup 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 users 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 voicemail. **Default** will use the 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 not return, it will just continue to ring. **Default** will use the current Ring Time. If voicemail is disabled and their is not destination 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.

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

Disable/Enable: 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 when they are. Setting this to 'Ignore State' will make the Queue ignore all state information thus always trying to contact this member. Certain side affects can occur when this route is taken due to the nature of how Queues handle Local channels, such as 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.

Use Certificate

The Certificate to use from Certificate Manager

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

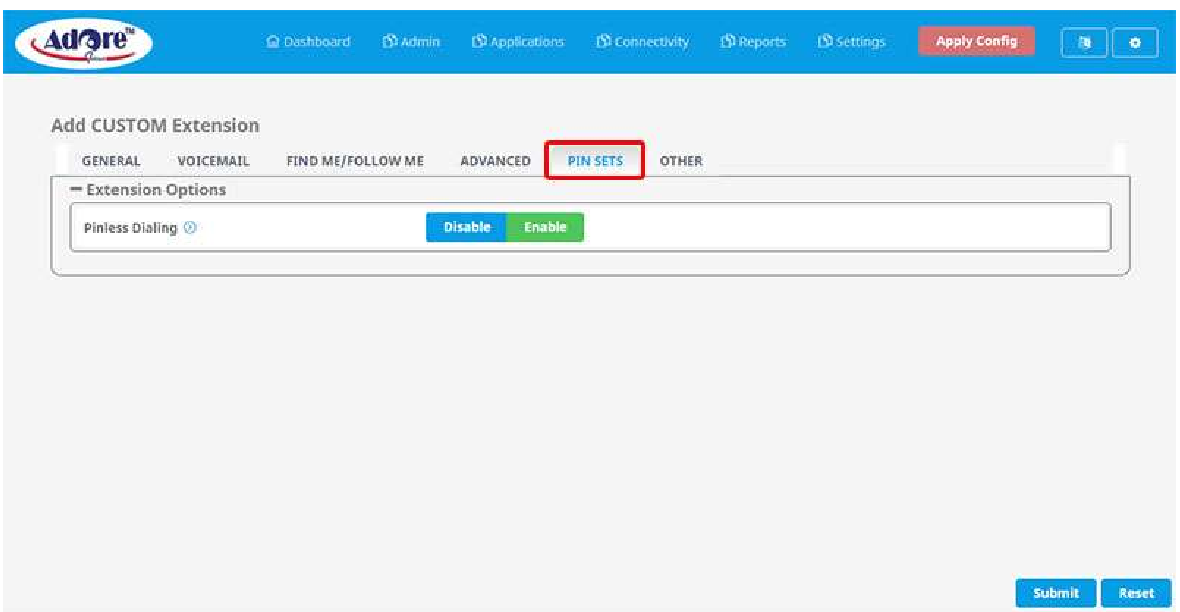
Optional destination the call is routed to when the phone is offline, such as a softphone currently off or a phone unplugged.

CID Prefix

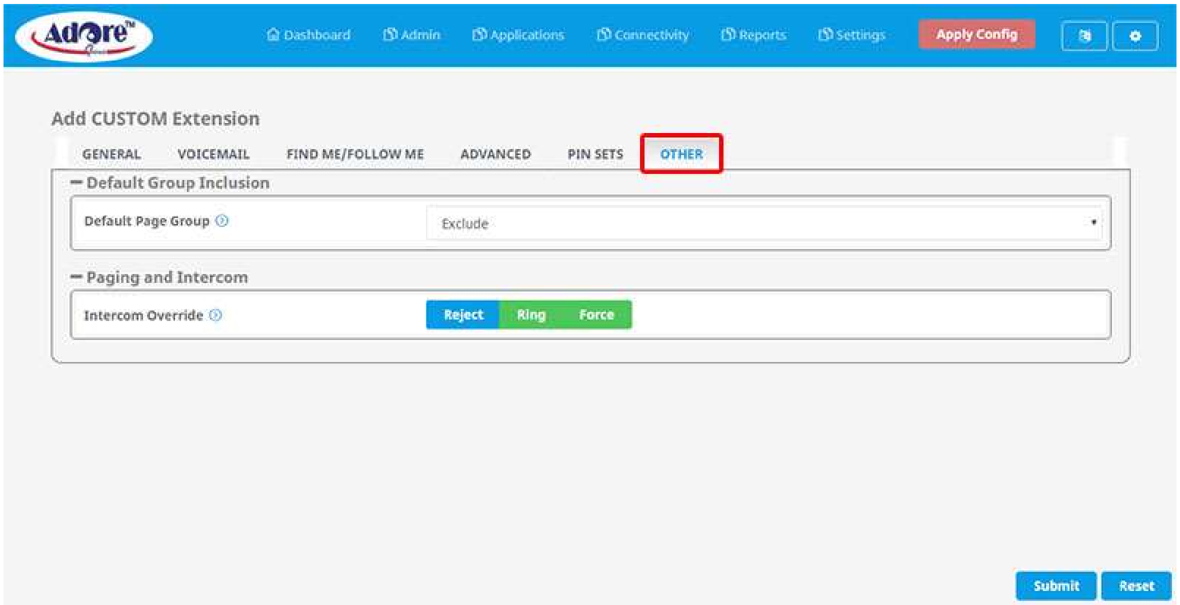
Optional CID Prefix to add before sending to this not reachable destination.

PIN SETS

Enabling Pinless Dialing will allow this extension to bypass any pin codes set under routes password required on outbound calls. Youneed to just Disable or Enable the option.



Other



Saving the Extension

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

3.7. Follow Me

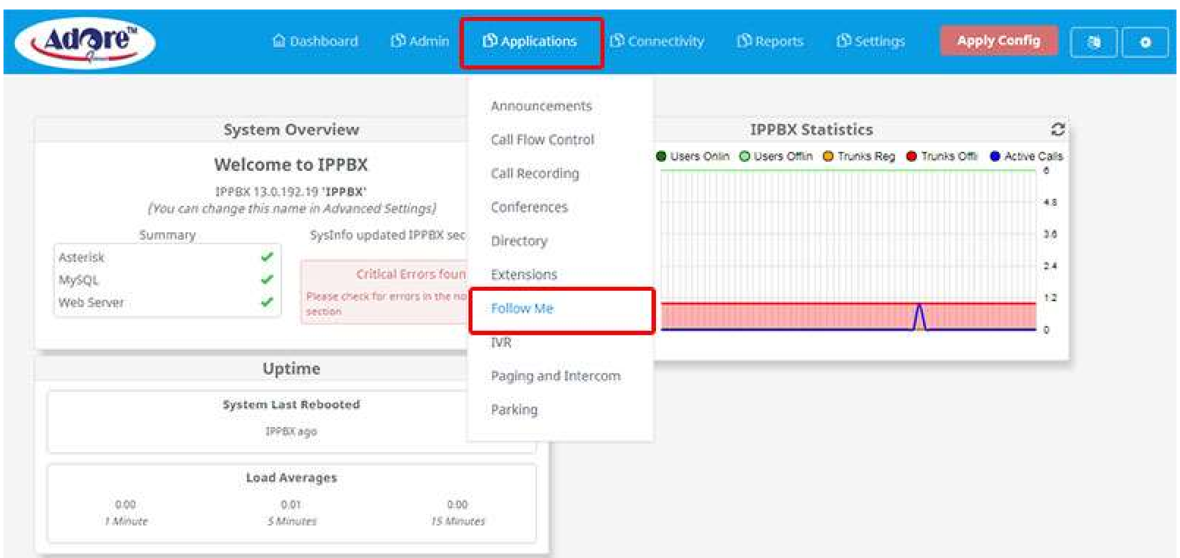
Follow Me

Follow Me (also known as **Find Me / Follow Me** or **FMFM**) allows you to redirect a call that is placed to one of your extensions to another location. You can program the system to ring the extension alone for a certain period of time, then ring some other destination(s), such as a mobile phone or a related extension, and then go to the original extension's voicemail if the call is not answered. Follow Me can also be used to divert calls to another extension without ringing the primary extension.

Follow Me can be used in connection with VMX Locater to allow a caller who reaches voicemail to Press 1 and be diverted to the Follow Me call flow.

Your users can also modify certain Follow Me settings using the User Control Panel (UCP 12+), and can disable and enable Follow Me using a feature code defined in the **Feature Codes** module.

Go to **Applications** -> **Follow Me**



On click **Follow Me** following screen will appear.

Enabling Follow Me

- Click the **Yes** button for the extension.
- An alert window will pop up to advise you that follow me has been enabled. Click **OK** to acknowledge the alert. Note: there is no submit or apply config button.

Disabling Follow Me

- Click the **No** button for the extension.
- An alert window will pop up to advise you that follow me has been disabled. Click **OK** to acknowledge the alert. Note: there is no submit or apply config button.

Editing Follow Me Settings for an Extension

- Click the extension. This will take you to the Find Me / Follow Me tab of the **Extensions** module.
- Edit settings as desired. These settings are explained in the relevant user guide for the Extensions module: DAHDI, IAX2, Other (Custom), PJSIP, SIP, and Virtual.
- Click the **Submit** button in the Extensions module.
- Click the **Apply Config** button in the Extensions module.

3.8. IVR

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."

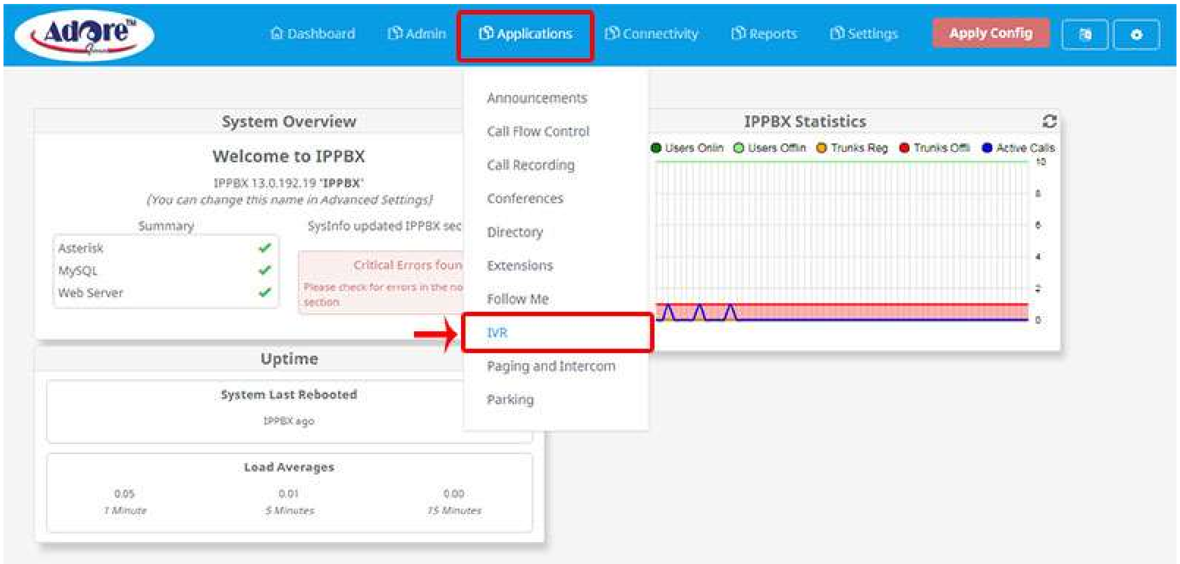
The IVR module plays messages that you record or upload in the System Recordings module.

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

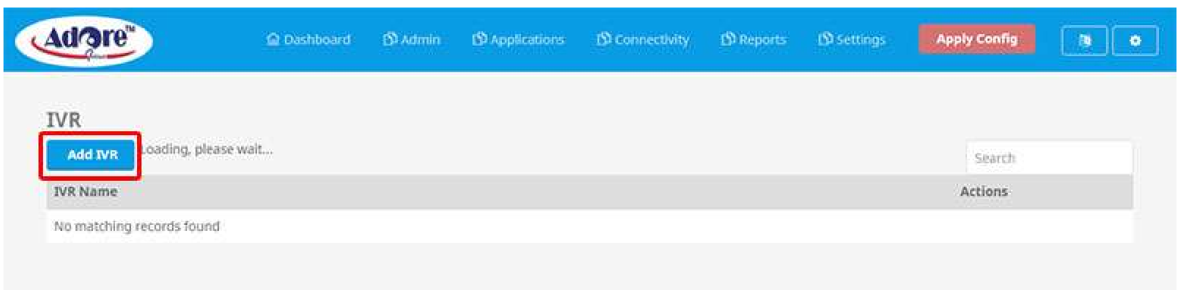
The IVR Module also works together with any module that can act as a destination, because the IVR Module is used to route calls.

The IVR will only show destination options if they have already been defined in the PBX. More destination options will appear in the IVR as new destinations are configured in the system. (For example, new system announcements or conference rooms.)

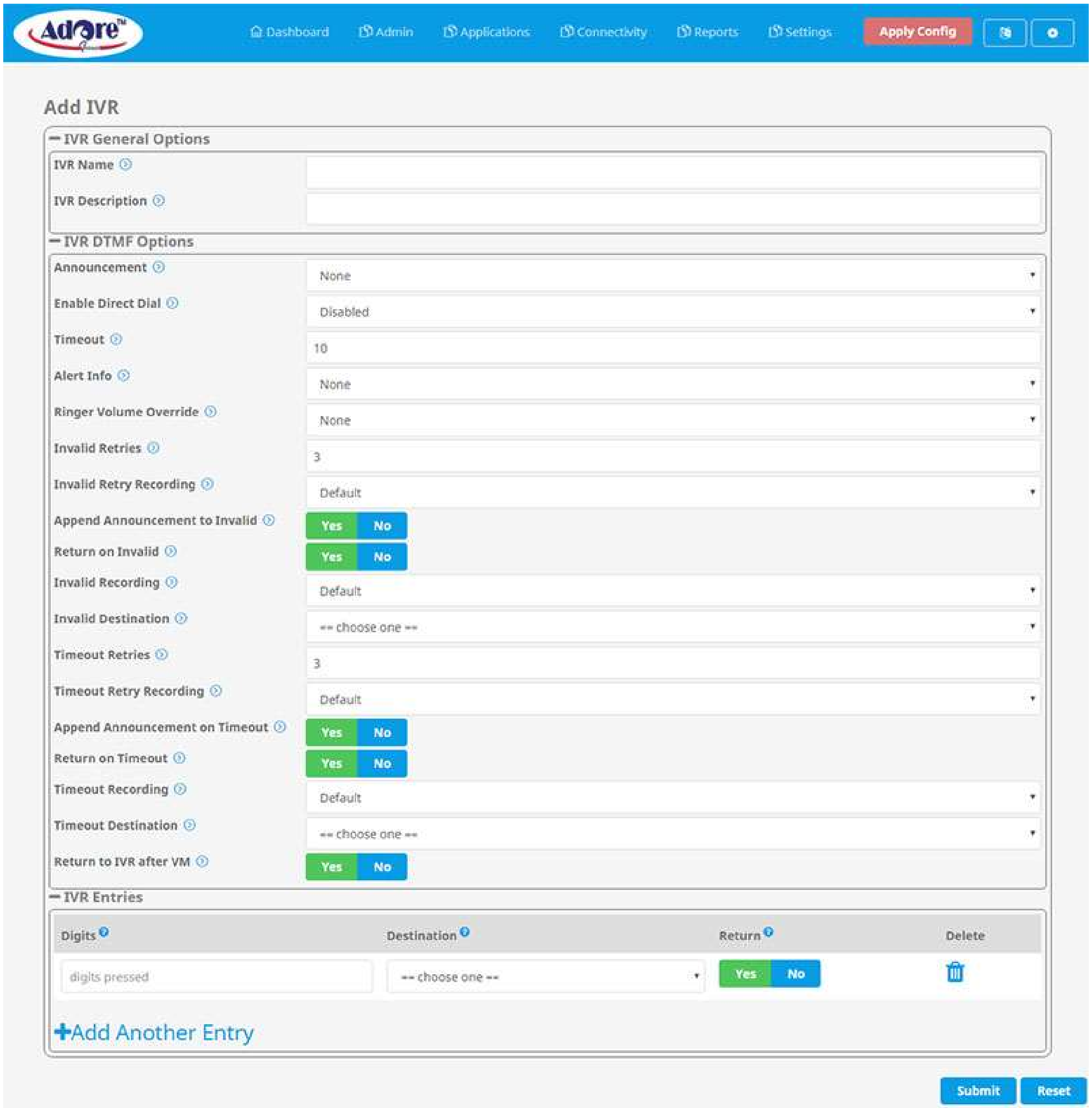
Go to **Applications** -> **IVR**



On click **IVR** following screen will appear. 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.

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."

You can add hidden options for things like remote voicemail access for your employees. "Hidden" means the option exists but you don't provide verbal instructions for it in your announcement to callers. Note: the caller could still "guess" the option, so be sure to use difficult-to-guess options and/or use password protection if security is an important consideration.

For example, you could create an Ext entry of "98." For the destination, you could choose “Feature Code Admin” and "Dial Voicemail <*98>" to dial voicemail. Do not prompt the caller with this option in your announcement. Inform your users that in the main IVR, they can dial 98 to access their voicemail. They will then be prompted to enter the extension number and the voicemail password.

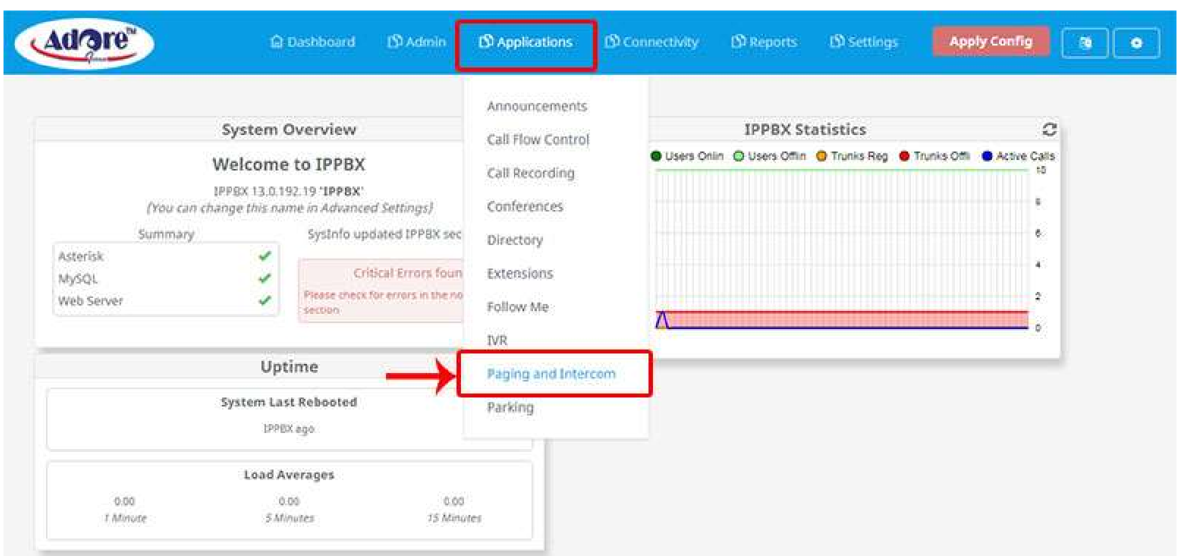
3.9. Paging and Intercom

Paging and Intercom

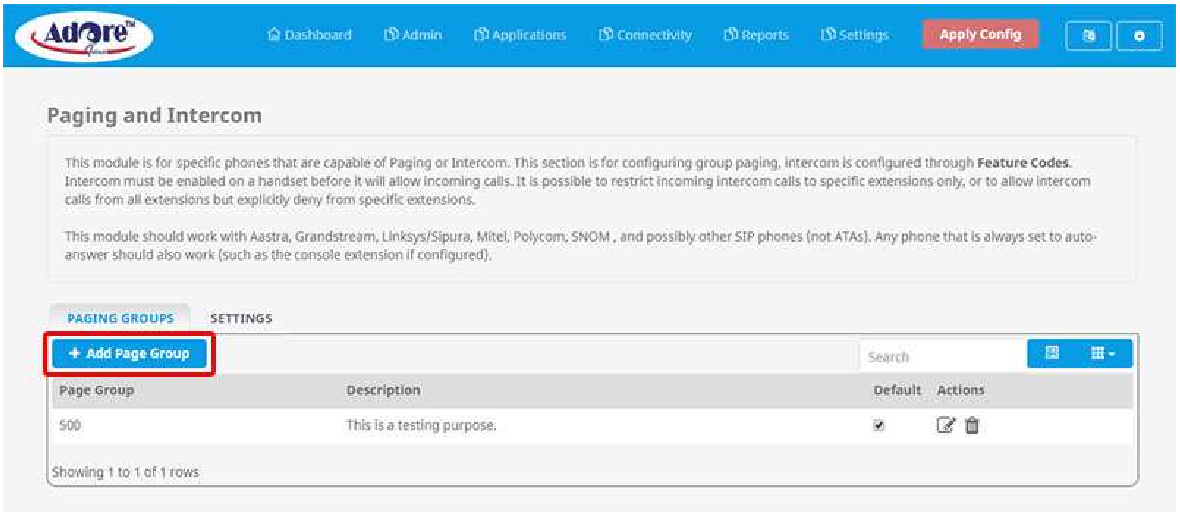
This module is for specific phones that are capable of Paging or Intercom. This section is for configuring group paging, intercom is configured through Feature Codes. Intercom must be enabled on a handset before it will allow incoming calls. It is possible to restrict incoming intercom calls to specific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

This module should work with Aastra, Grandstream, Linksys/Sipura, Mitel, Polycom, SNOM , and possibly other SIP phones (not ATAs). Any phone that is always set to auto-answer should also work (such as the console extension if configured).

