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1. Getting Started

Adore Multi Tenant PBX

The term "**Multi Tenant**" refers to a software architecture in which a single instance of software runs on a server and serves multiple tenants. A tenant is a group of users who share a