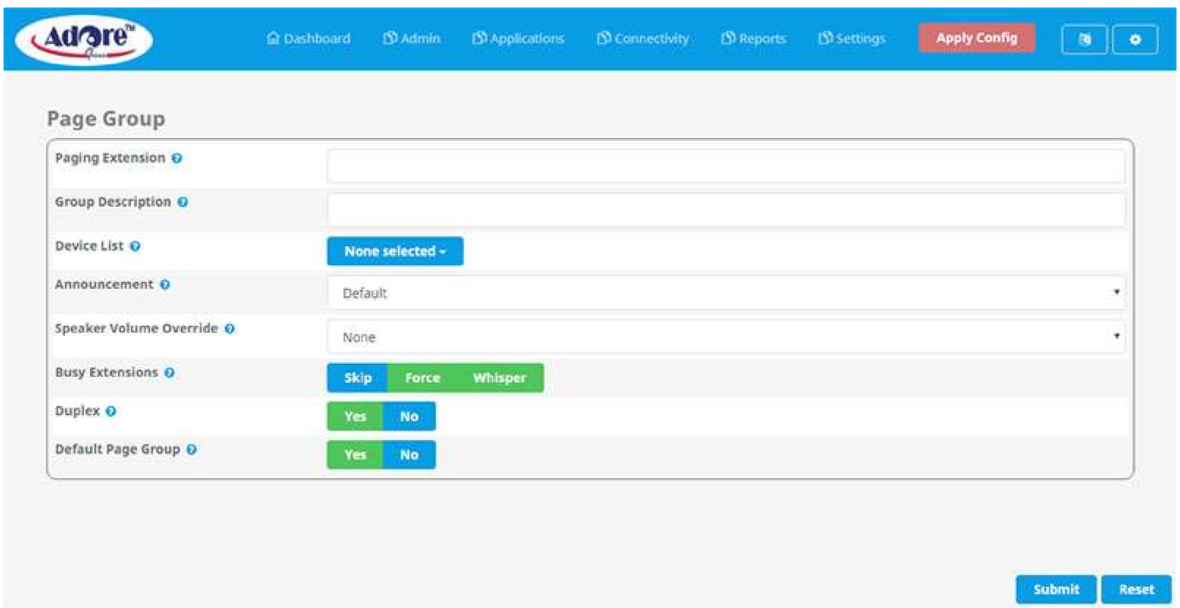
Go to **Applications** -> **Paging and Intercom**



On click **Paging and Intercom** following screen will appear. Click on **+Add Page Group** button.



On click **+Add Page Group** you will see following page, fill the information and click on **submit** button.



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.

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.

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.

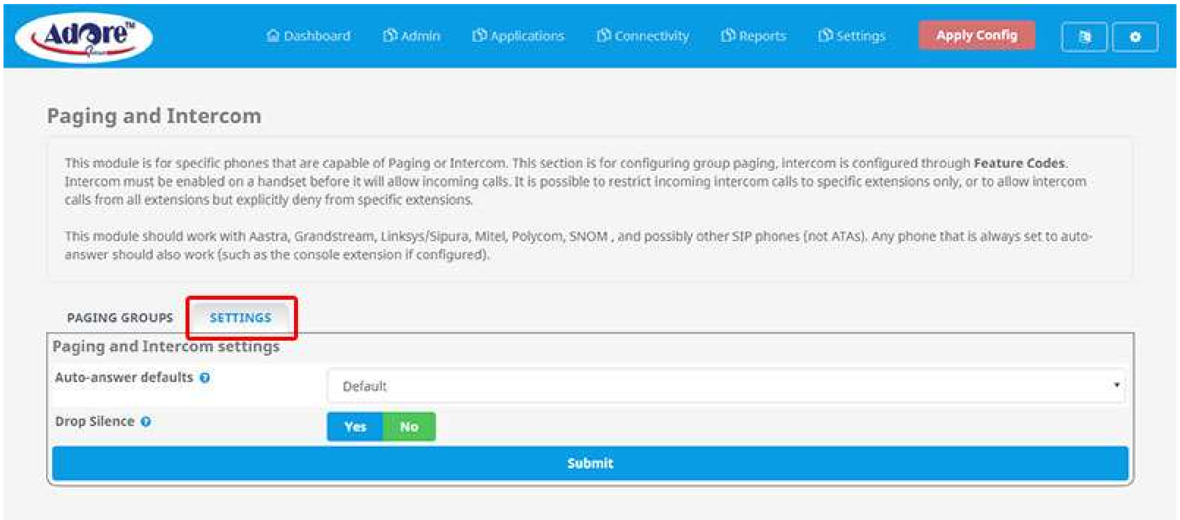
Any user can dial *1 to un-mute themselves at any time, regardless of whether Duplex is enabled here.

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.

Settings

Click the Settings tab to set global settings for page groups



Auto-answer defaults

This is the announcement to be played to the remote party when their phone answers a page. The default is a beep. You can also select a system recording. Select None if you do not want to play a beep or other sound.

Save

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

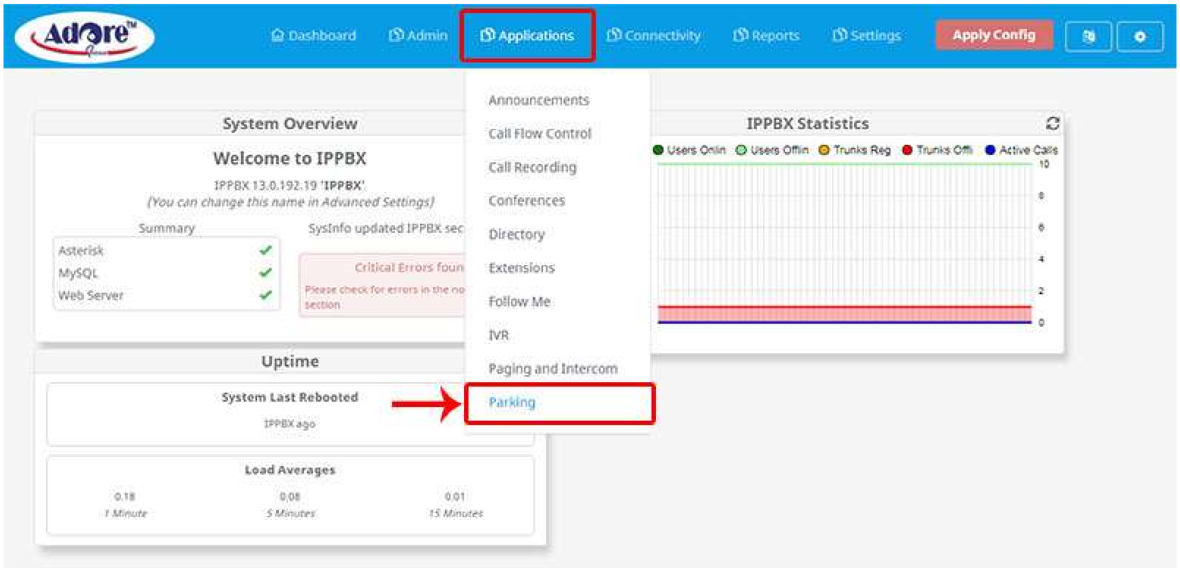
3.10. Parking

Parking

This 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. The standard module allows for the configuration of a single parking lot available to all phones on the system while the Parking Pro module allows multiple parking lots to be configured as well as other features discussed below. 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 and announced to the ‘parker.’ The slot can be dialed from other phones to retrieve the parked call or if the call times out and is not retrieved in a timely manner, the parked call can be configured to ring back to the parker or sent to other destinations configured in the system. Parking can be greatly enhanced by programming a phone’s BLF buttons to the configured parking slots as well as when used in conjunction with visual tools like XactView operator panels and modules like Phone apps.

When combined with Paging Pro, Parking Pro offers a Park and Announce capability that provides a very powerful and automated way of parking a caller. Park and Announce allows for a call to be parked in conjunction with an automatic announcement sent to the configured Page Group. The parked call can then be announced to that page group with a configured message, the parking slot that the caller is parked in and an optional recorded message or name by the caller being parked. This ability can be manually triggered just like normal parking, by transferring a caller into a Park and Announce extension, or can be fully automated by directing a call flow, usually from an inbound route or IVR, to a Park and Announce destination.

Go to **Applications** -> **Parking**



On click **Parking** following screen will appear.

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.

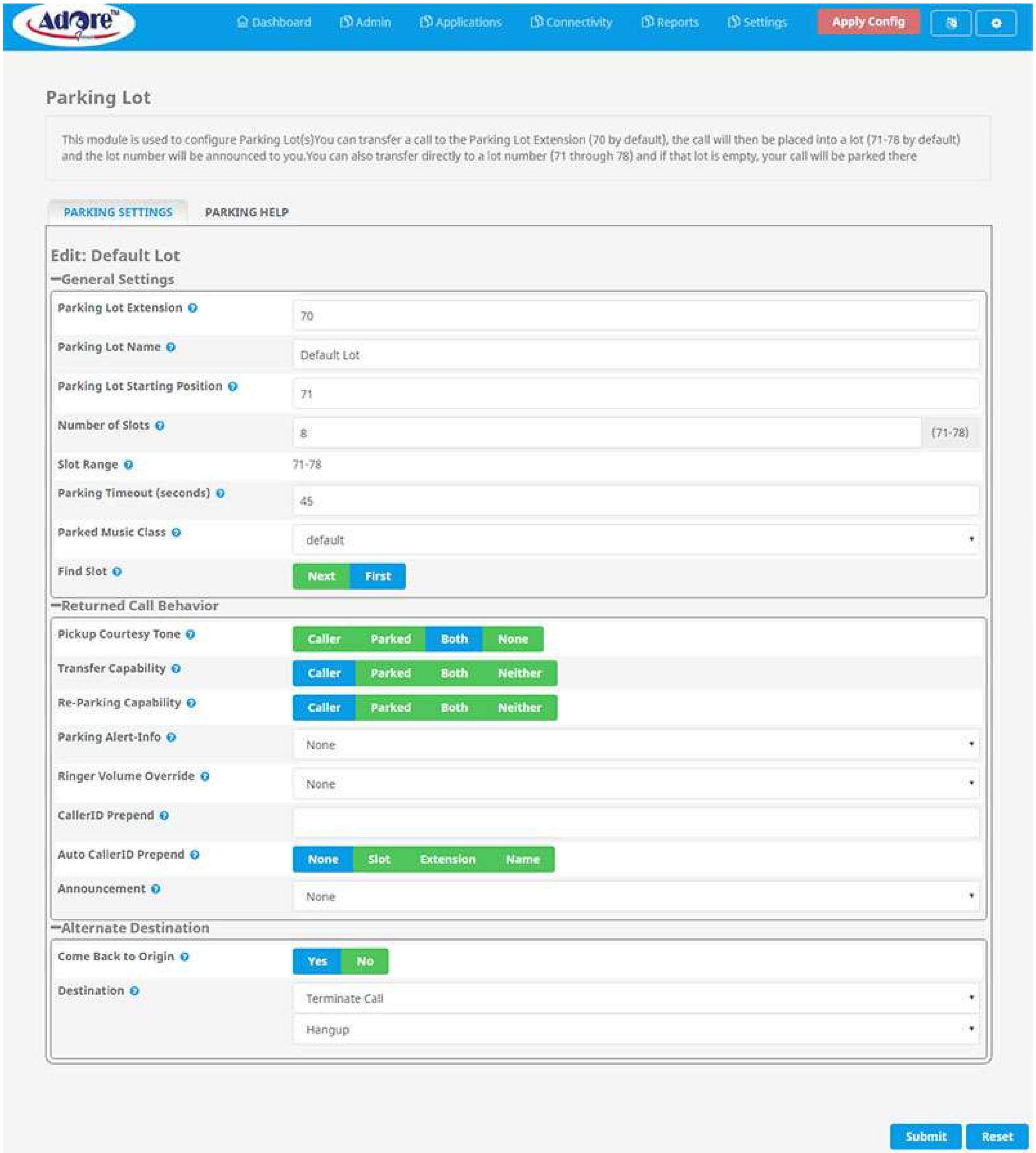
The most important items to configure with parking are:

- 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.



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.

BLF Capabilities

Yes/No: Whether to enable busy lamp field (BLF) capabilities. Each parking slot can have an Asterisk BLF “hint” associated with the parking slot. This allows a phone to have buttons programmed to the parking slots. When a call is parked in that slot, the BLF light will illuminate. You must select **Yes** if you want hints to be enabled.

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.

4. Connectivity

Connectivity

Here you can handle connectivity setting of PBX system.

- **Inbound Routes**
- **Outbound Routes**
- **Trunks**

4.1. Inbound Routes

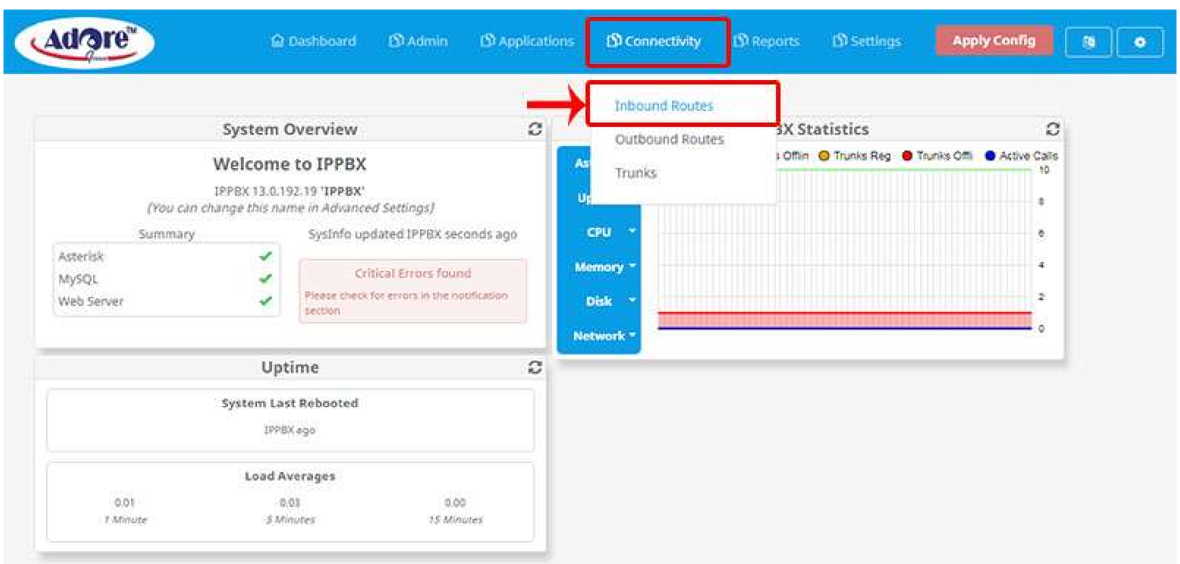
Inbound Routes

When a call comes into your system from the outside, it will usually arrive along with information about the telephone number that was dialed (also known as the "DID") and the Caller ID of the person who called.

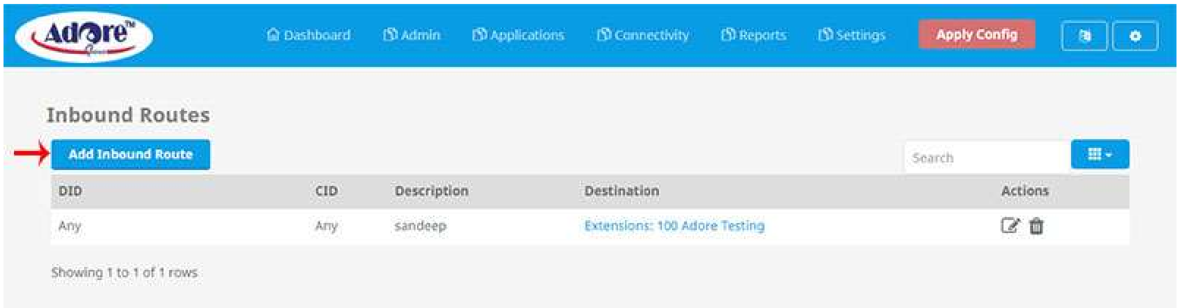
The Inbound Routes module is used to tell your system what to do with calls that come into your system on any trunk that has the "context=from-trunk" parameter in the PEER details.

The Inbound Routes module works together with most other modules. Calls come into your system on trunks that are configured in the Trunks module. The Inbound Routes module then tells the PBX where to send these calls. Calls can be sent to a variety of destinations, including an Extension, Ring Group, Queue, IVR, Time Condition, Feature Code, DISA, Conference, etc., all of which are configured in their own modules.

Go to **Connectivity** -> **Inbound Routes**

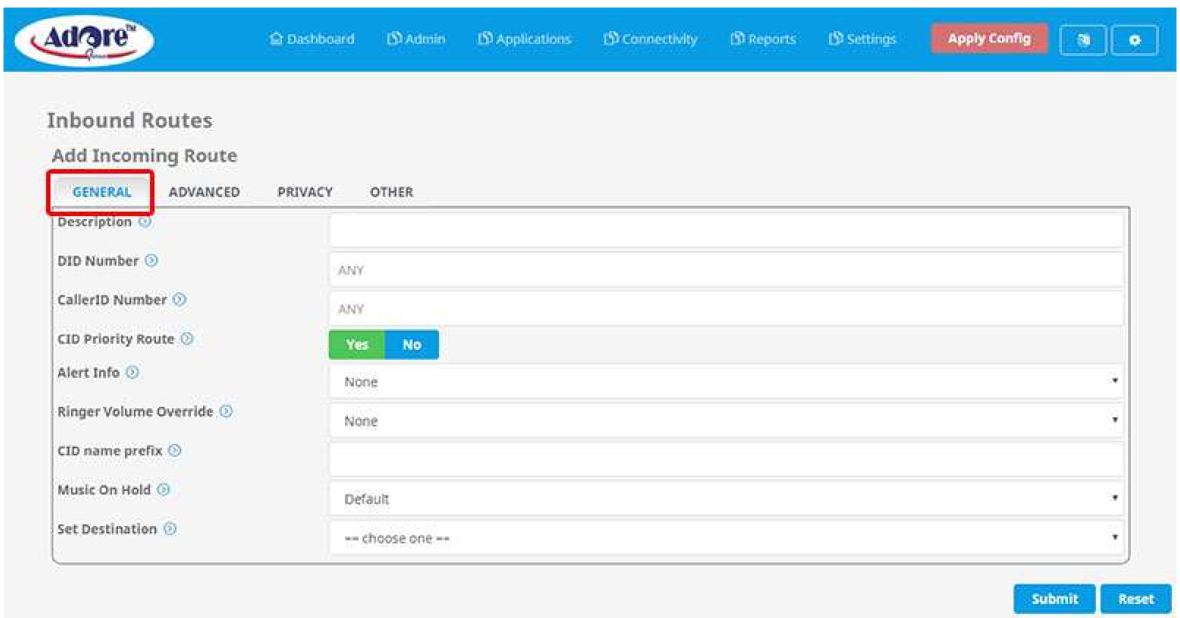


On click **Inbound Routes** following screen will appear. Click the **+ Add Inbound Route** button.



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.



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.

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:

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.

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.”

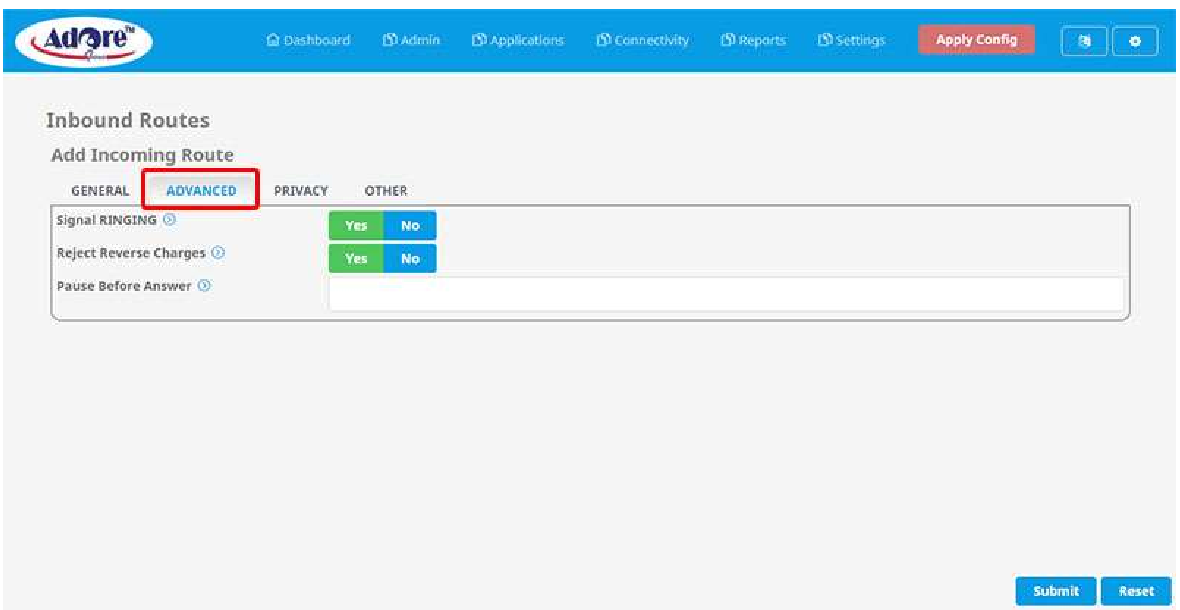
Music On Hold

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.

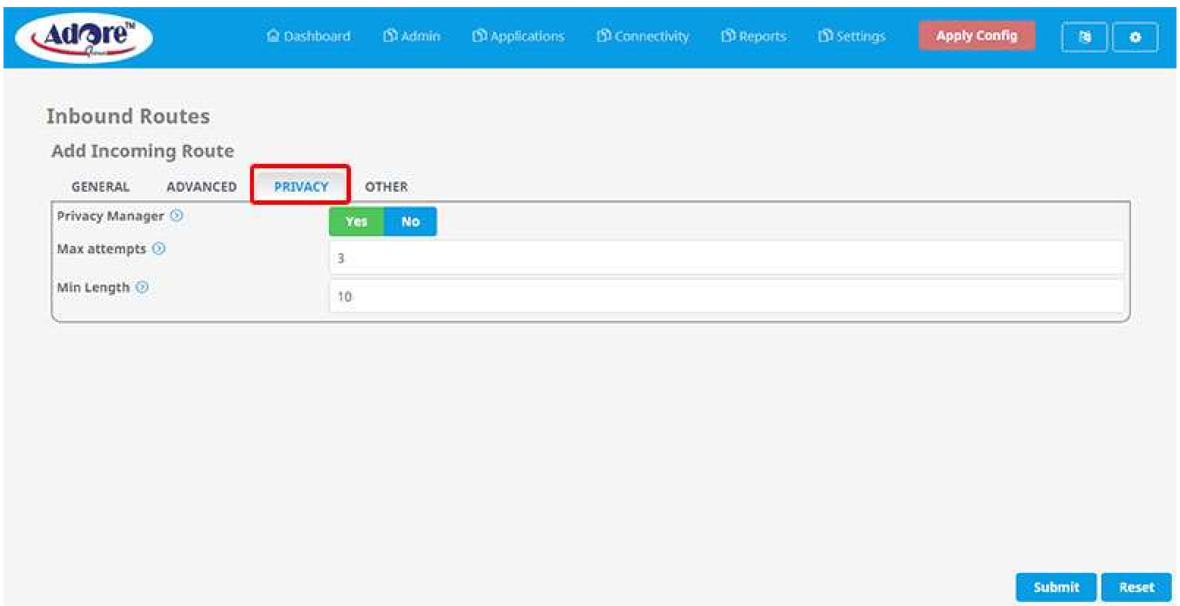
Reject Reverse Charges

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.

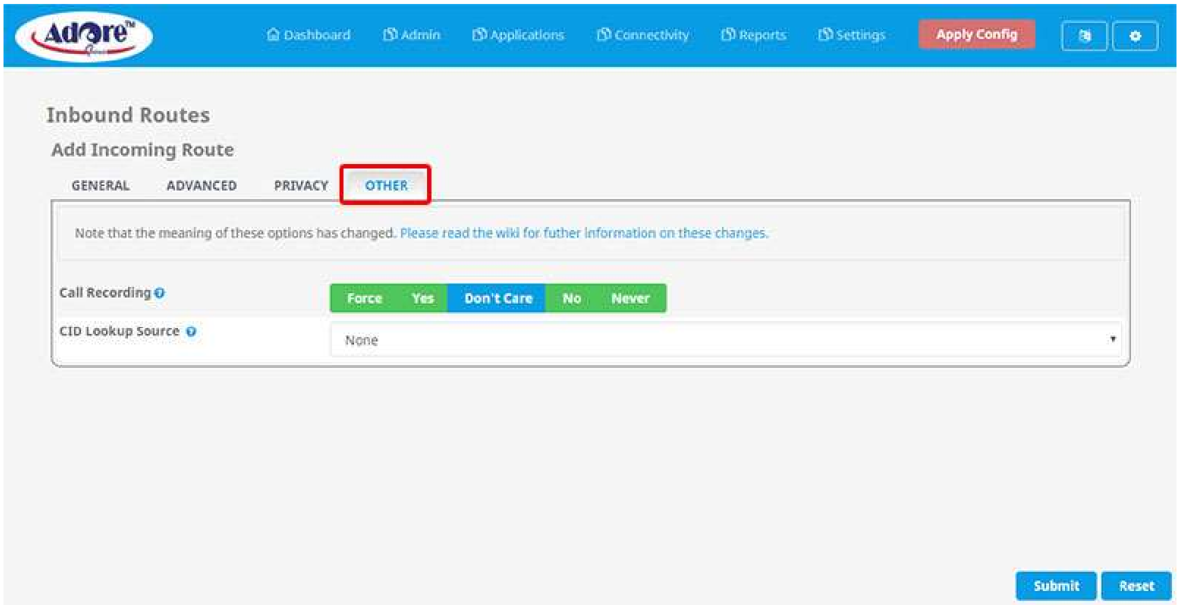
Max attempts

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.

Other



Call Recording

Force/Yes/Don't Care/No/Never: This setting controls or overrides the call recording behavior for calls using this route. See the Call Recording Walk Through for details on the modes.

Save

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

4.2. Outbound Routes

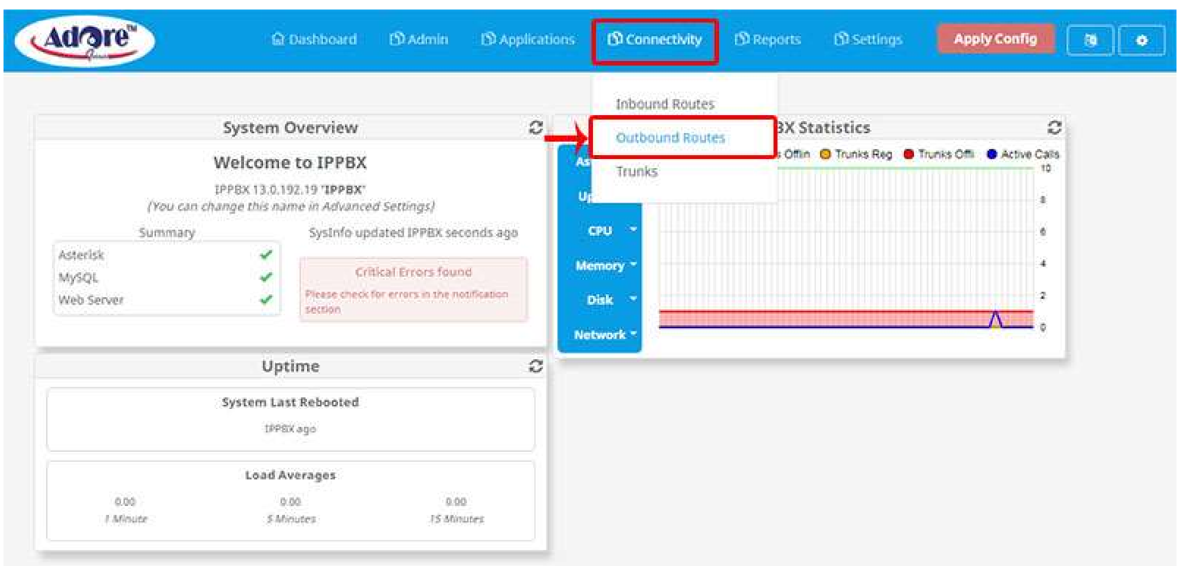
Outbound Routes

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.

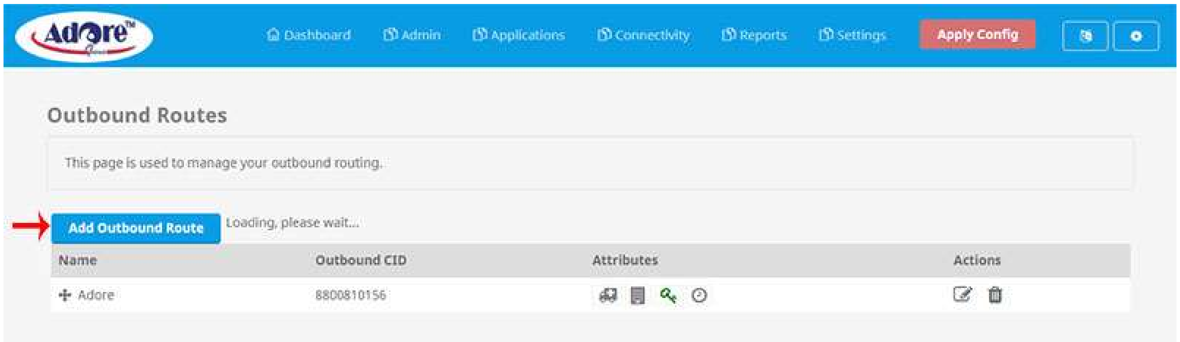
You will want to make sure you define routes for all types of calls. Not defining a route can leave your users frustrated when they need to make an important call.

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.

Go to **Connectivity** -> **Outbound Routes**



On click **Outbound Routes** following screen will appear.



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

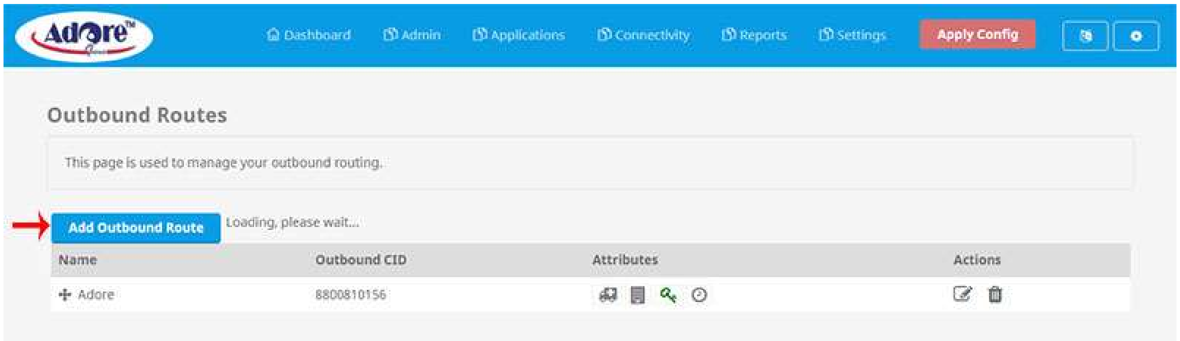
Actions

= View or Edit

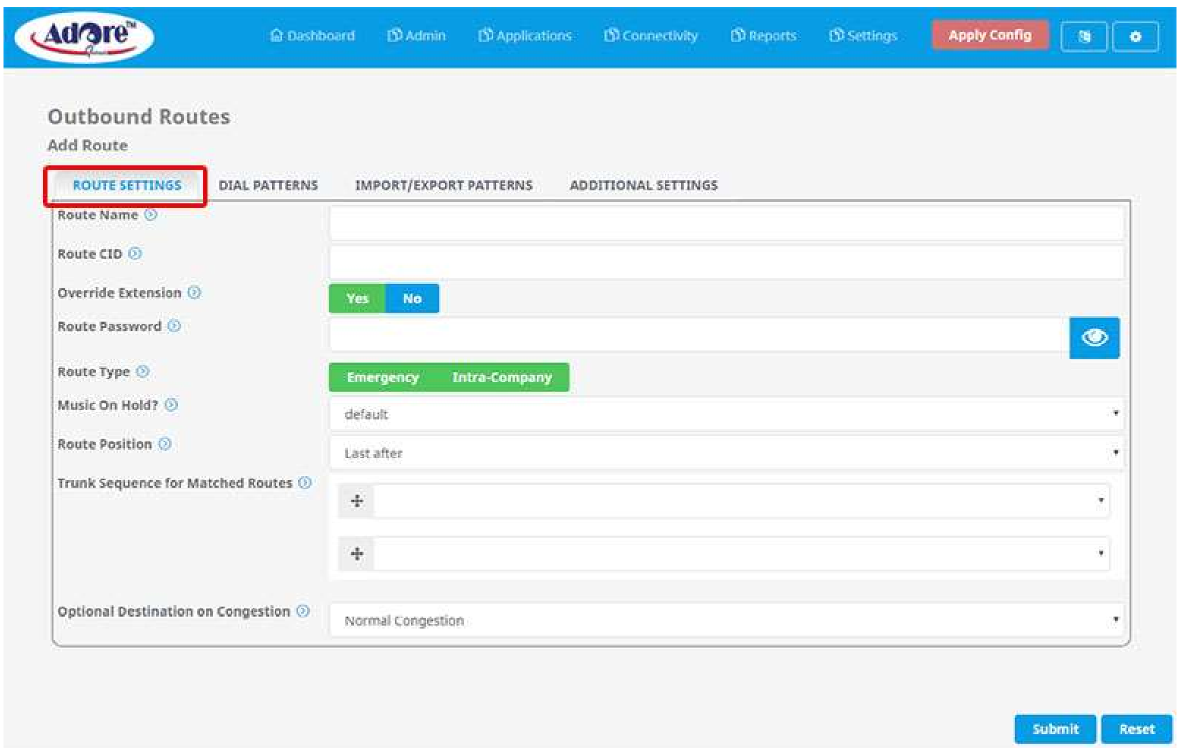
= Delete

Adding an Outbound Route

Click the +**Add Outbound Route** button.



Route Settings



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

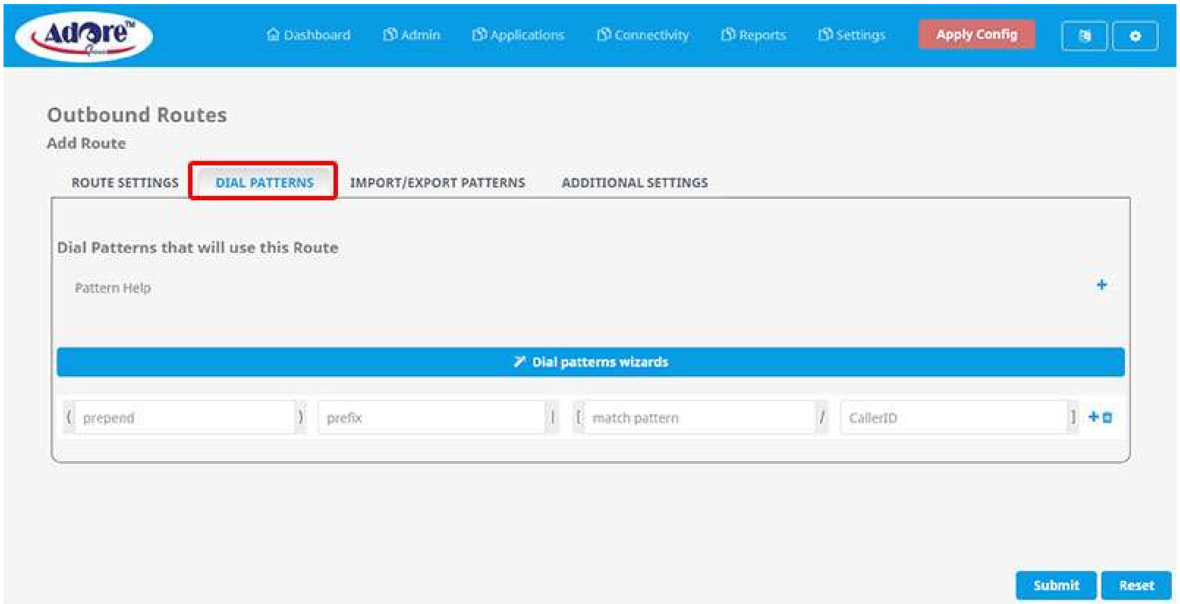
Controls the order of trunks that will be used when the above dial patterns are matched. For dial patterns that match long distance numbers, for example, you'd want to pick the cheapest routes for long distance (i.e., VoIP trunks first) followed by more expensive routes (POTS lines).

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



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:

Patterns	**Description**
X	Any whole number from 0-9
Z	Any whole number from 1-9

N	Any whole number form 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.

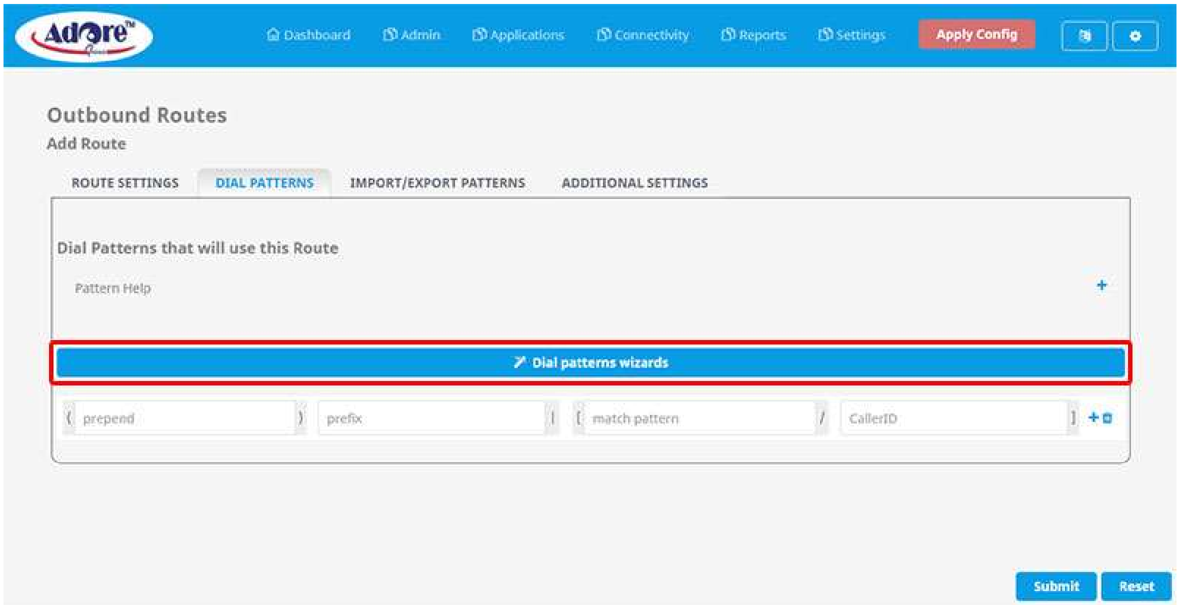
CallerID

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.

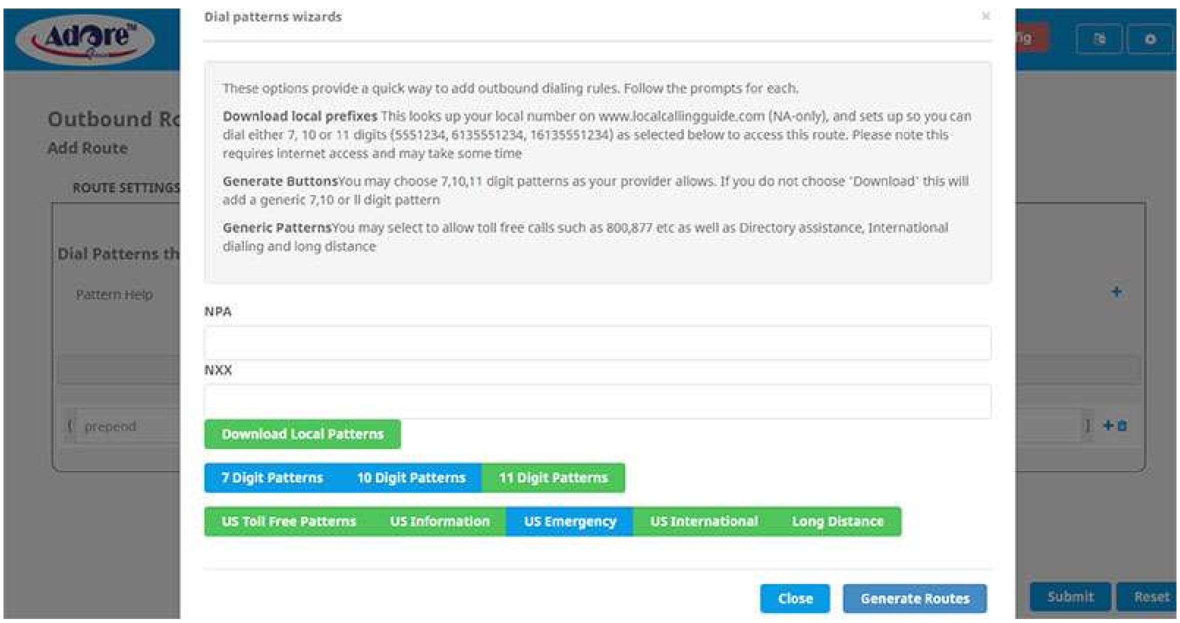
Dial Patterns Wizards

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 wizards** button.



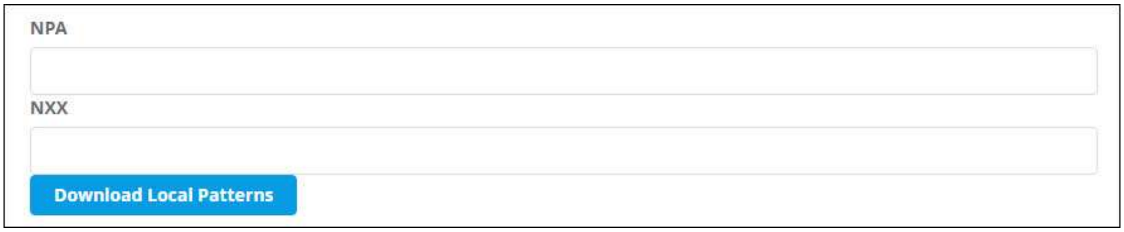
This displays a pop-up window where you can generate various dial patterns.



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). This feature is only available for North American numbers.



- 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.

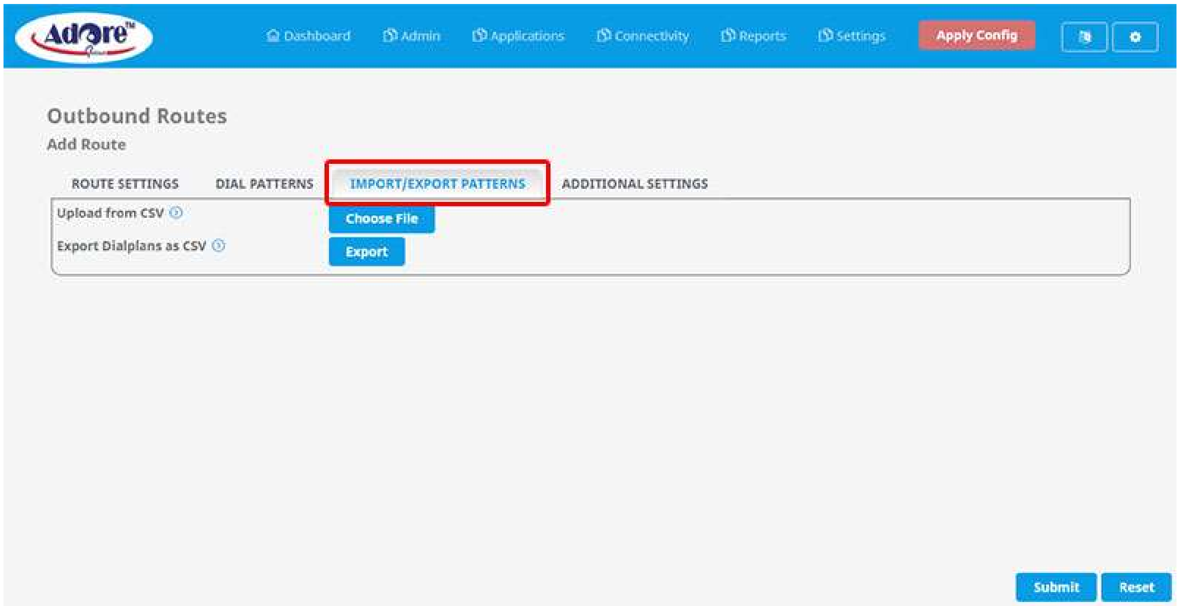
How to Generate Toll-Free, US Information, US Emergency, US International, and Long Distance Dial Patterns

-
    - 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

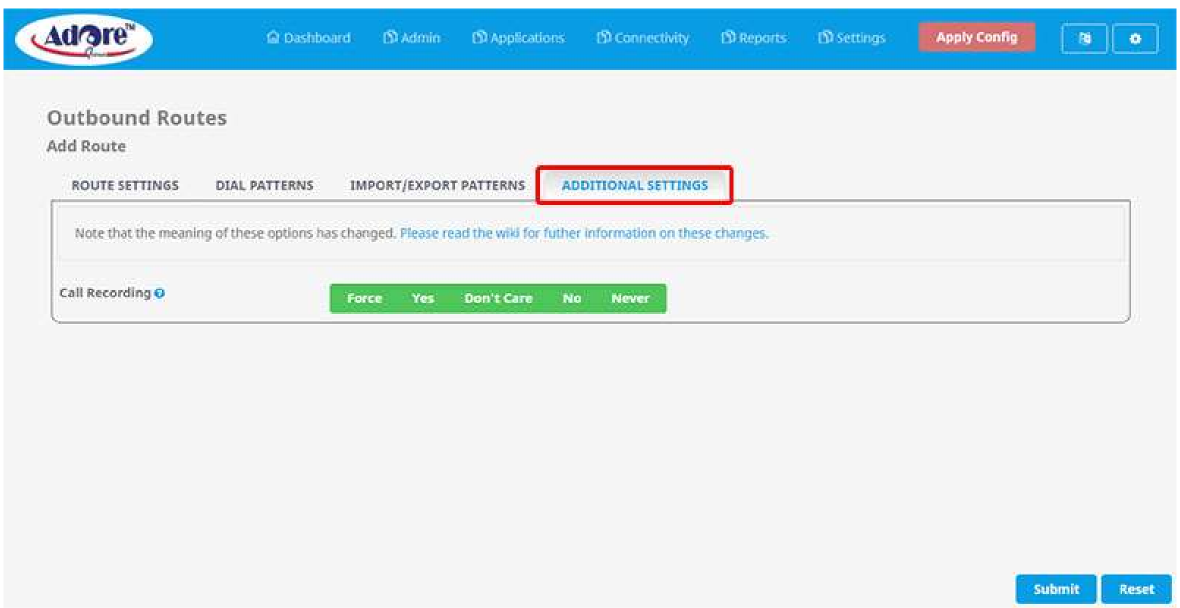
Click the **Export** button to download a list of patterns as a CSV file with headers listed as: **prepend**, **prefix**, **match pattern** and **callerid** in the first row

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.

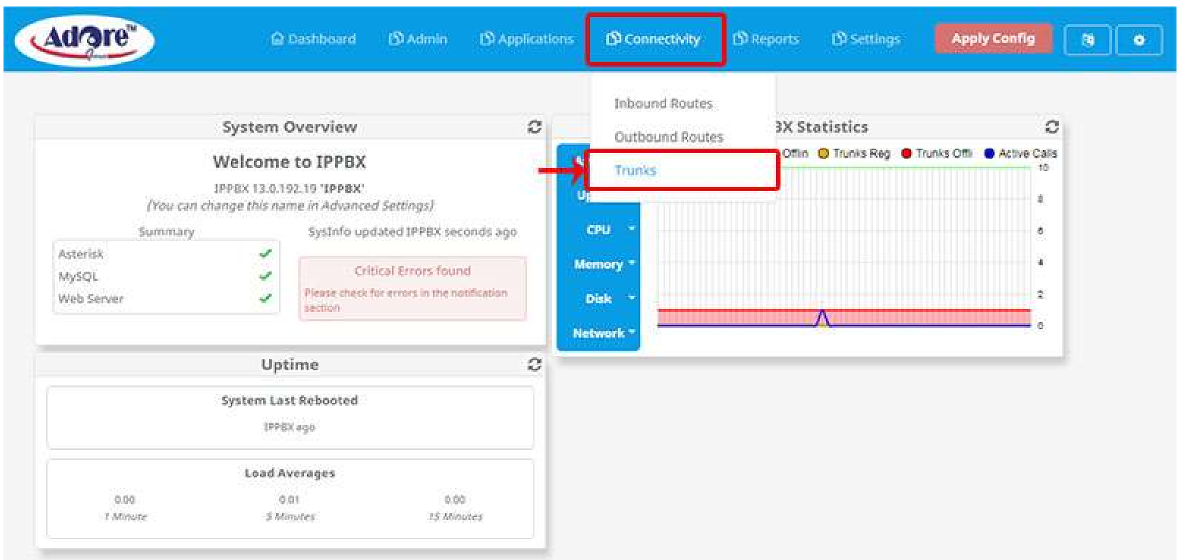
Click the **Submit** button, then click the **Apply Config** button to save the changes.

4.3. Trunks

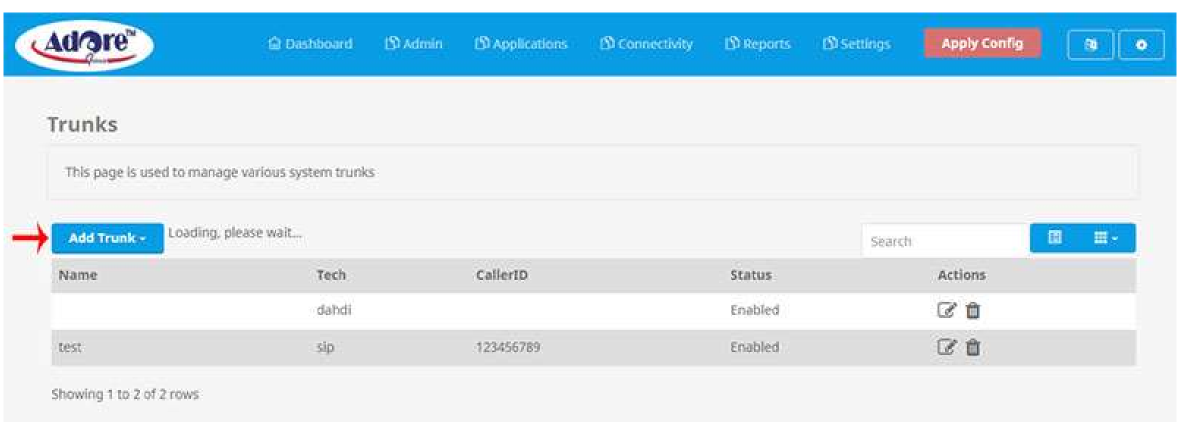
Trunks

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 and DAHDi (or Zap). 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.

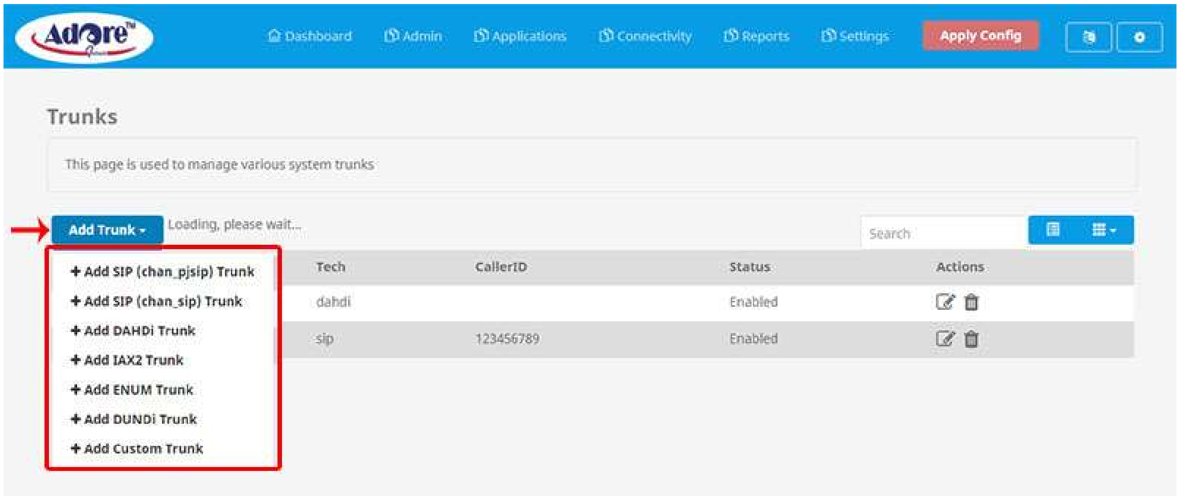
Go to **Connectivity** -> **Trunks**



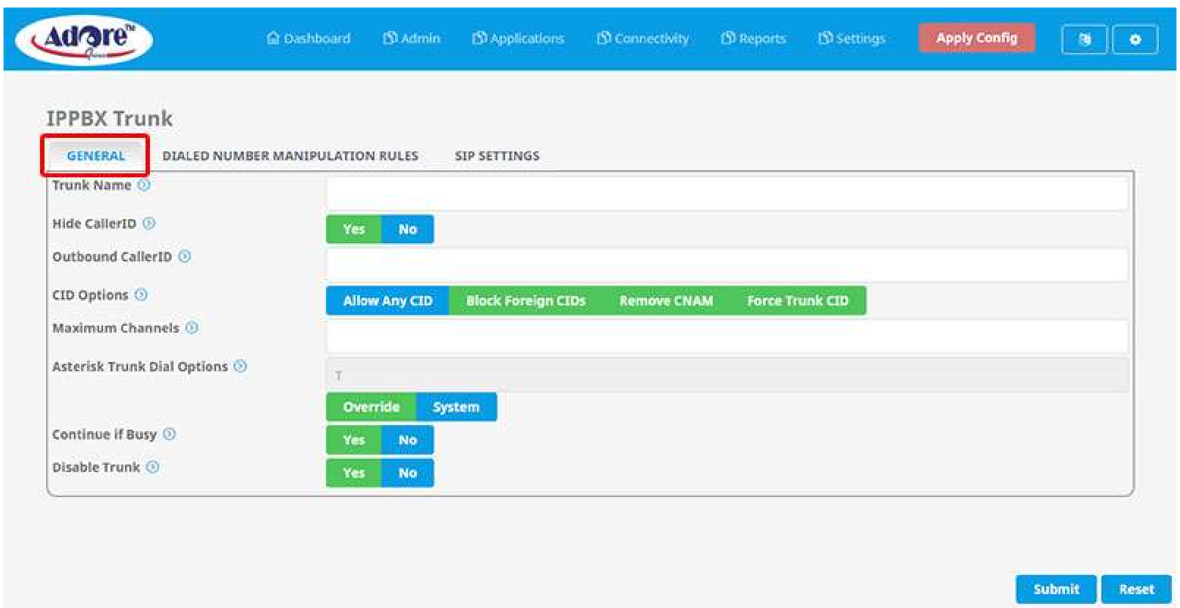
On click **Trunks** following screen will appear. Click the **+ Add Trunk** button.



You will want to click on the trunk type you wish to create.



General



Trunk Name

Set a descriptive name for the trunk.

Hide CallerID

Hide the outbound Caller ID, The same as adding hidden to Outbound CID

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.

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.

Asterisk Trunk Dial Options

Asterisk Dial command options to be used when calling out this trunk. To override the Advanced Settings default, check the box and then provide the required options for this trunk

Continue if Busy

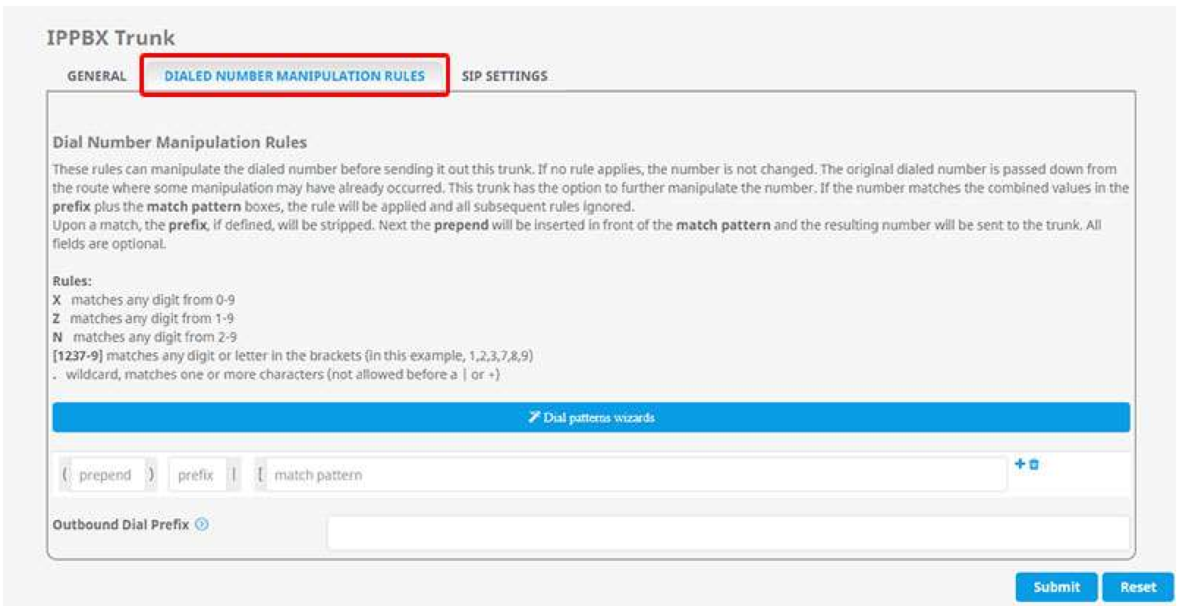
Normally the next trunk is only tried upon a trunk being 'Congested' in some form, or unavailable. Checking this box will force a failed call to always continue to the next configured trunk or destination even when the channel reports BUSY or INVALID NUMBER. This should normally be unchecked

Disable Trunk

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

Dialed Number 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.



5. Reports

Reports

For more information on a particular Reports Module, select the Module from the list below.

- **CDR Reports**
- **Call Event Logging**
- **LogFiles**
- **Print Extensions**

5.1. CDR Reports

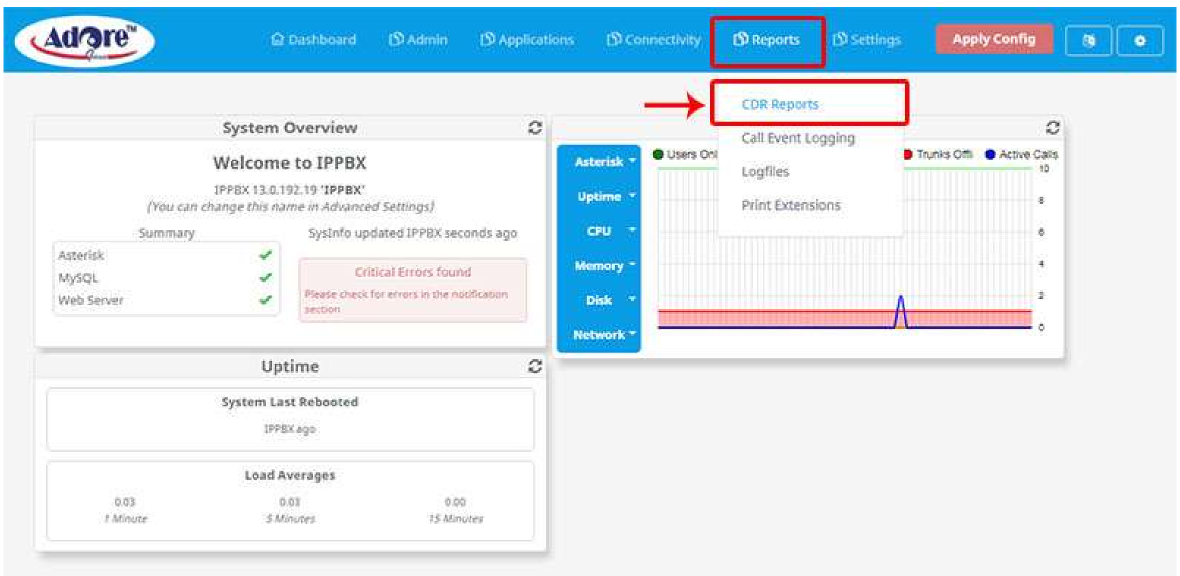
CDR Reports

The CDR Reports Module allows you to view a report showing the telephone calls made from and received to your system.

You can choose to view a complete history or calls, or to search by date, date range, number called, caller ID, etc.

The CDR Reports Module includes tool tips that help to explain what the options means.

Go to **Reports** -> **CDR Reports**

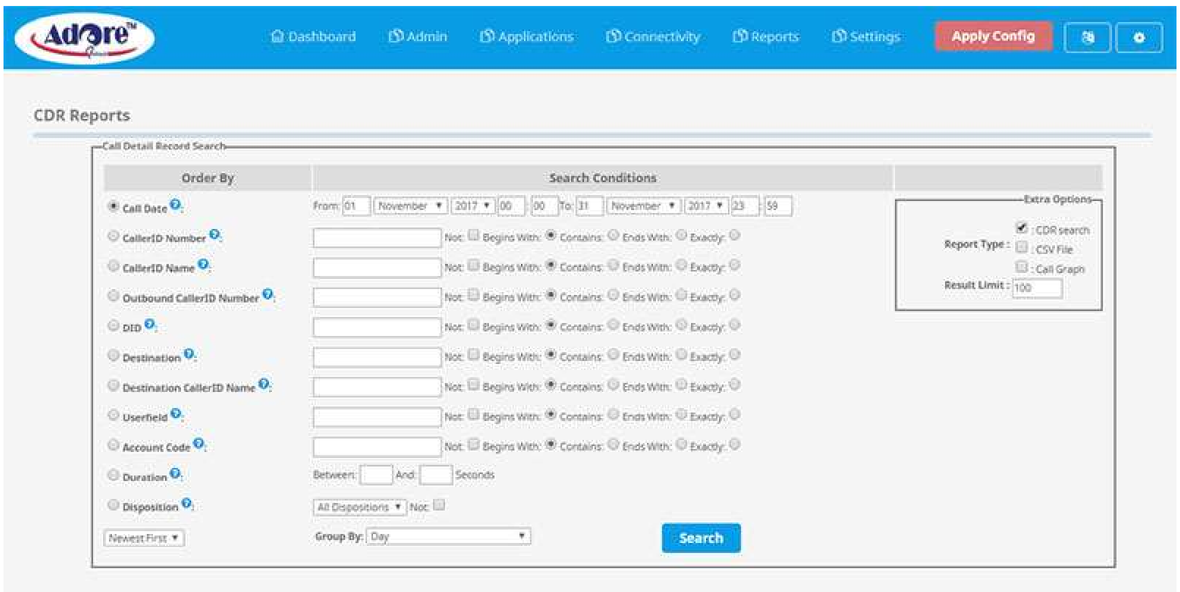


On click **CDR Reports** following screen will appear. Call Reports is designed to be the raw data of all call activity on your phone system. It can be a very challenging module to work with because it's not in a very user-friendly format and there is so much raw Call Detail Records (CDR). It is really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

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

- Call Date
- CallerID Number
- CallerID Name
- Outbound CallerID Number

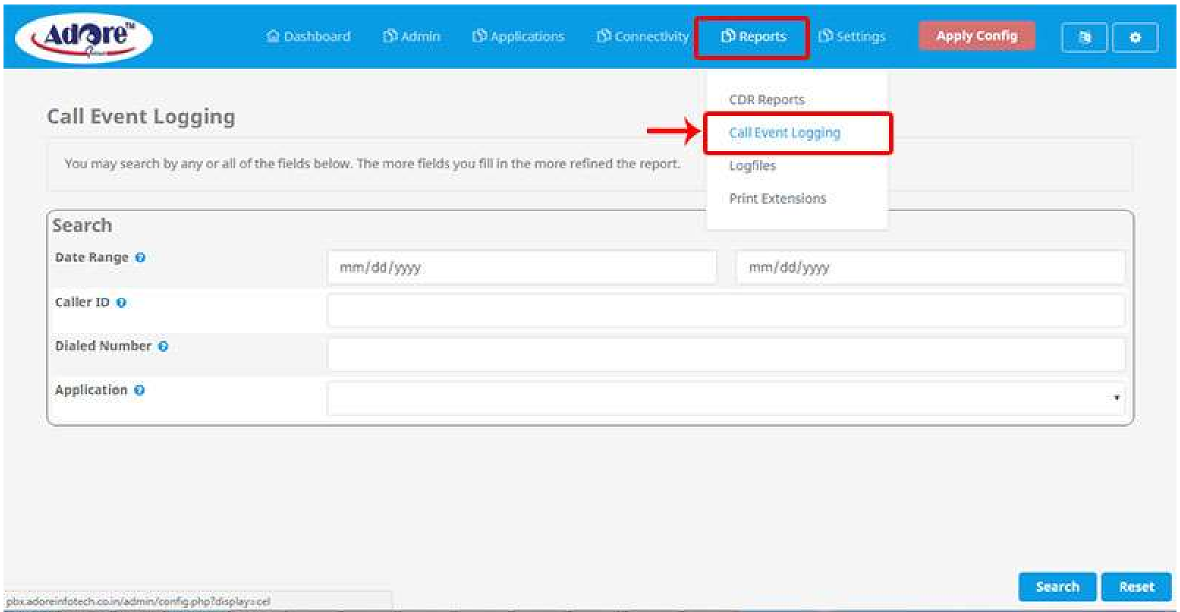
- DID
- Destination
- Destination CallerID Name
- Userfield
- Account Code
- Duration
- Disposition



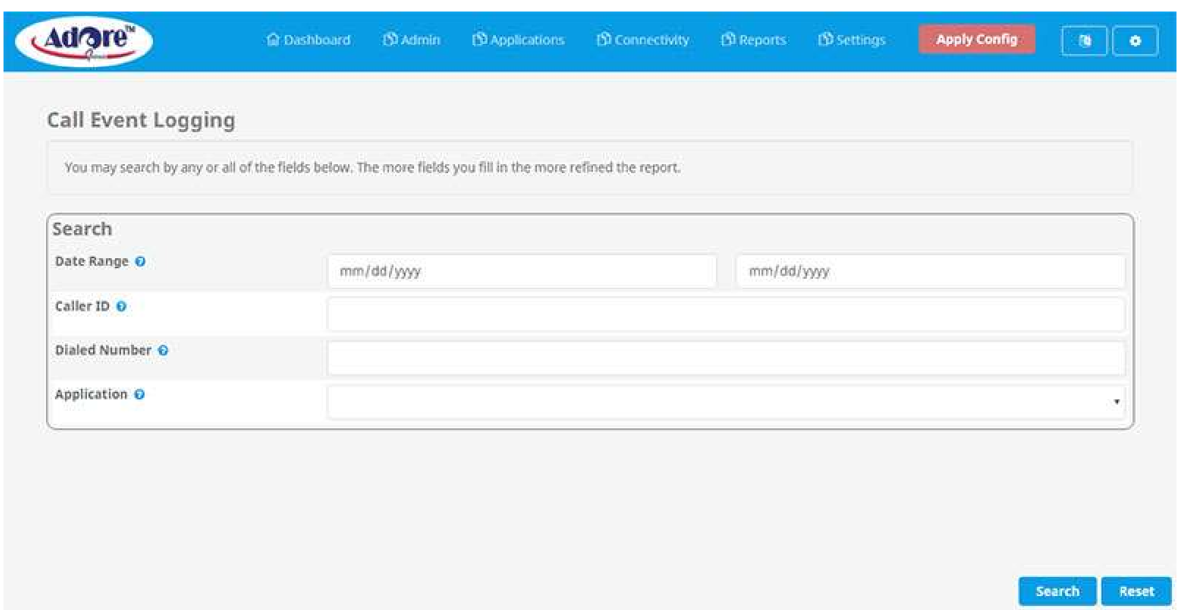
5.2. Call Event Logging

Call Event Logging

Go to **Report** -> **Call Event Logging**



On click **Call Event Logging** following screen will appear. Here you can find all Call Event Log Report by using search section.

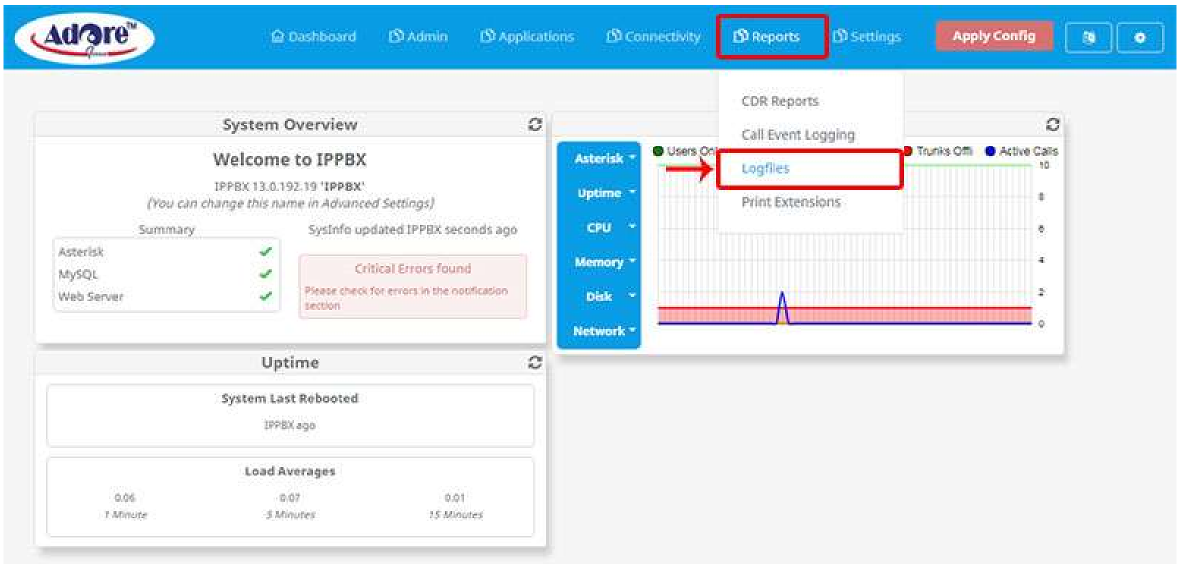


5.3. LogFiles

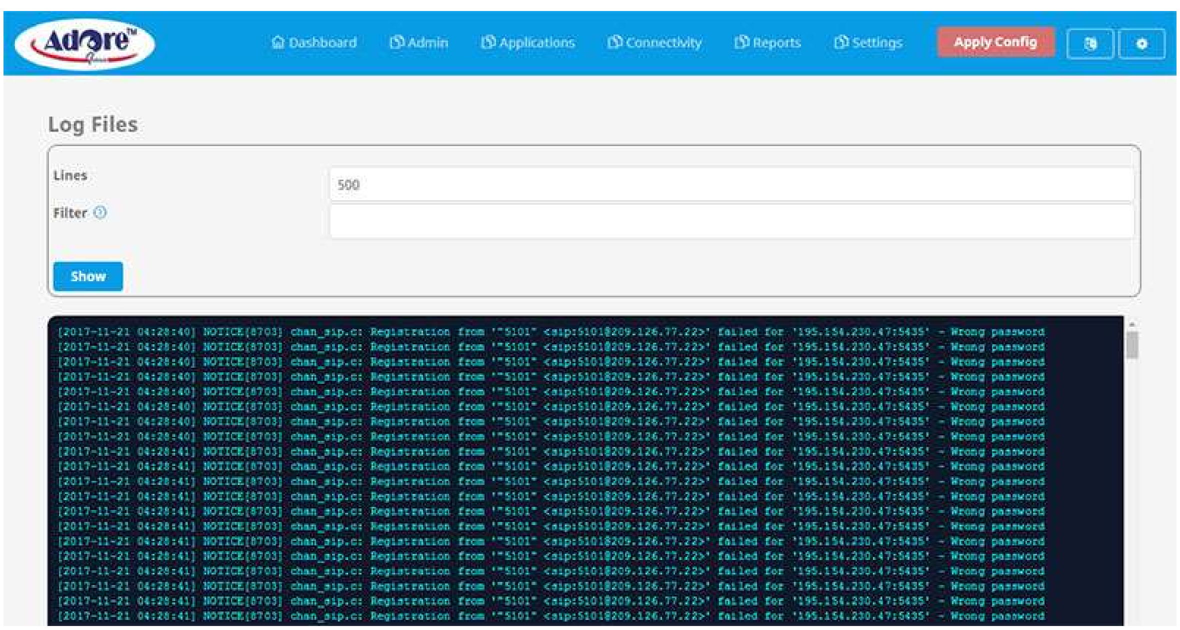
LogFiles

The Logfiles Module is an easy way to view portions of the Log. However, this Module is only useful when you want to view a very recent event in the Log.

Go to **Reports** -> **LogFiles**



On click **LogFiles** following screen will appear.Here you can see LogFiles of the PBX System.

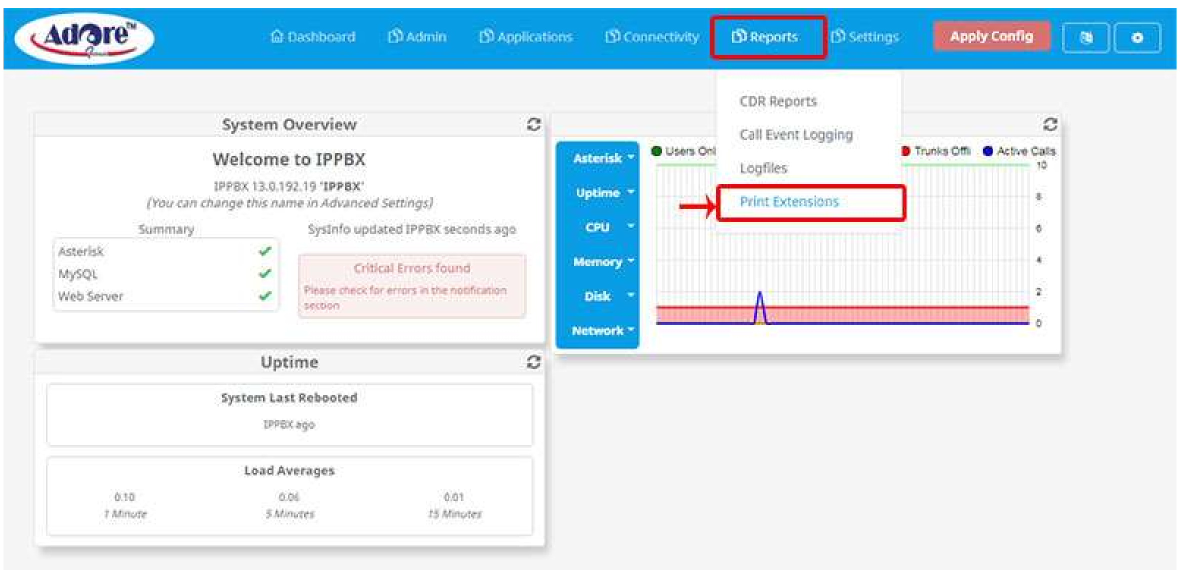


5.4. Print Extensions

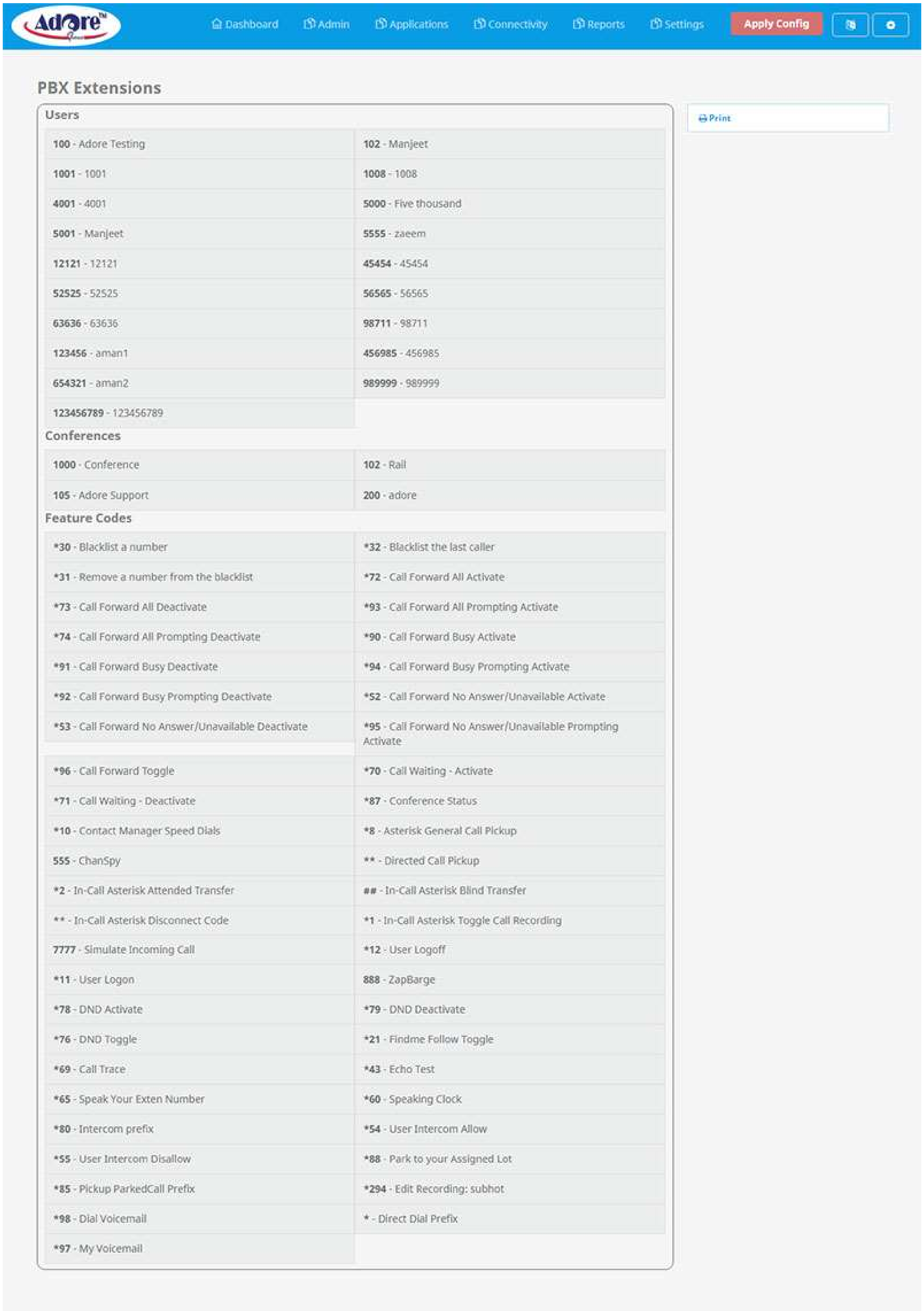
Print Extensions

The Print Extensions Module is a useful tool that allows you to print a list of all numbers that can be dialed from or to your system, and where callers who dial them will end up. The Module begins by showing you the Inbound Routes, which are numbers that outside callers (any caller coming into a trunk that uses context=from-trunk) can dial, and where those calls will be routed. The remaining information listed are phone numbers that can be dialled by internal callers (any caller that uses a context=from-internal, including your Extensions).

Go to **Reports** -> **Print Extensions**



On click **Print Extensions** following screen will appear.



PBX Extensions

Users

100 - Adore Testing	**102** - Manjeet
1001 - 1001	**1008** - 1008
4001 - 4001	**5000** - Five thousand
5001 - Manjeet	**5555** - zaeem
12121 - 12121	**45454** - 45454
52525 - 52525	**56565** - 56565
63636 - 63636	**98711** - 98711
123456 - aman1	**456985** - 456985
654321 - aman2	**989999** - 989999
123456789 - 123456789	

Conferences

1000 - Conference	**102** - Rail
105 - Adore Support	**200** - adore

Feature Codes

***30** - Blacklist a number	***32** - Blacklist the last caller
***31** - Remove a number from the blacklist	***72** - Call Forward All Activate
***73** - Call Forward All Deactivate	***93** - Call Forward All Prompting Activate
***74** - Call Forward All Prompting Deactivate	***90** - Call Forward Busy Activate
***91** - Call Forward Busy Deactivate	***94** - Call Forward Busy Prompting Activate
***92** - Call Forward Busy Prompting Deactivate	***52** - Call Forward No Answer/Unavailable Activate
***53** - Call Forward No Answer/Unavailable Deactivate	***95** - Call Forward No Answer/Unavailable Prompting Activate
***96** - Call Forward Toggle	***70** - Call Waiting - Activate
***71** - Call Waiting - Deactivate	***87** - Conference Status
***10** - Contact Manager Speed Dials	***8** - Asterisk General Call Pickup
555 - ChanSpy	****** - Directed Call Pickup
***2** - In-Call Asterisk Attended Transfer	**##** - In-Call Asterisk Blind Transfer
****** - In-Call Asterisk Disconnect Code	***1** - In-Call Asterisk Toggle Call Recording
7777 - Simulate Incoming Call	***12** - User Logoff
***11** - User Logon	**888** - ZapBarge
***78** - DND Activate	***79** - DND Deactivate
***76** - DND Toggle	***21** - Findme Follow Toggle
***69** - Call Trace	***43** - Echo Test
***65** - Speak Your Exten Number	***60** - Speaking Clock
***80** - Intercom prefix	***54** - User Intercom Allow
***55** - User Intercom Disallow	***88** - Park to your Assigned Lot
***85** - Pickup ParkedCall Prefix	***294** - Edit Recording: subhot
***98** - Dial Voicemail	***** - Direct Dial Prefix
***97** - My Voicemail	

6. Settings

Settings

For more setting on a particular Module, select the Module from the list below.

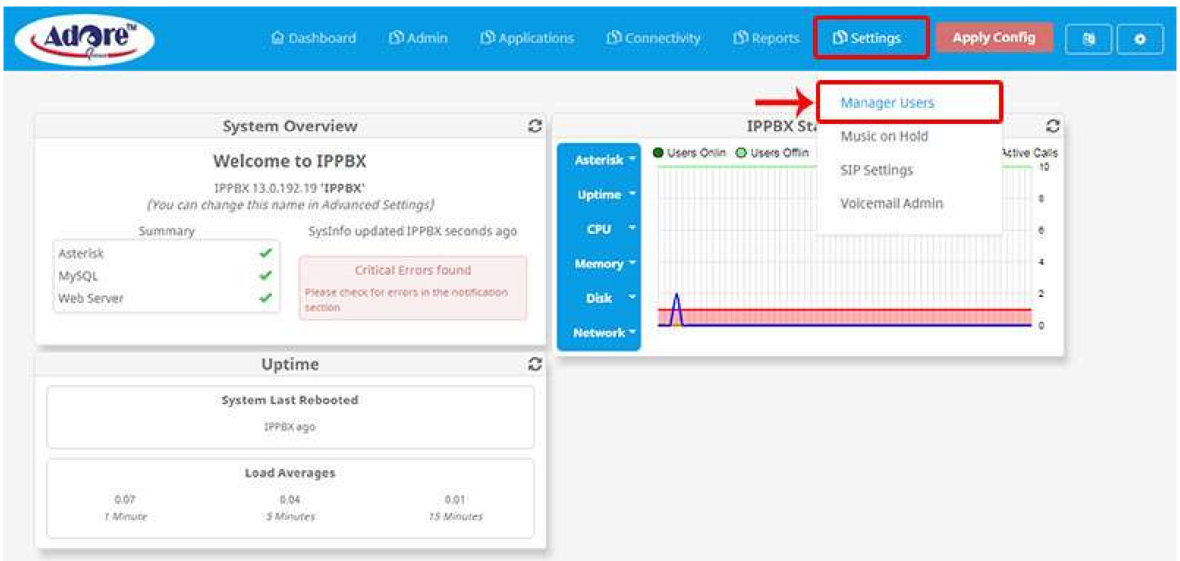
- **Manage Users**
- **Music On Hold**
- **SIP Settings**
- **Voicemail Admin**

6.1. Manage Users

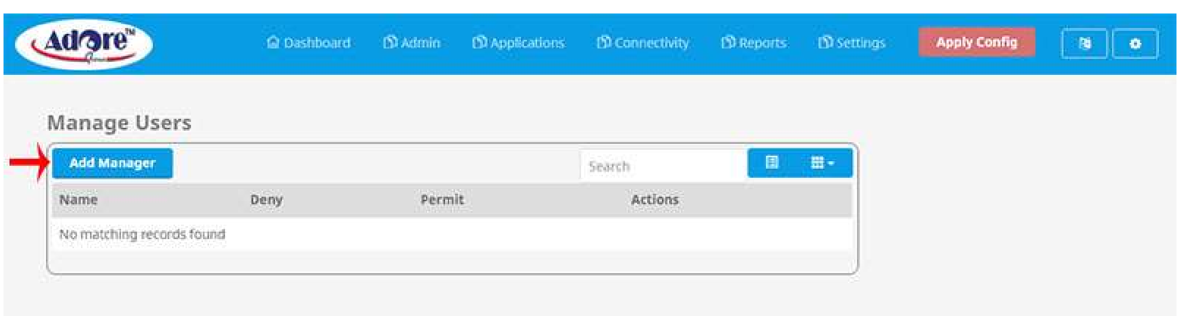
Manage Users

This Module is the Application Program Interface for/to the User Manager and allows for external systems to connect via TCP/IP to issue commands and readevents. Common examples of usage include Dialers, CRM, Management Console and so on.

Go to **Settings** -> **Manage Users**

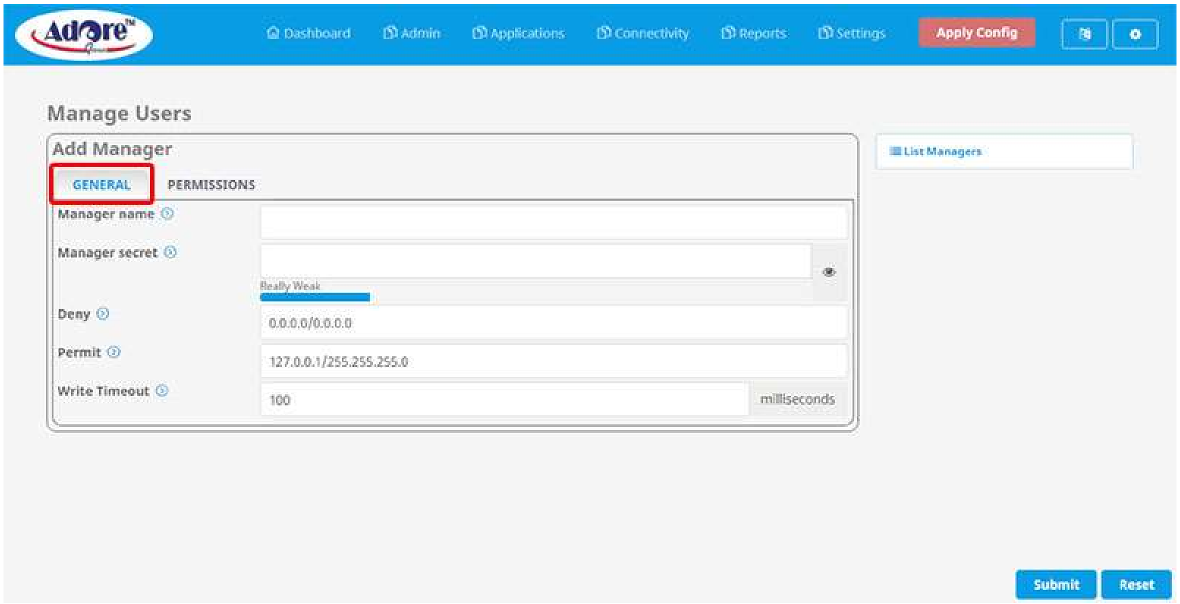


On click **Manage Users** following screen will appear. Here you can add Manager by clicking on **Add Manager** button.



General

On click **Add Manager** following screen will appear.



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

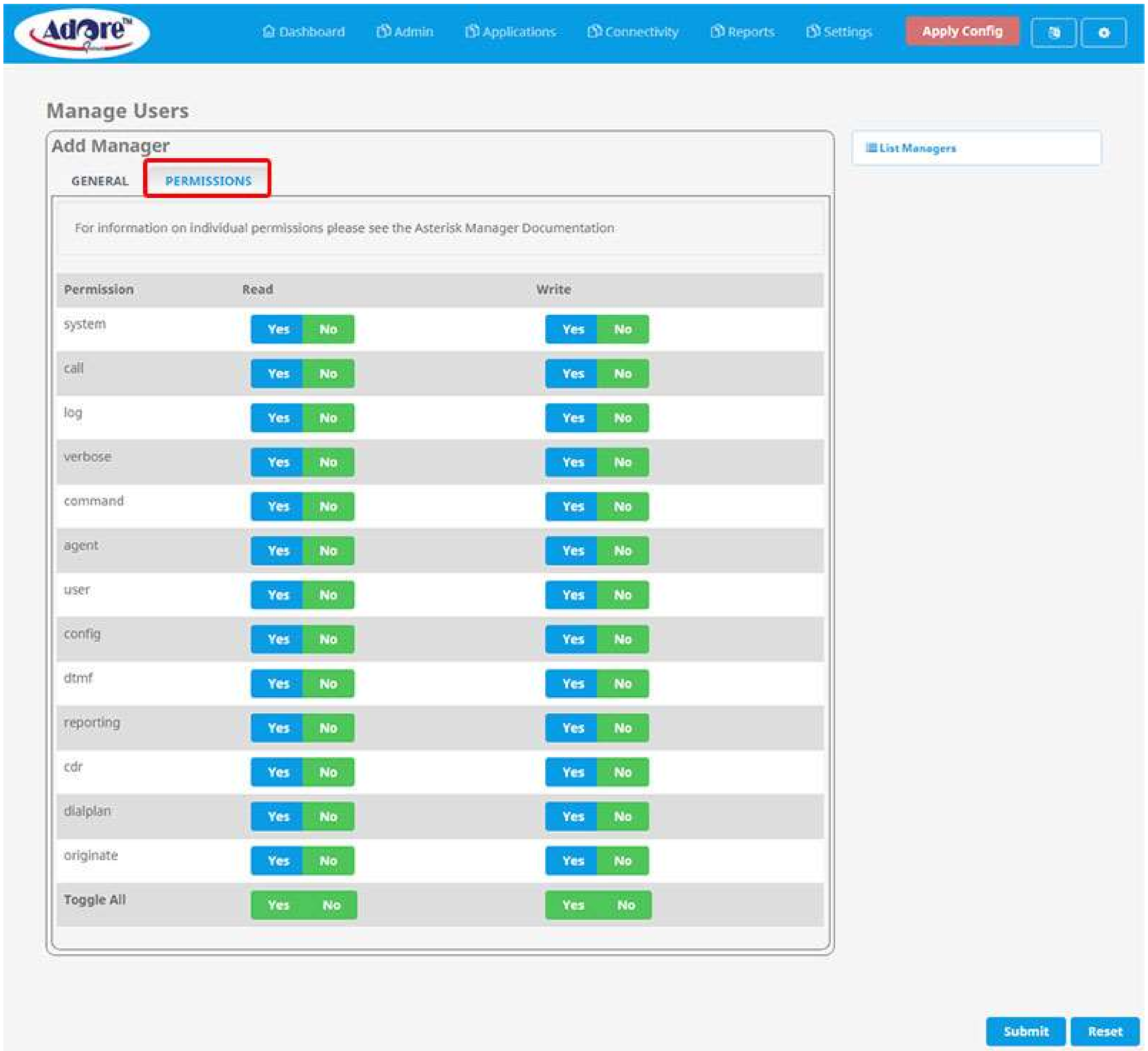
Here you define an IP Address/Subnet Mask Permit statement. You may define more than one network or device as with the Deny statement.

Write Timeout

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

Permission

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

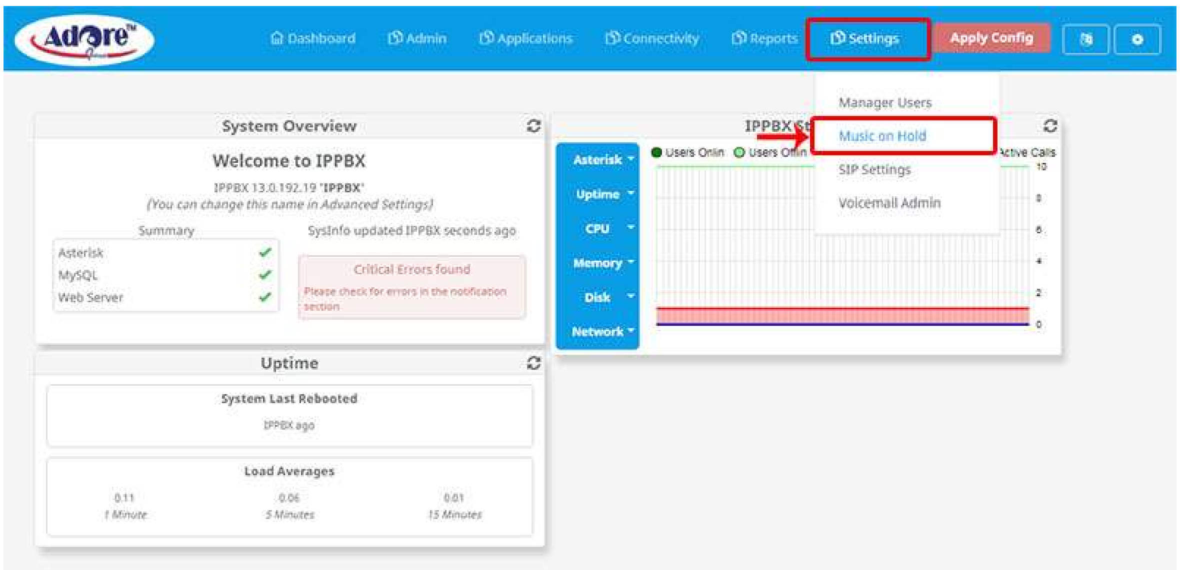


6.2. Music on Hold

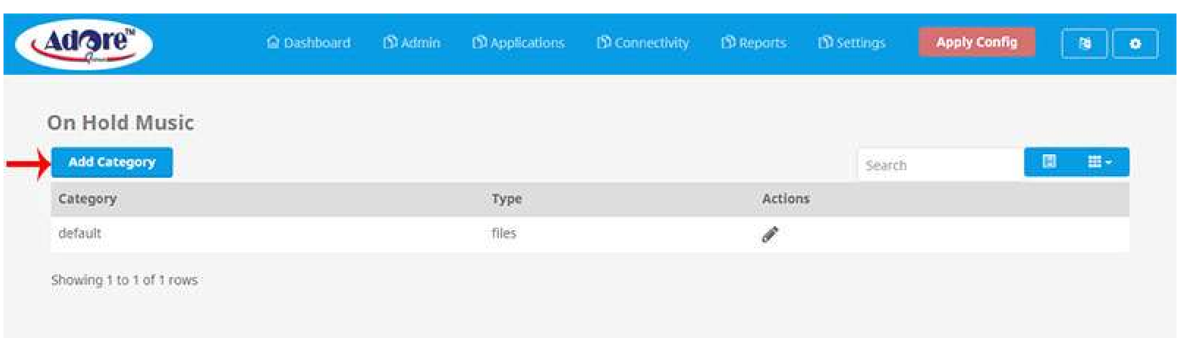
Music on Hold

The Music on Hold Module is used to upload audio files to your PBX from your web browser and to divide the files into various Music on Hold Categories. You can select which category to use for particular calls in the Inbound Routes, Outbound Routes, Ring Groups, and Queues Modules.

Go to **Settings** -> **Music on Hold**



On click **Music on Hold** following windows will appear. Click on **Add Category** button. For edit the **default** category please click on Edit icon.



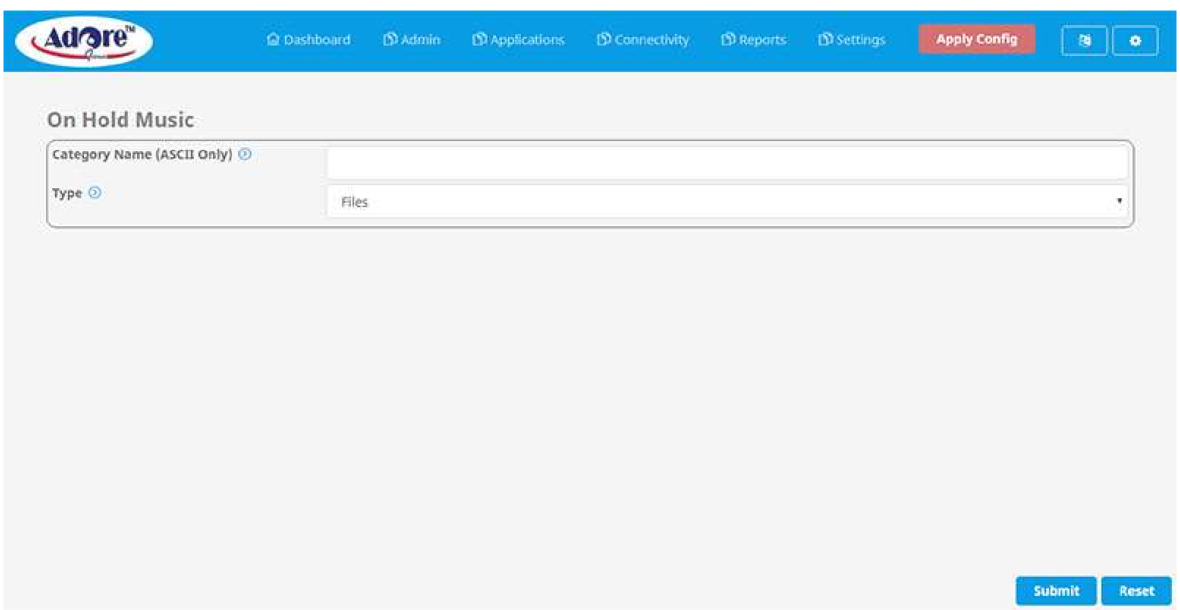
Music on Hold module is intended to reassure callers that they are still connected to their calls. The PBX comes with 11 built in songs that are the default hold music. You can easily add your own music or sound files to the system by uploading them in .wav or .mp3 format, or stream a live feed. Adding custom music on hold is a great way to bring personality to your phone system. The PBX allows two styles of music on hold – static files and streaming. Static files are audio files (WAV or MP3 files) that are uploaded to the server and played back when a caller is placed on hold. Streaming audio is used to connect to a live audio feed from a particular source. This could be an Internet stream, or a stream from a sound card or other audio device. MoH categories can be applied to inbound routes as well as to queues, ring groups, outbound routes and conferences. Categories assigned at the call level will override the MOH category for that target only. Once a call leaves that target, it will fall back into the category specified in the inbound route it was matched against.

Adding Music Categories

The PBX has one category, which is “default.” You can create additional categories by clicking on the “Add Music Category” in the top right hand side of the screen.

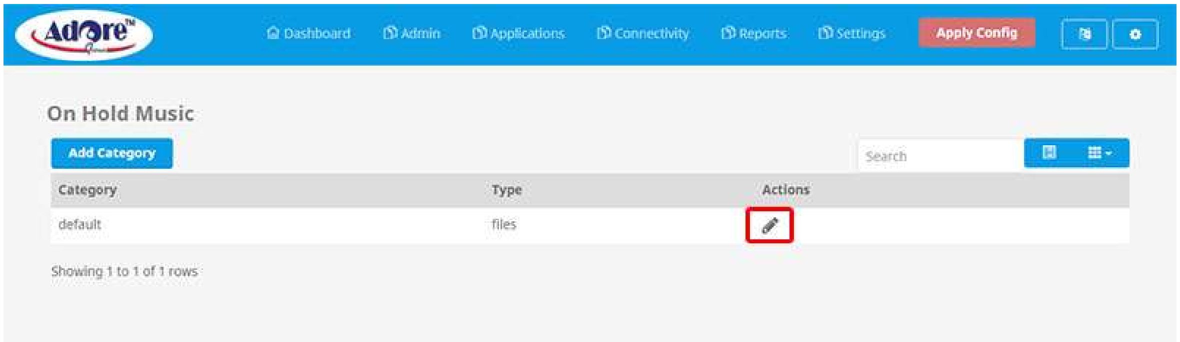
Category Name

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

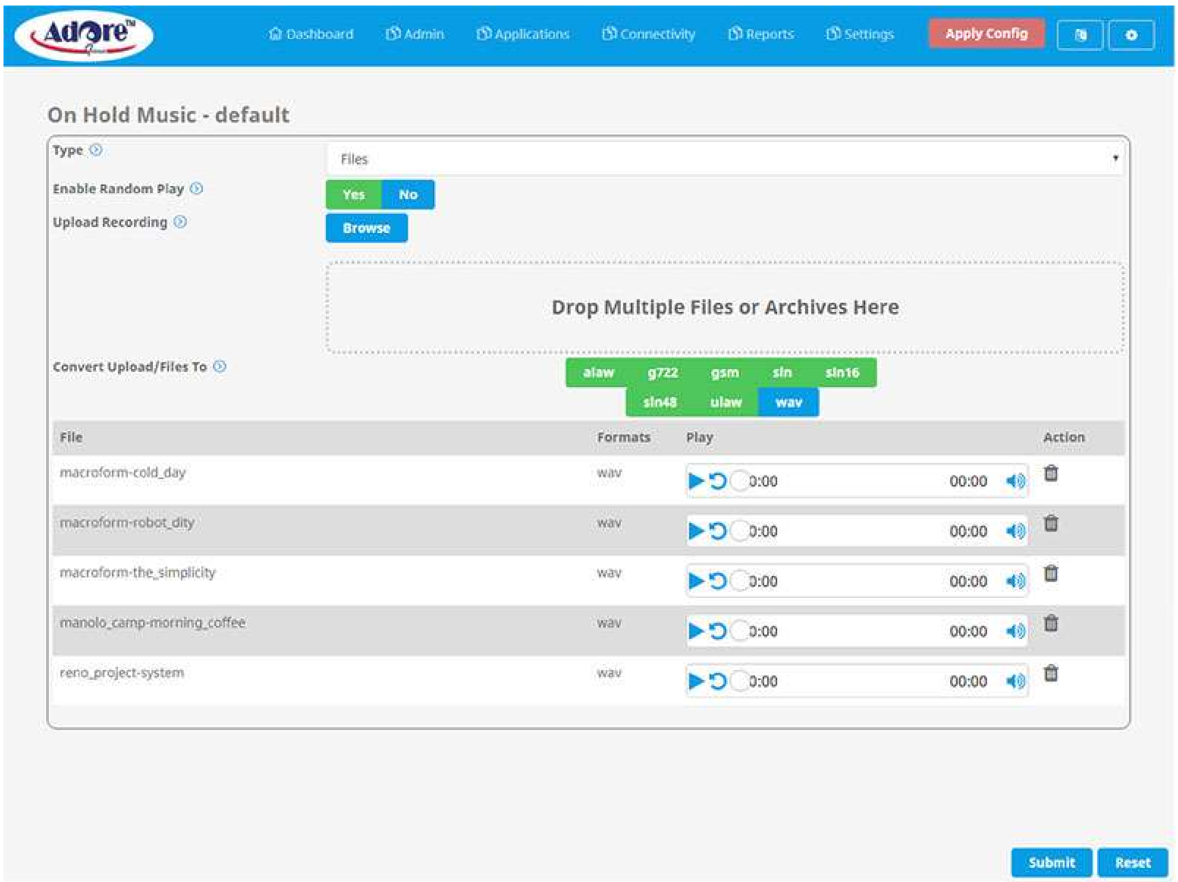


Edit the Default Category

For editing the default category if you want, click on Pencil Edit icon.



Edit the section as per your wish and click submit button.

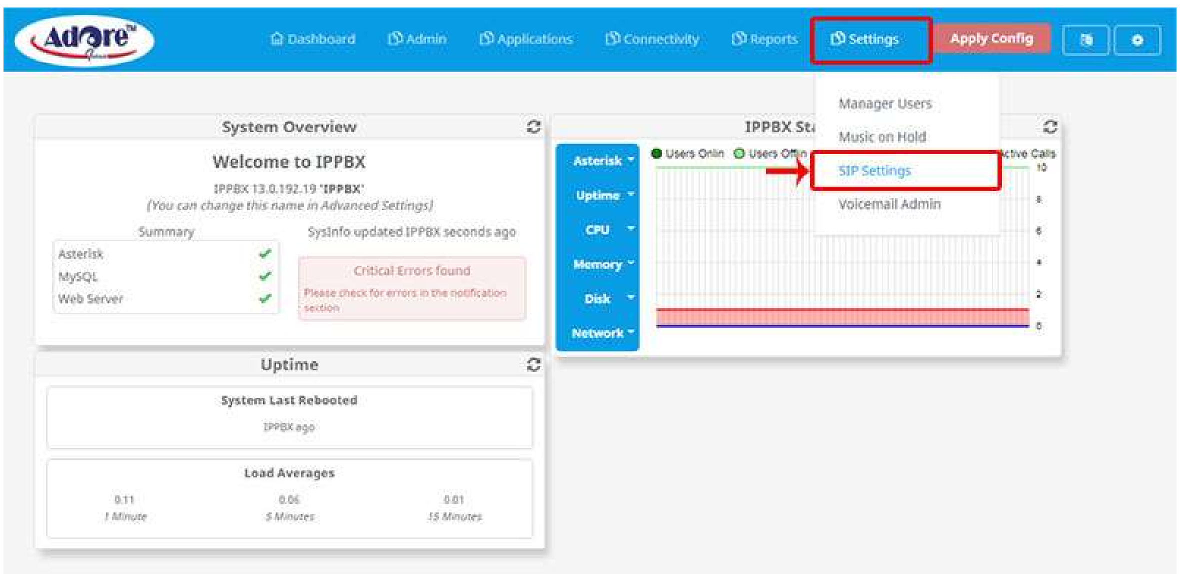


6.3. SIP Settings

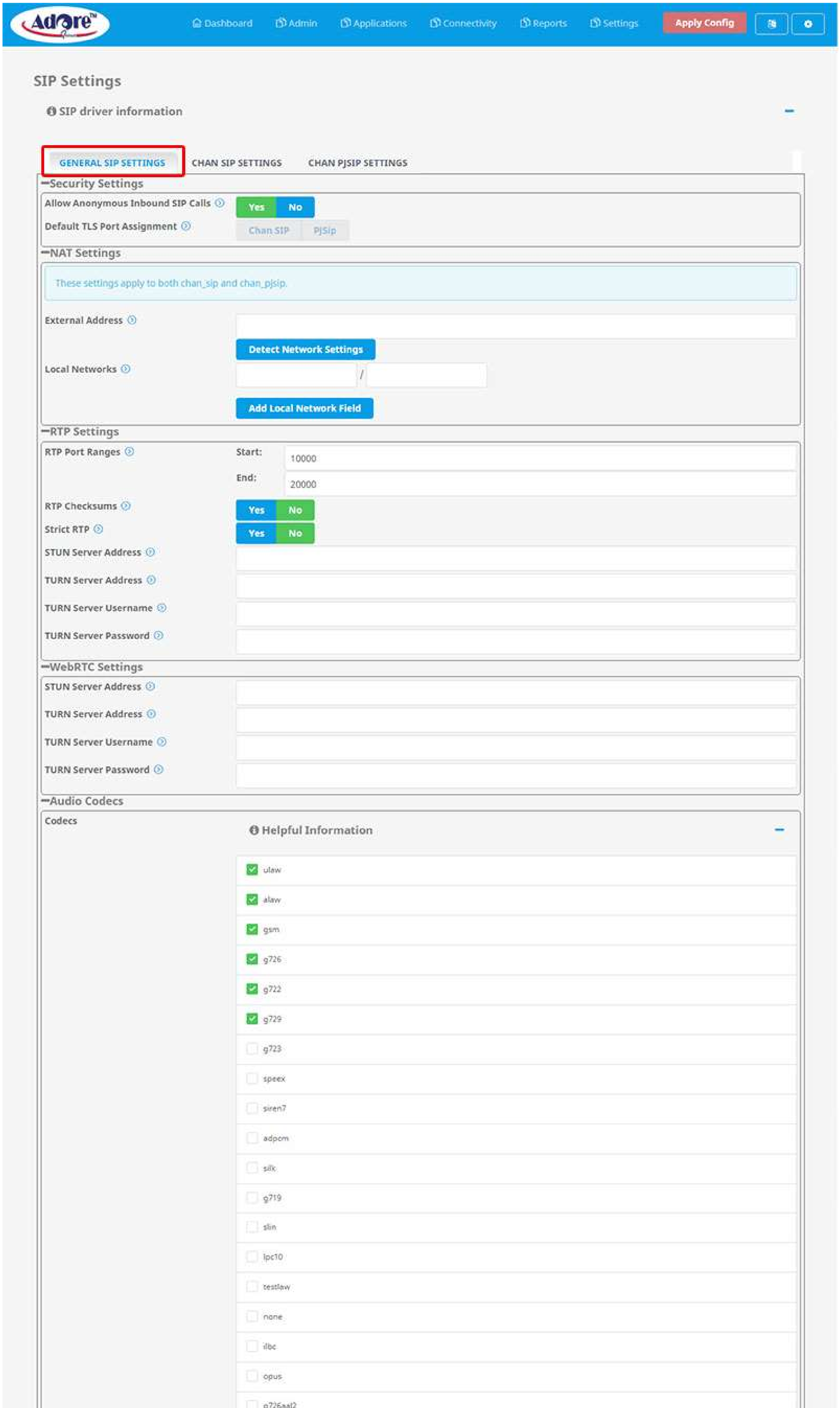
SIP Settings

The SIP Settings Module is used to configure the default settings used for SIP calls. Since most VOIP calls are sent using SIP, these settings can be very important to the operation of your PBX. Because this module sets the default settings, most of these settings can be overriden for a particular extension in the Extensions Module or for a particular trunk in the Trunks Module.

Go to **Settings** -> **SIP Settings**



On click **SIP Setting** following screen will appear.


Dashboard
Admin
Applications
Connectivity
Reports
Settings
Apply Config
SIP Settings
SIP driver information
GENERAL SIP SETTINGS
CHAN SIP SETTINGS
CHAN PJSIP SETTINGS
Security Settings
Allow Anonymous Inbound SIP Calls
Yes
No
Default TLS Port Assignment
Chan SIP
PJSip
NAT Settings
These settings apply to both chan_sip and chan_pjsip.
External Address
Detect Network Settings
Local Networks
Add Local Network Field
RTP Settings
RTP Port Ranges
Start:
10000
End:
20000
RTP Checksums
Strict RTP
STUN Server Address
TURN Server Address
TURN Server Username
TURN Server Password
WebRTC Settings
STUN Server Address
TURN Server Address
TURN Server Username
TURN Server Password
Audio Codecs
Codecs
Helpful Information
ulaw
alaw
gsm
g726
g722
g729
g723
speex
siren7
adpcm
silk
g719
slin
lpc10
testlaw
none
ilbc
opus

General SIP Settings

Allow Anonymous inbound SIP Calls

Allowing Inbound Anonymous SIP calls means that you will allow any call coming in from an unknown IP source to be directed to the 'from-pstn' side of your dialplan. This is where inbound calls come in. Although FreePBX severely restricts access to the internal dialplan, allowing Anonymous SIP calls does introduce additional security risks. If you allow SIP URI dialling to your PBX or use services like ENUM, you will be required to set this to Yes for Inbound traffic to work. This is NOT an Asterisk sip.conf setting, it is used in the dialplan in conjunction with the Default Context. If that context is changed to something custom, this setting may be rendered useless as well as if 'Allow SIP Guests' is set to no.

NAT Setting

This setting is used to indicate whether the other systems you connect with are behind a router that provides Network Address Translation ("NAT"). If is it set to "Yes," Asterisk will ignore the from address specified by the remote system and instead send response packets to the address that the packets actually came from. In most cases, it is safe to set this to "Yes."

This setting is overriden on a per extension basis using the NAT field (which defaults to "No" for extensions) and can be overriden on a per trunk basis by including "nat=" in the Trunks' PEER details. Contrary to popular belief, this setting is NOT used to indicate whether your system is behind a NAT.

IP Configuration: This field is used to tell Asterisk whether your PBX has public IP or is behind a NAT router.

If your system has its own public IP address, select "Public IP."

If your system is behind a router that provides "NAT," and your router has a static (non-changing) IP address assigned by your ISP, then select "Static IP." Then fill-in the "External IP" field with your Static IP address and the Local Networks field with the IP/Subnet of all of your internal networks.

Local Networks

Local network settings in the form of “ip/mask” such as, “192.168.1.0/255.255.255.0.” For networks with more than one LAN subnet, such as VPN network, us the “Add Local Network” button to add more fields. Blank fields will be removed upon submitting.

RTP Settings

RTP Ranges

The start and end ports for UDP RTP traffic. Default 10000-20000. You should have at least 4 ports per potential call.

RTP Checksums

Weather or not to enable UDP checksums for RDP traffic

Strict RTP

This will drop RTP packets that do not come from the source of the RTP stream. It is unusual to turn this off.

Codecs: Most SIP calls in North America are sent using the ULAW Codec. Some parts of the world use ALAW. If you want to enable "High Definition" audio, you'll also want to enable G722. If you want a low bandwidth Codec, you can enable GSM. Again, bear in mind that these are the defaults. Because most Trunk providers will override these settings in their PEER details with a disallow=all and then an allow=(whatever codecs they allow), the Codecs you specify here will generally only affect internal calls. The defaults are usually fine.

Save

Click Submit to save

6.4. Voicemail Admin

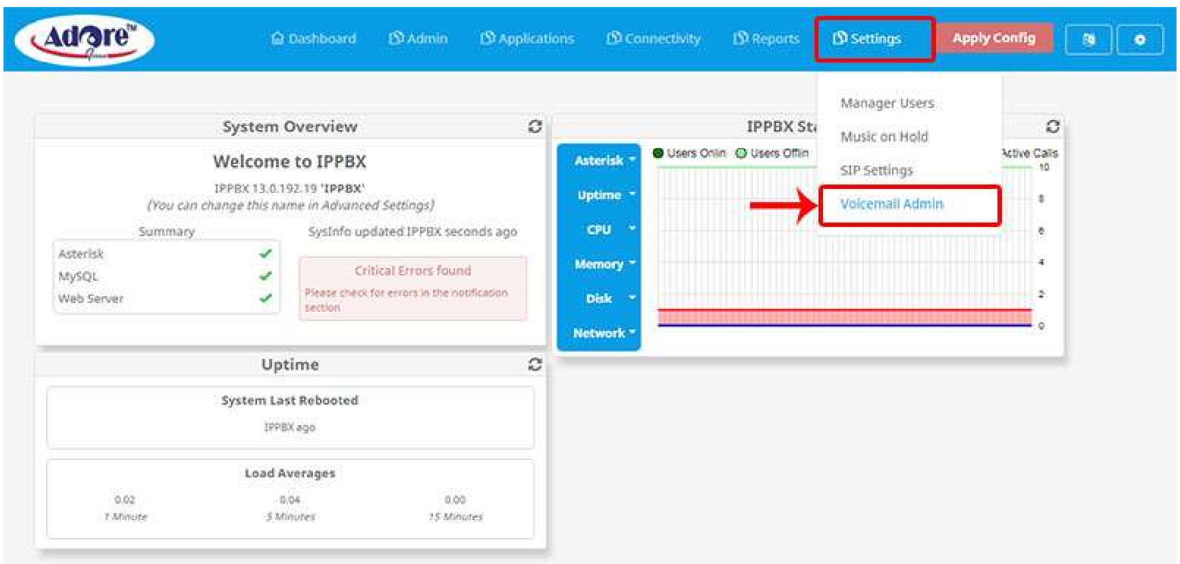
Voicemail Admin

The "Voicemail Admin " 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.

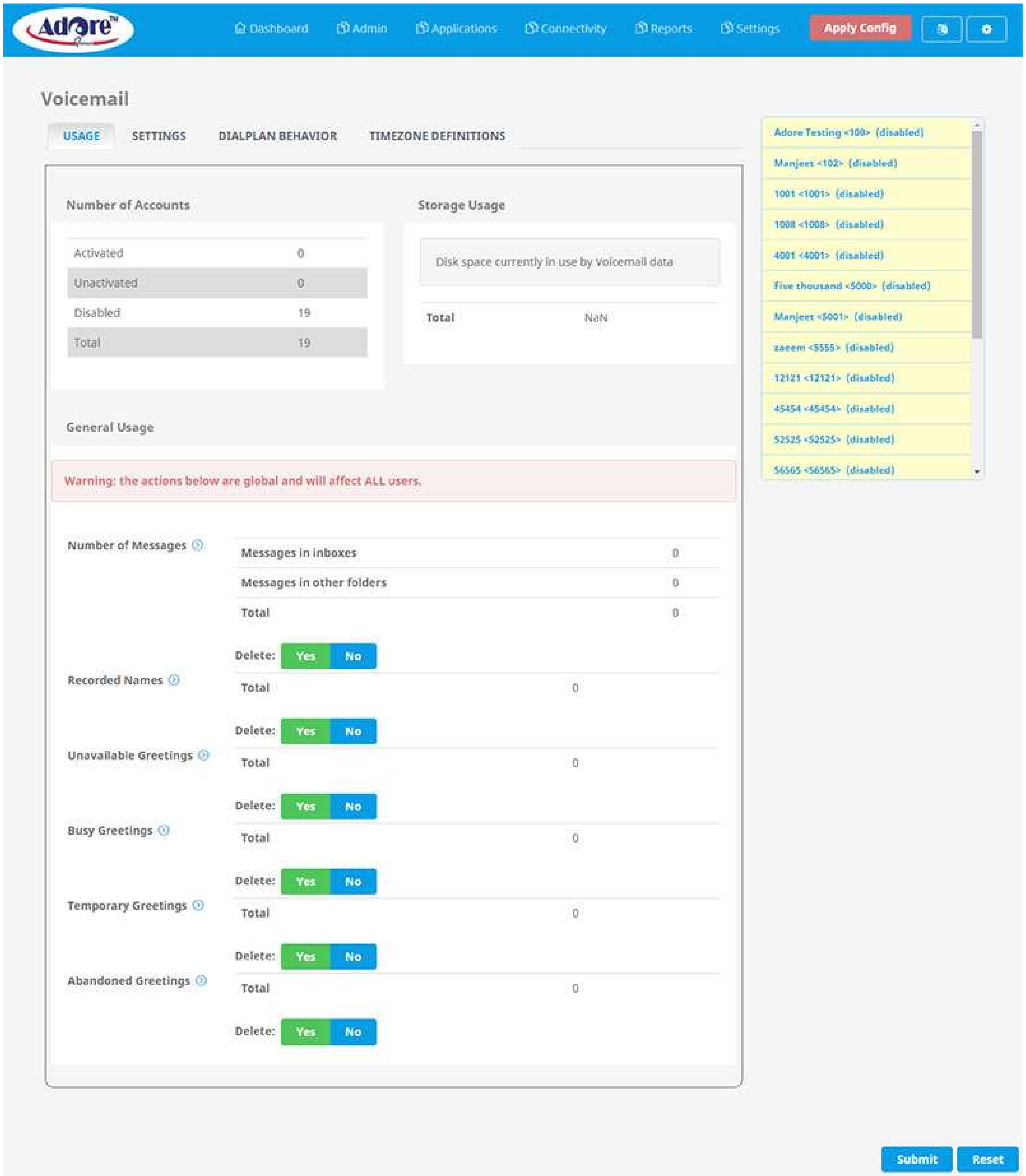
Go to **Settings** -> **Voicemail Admin**



On click **Voicemail Admin** Following screen will appear.

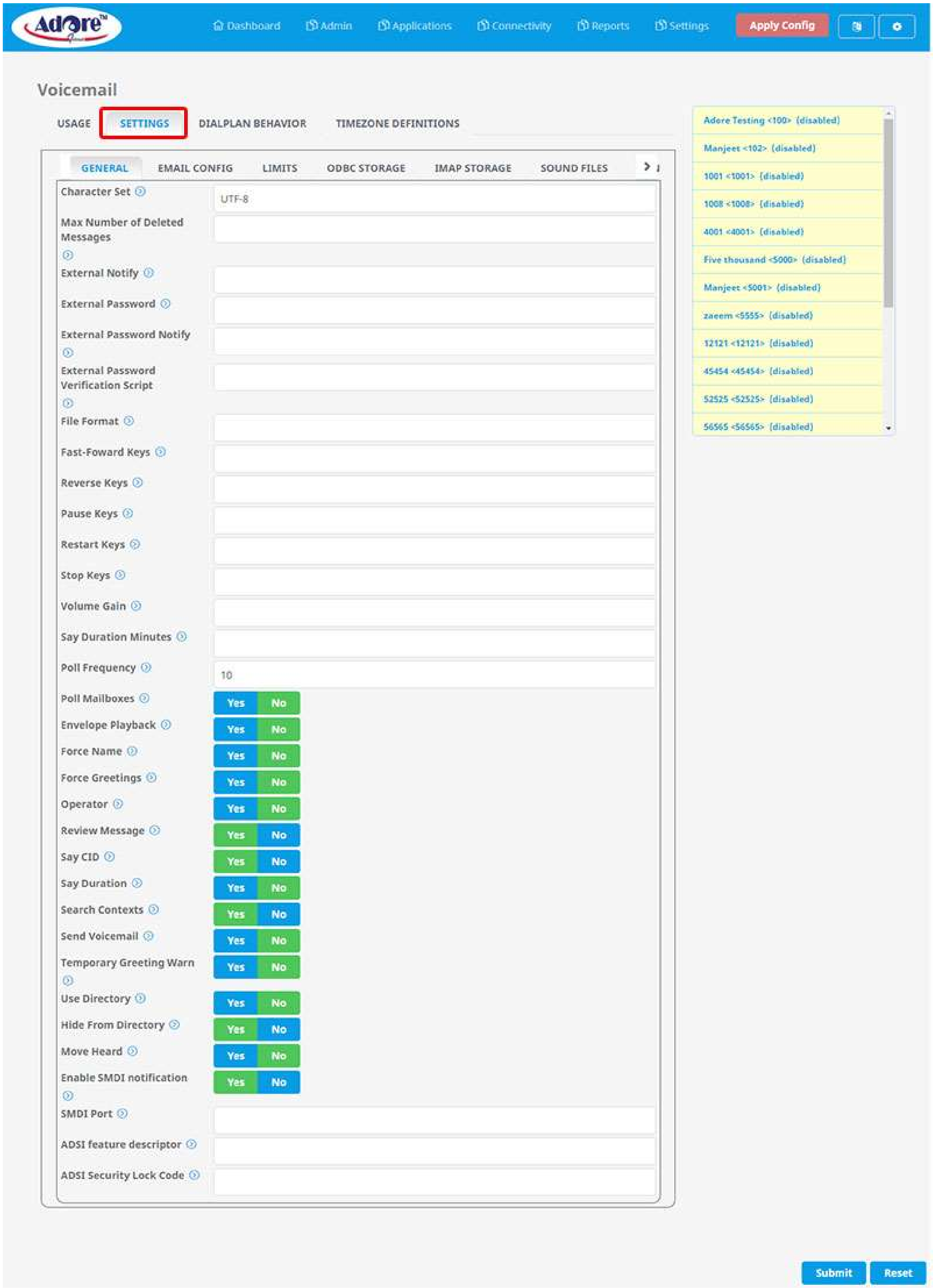
Usage

Here you can set Voicemail as per given settings option.



Settings

These settings are for advanced use cases. In most cases, the defaults will work. These should not be touched unless you know what you are doing.



Dialplan Behavior

These settings are for advanced use cases. In most cases the defaults will work. These should not be touched unless you know what you are doing.

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

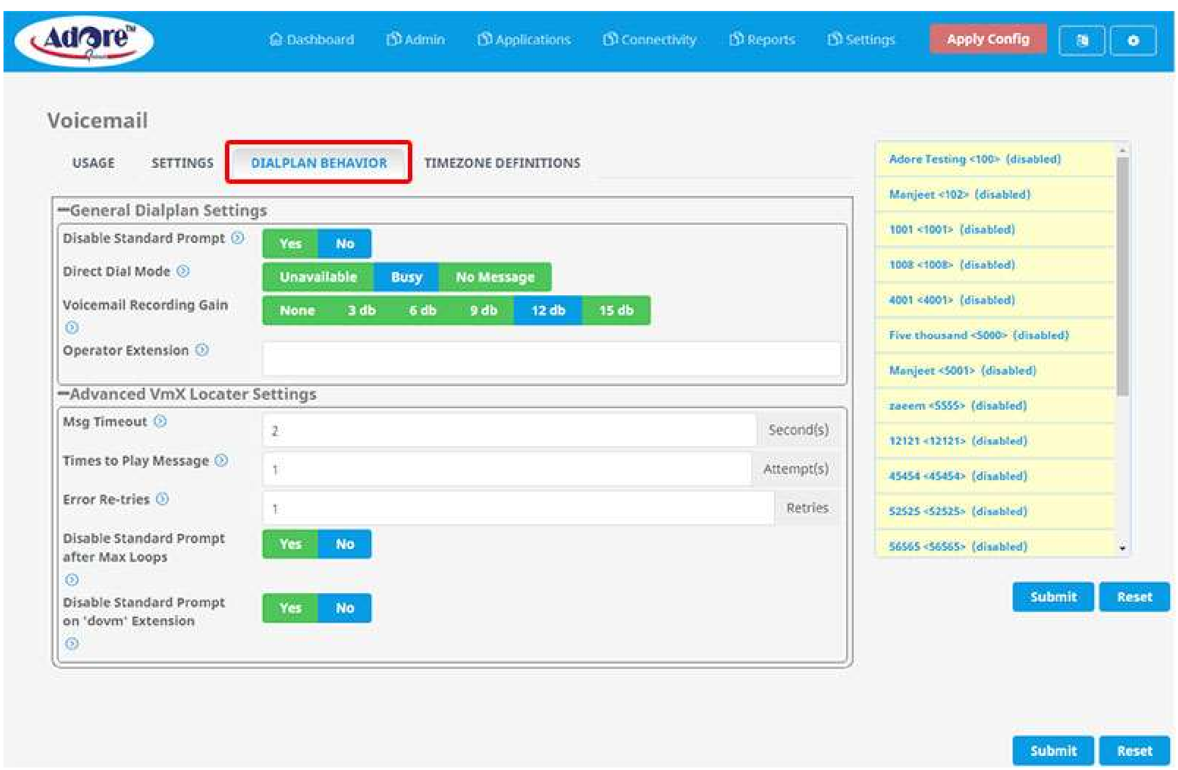
Number of times to play invalid options and repeat the message upon receiving an undefined option. One retry means it will repeat at one time after the intial failure.

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.



Timezone Defenations

These settings are for advanced use cases. In most cases the defaults will work. These should not be touched unless you know what you are doing.

New Name

Descriptive name for time zone definition

New Timezone Definition

Time announcement for message playback.

Timezone definition format is: **timezone|values**

7. UCP (User Control Panel)

UCP (User Control Panel)

User Control Panel (UCP) is a replacement for the aging ARI module in PBX. It currently has the following modules/sections.

- **Call History**
- **Call Events Logs**
- **Contact**
- **Presence**
- **Endpoint Settings**

7.1. Call History

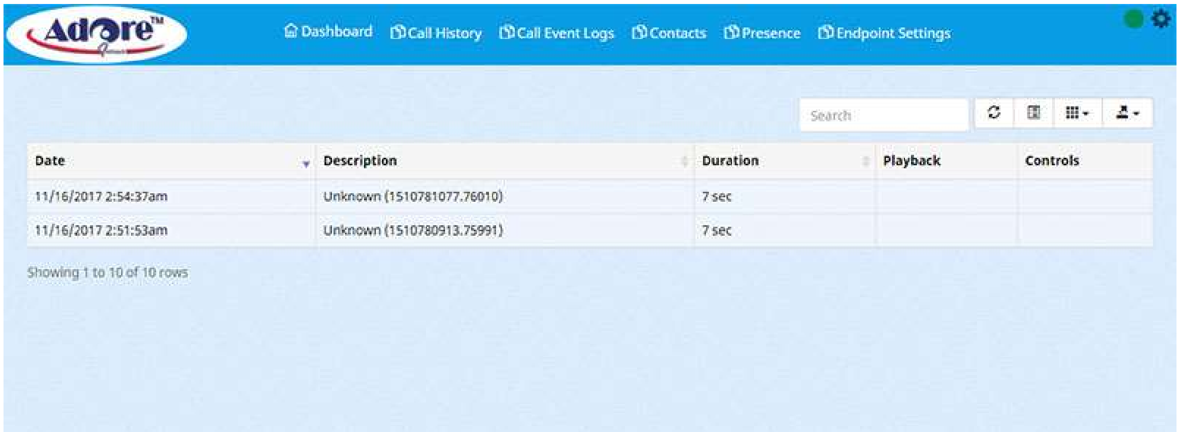
Call History

The Call History section allows you to see all inbound and outbound calls for your user and listen to any call recordings that are associated with that call.

Click on **Call History**

In our example we can see my user is setup to to view numerous extensions as defined in the User Management module. When clicking on Call History I pick which extension I want to view calls for.

Once we pick a extension we can now see our recent call history.



For each Call record we have the following

Date- Date and Time call was received or placed.

Description-For each call we have the following icons under the Description

Duration- Length of call in Hours- Minutes-Seconds

Controls- If the call has a Call Recording associated with it you will get a play and download icon for the Call Recording.

You can search for any call by using the Search bar at the top and putting in the number of the person you called or who called you and pressing GO

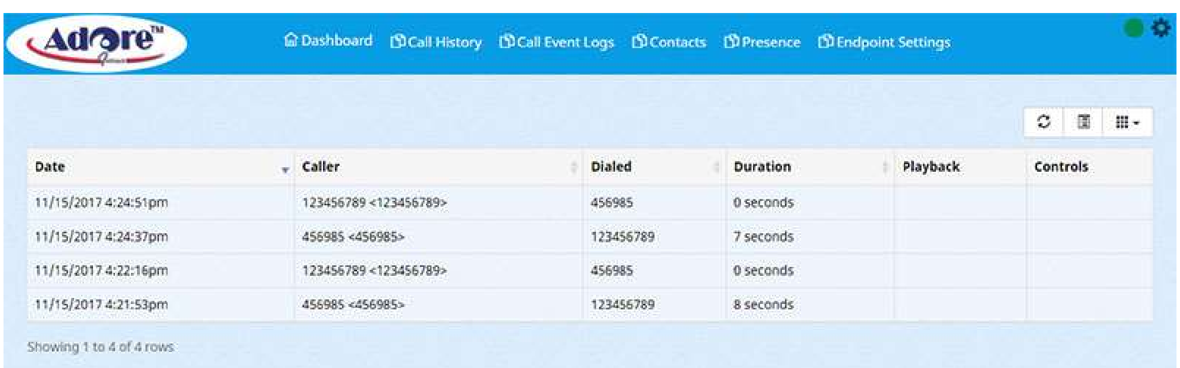
7.2. Call Events Logs

Call Events Logs

The Call Events widget allows you to see all inbound and outbound calls for your user and listen to any call recordings that are associated with that call. The User Management Module controls which call events a user will be able to add as a widget in UCP.

Click On **Call Events Logs**

On click **Call Events Logs** following screen will appear.



Date	Caller	Dialed	Duration	Playback	Controls
11/15/2017 4:24:51pm	123456789 <123456789>	456985	0 seconds		
11/15/2017 4:24:37pm	456985 <456985>	123456789	7 seconds		
11/15/2017 4:22:16pm	123456789 <123456789>	456985	0 seconds		
11/15/2017 4:21:53pm	456985 <456985>	123456789	8 seconds		

Showing 1 to 4 of 4 rows

For each Call record we have the following

- **Date**: Date and Time call was received or placed.
- **Caller**: Who was called
- **Dialed**: Who dialed
- **Duration**: Length of call in Hours- Minutes-Seconds
- **Playback**: If the call has a Call Recording associated with it you will be able to listen to the recording in your browser
- **Controls**: If the call has a Call Recording associated with it you will get a download icon for the Call Recording.

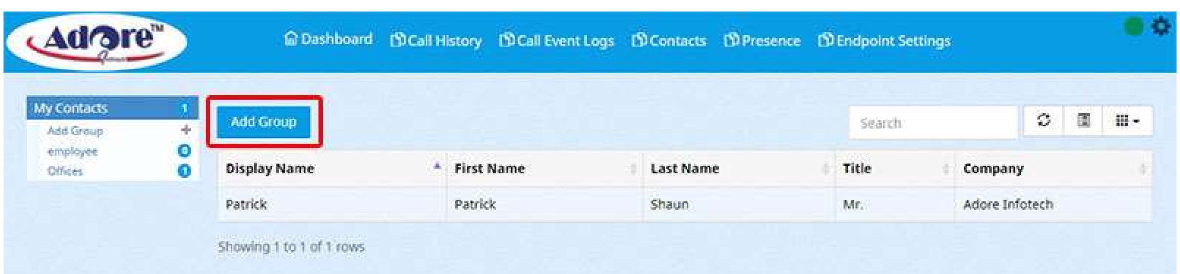
7.3. Contact

Contacts

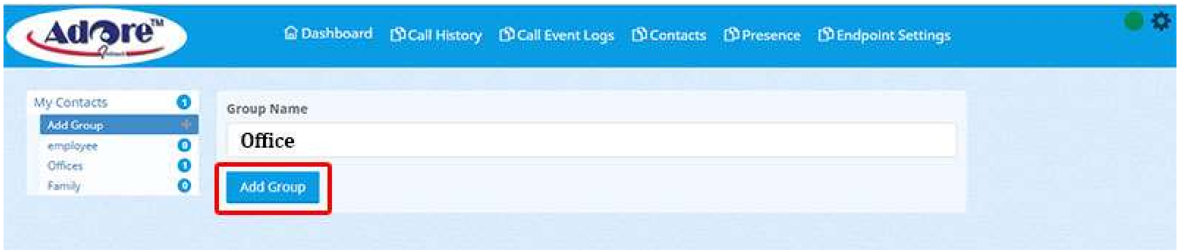
The Contacts widget allows you to see and create contacts that can be used in other widgets in UCP, Phone Apps and for speed dials.. The User Management Module controls which contact groups a user will be able to see in this widget.

Click on **Contacts**

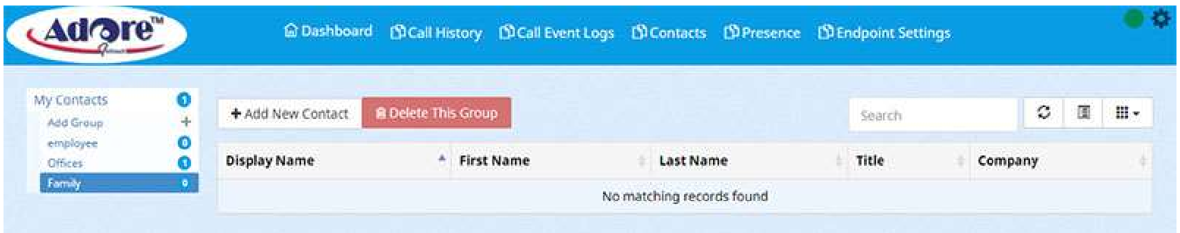
On click **Contacts** following screen will appear. For adding group click on **Add Group** button.



On click Add Group following screen will appear, here you can add group as per your wish.



After adding Group you can add contact under your created groups and also you can delete contact which you can added.



7.4. Presence

Presence

Presence state allows users to set different statuses for themselves depending on what they are currently doing. In UCP there are two places to add presence functionality. Through a widget or through the global side bar functionality.

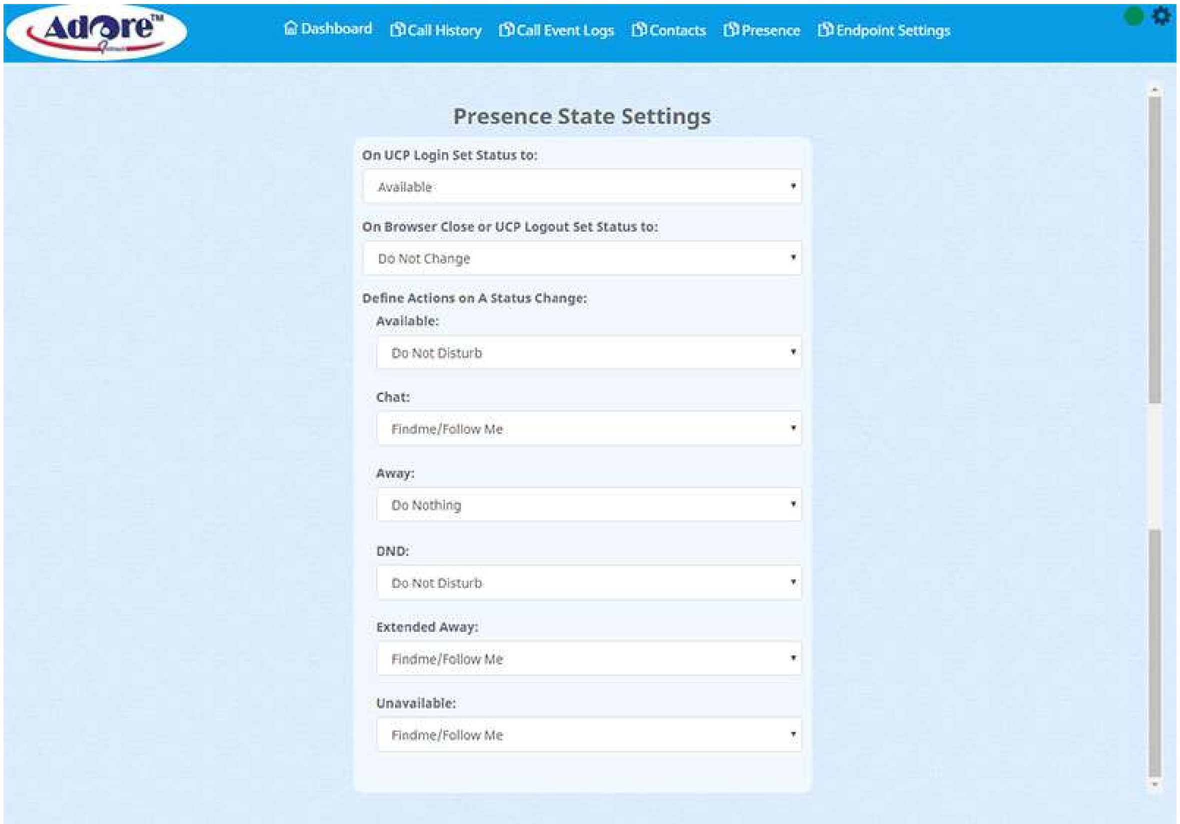
Presence allows you to control how the PBX interacts with you as a user.

Presence gives the system and other users in your organization a generalized status so they may respond accordingly.

Click on **Presence**

On click **Presence** following screen will appear.

These are managed by the PBX administrator using the Presence State Module



What are the state types?

Available

Means you are Available and around and able to take calls or chat.

Chat

Usually means you prefer to be contacted by chat only and not a phone call.

Away

Means you are away usually for a short period of time such as at lunch or in a meeting.

 # DND

"I am busy, do not bother me" – such as in a meeting.

 # Extended Away

Usually used when gone for long periods of time such as vacation or business meeting.

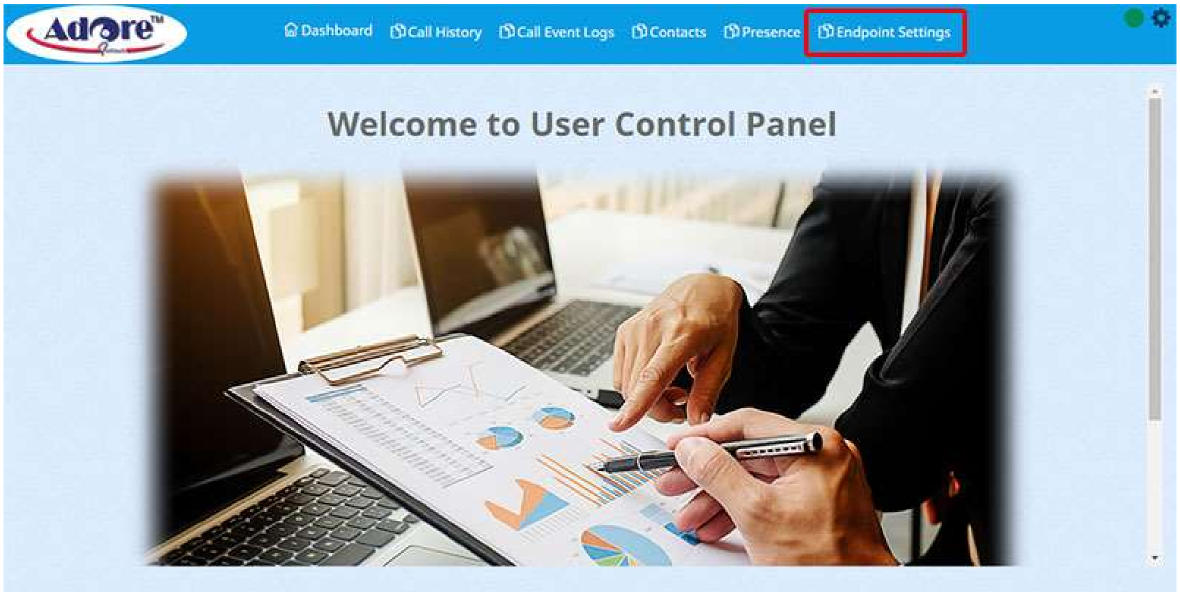
 # Unavailable

Means you generally won't show as online in a chat client.

7.5. Endpoint Settings

Endpoint Settings

Click on **Endpoint Settings**

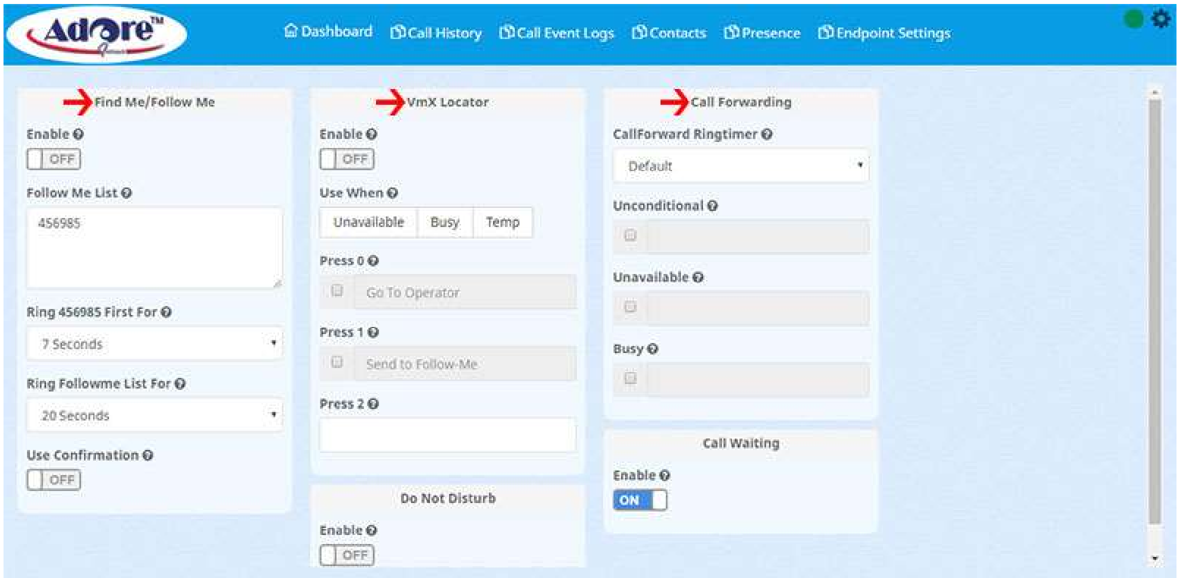


Here User can set following things :

*** Find Me/Follow Me**

*** VmX Locator**

*** Call Forwarding**



Adore
Dashboard
Call History
Call Event Logs
Contacts
Presence
Endpoint Settings
Find Me/Follow Me
Enable
OFF
Follow Me List
456985
Ring 456985 First For
7 Seconds
Ring Followme List For
20 Seconds
Use Confirmation
OFF
VmX Locator
Enable
OFF
Use When
Unavailable
Busy
Temp
Press 0
Go To Operator
Press 1
Send to Follow-Me
Press 2
Do Not Disturb
Enable
OFF
Call Forwarding
CallForward Ringtimer
Default
Unconditional
Unavailable
Busy
Call Waiting
Enable
ON

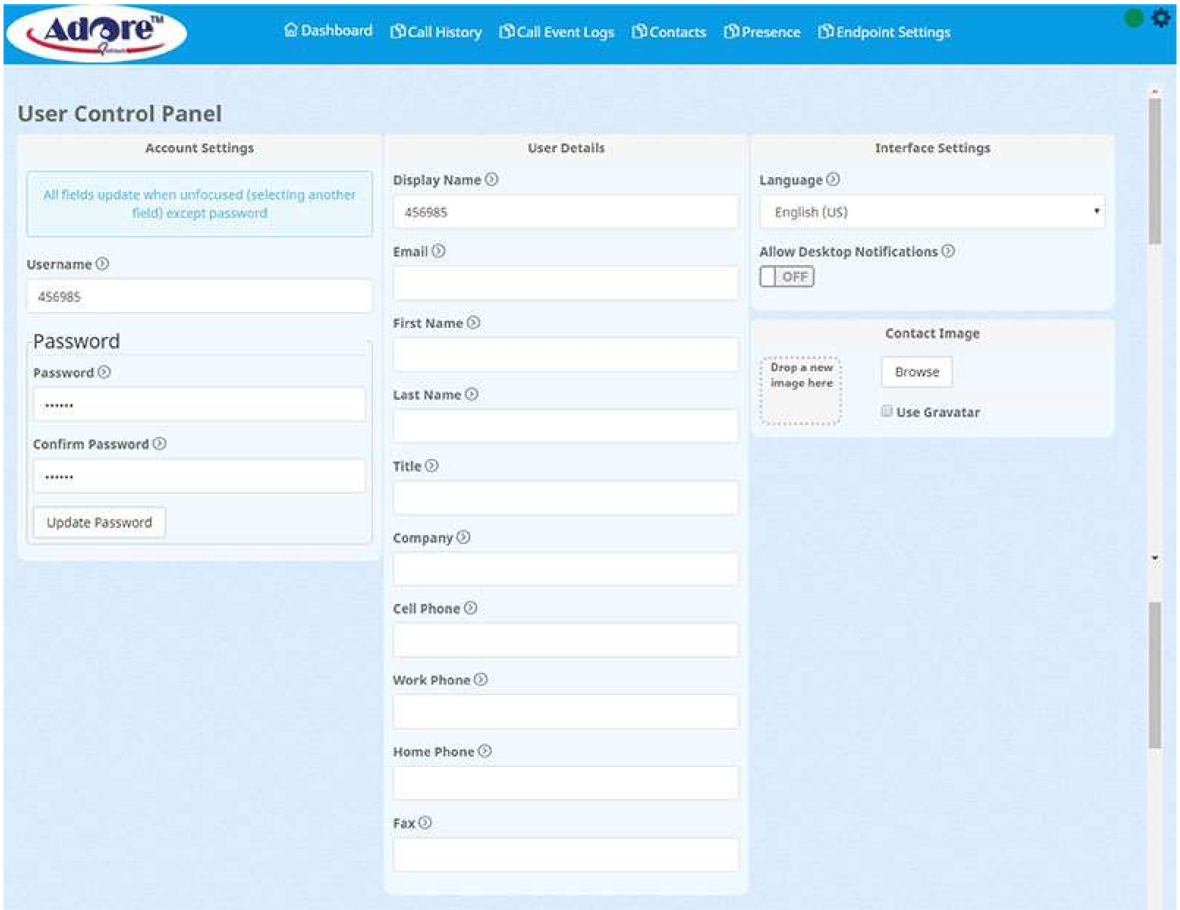
7.6. User Settings

User Settings

Here User can set own preferred settings as per own wish.

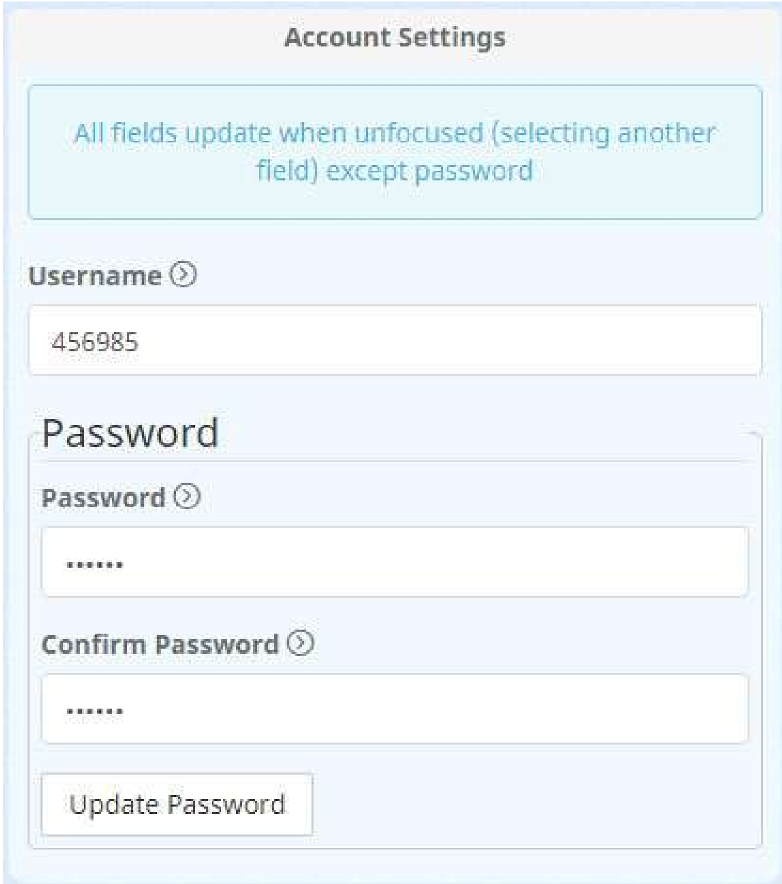
User Settings can be found in the **Top Right hand corner** of UCP (User Control Panel). Look for a **gear** icon, click on that than appear a option **User Settings** click on that.

On click User Settings following screen will appear.



Account Settings

Account Settings is the first tab. The options in here relate directly to your user. Some of these options may not be visible to you depending on your Admin configuration.



- **Username**: The Username of your account. You can change it here if you wish (you will be forced to logout)
- **Password**: The new password of your account. You can change it here but will need to reenter your password in the confirm box
- **Confirm Password**: The confirm password input. Used to validate you did not mistype your password
- **Update Password**: Click this button to update your password based on new input from Password and Confirm Password

User Details

User details is the second tab. This tab contains all of the details about the currently logged in user on the system

- **Display Name**: Alias/Name used in services such as Chat
- **Email**: Email address used for resetting the password
- **First Name**: The user's first name

- **Last Name**: The user's last name
- **Title**: The user's title

Interface Settings

The last tab is the Interface Settings tab.

Language: Select the language you'd like UCP to work in. Not all languages are fully translated. To help participate in translations see Translating IPPBX

Avatar: The user's avatar

You can drag and drop an avatar here or click the "Use Gravatar" function which will use the email address provided above to lookup an avatar from the Gravatar network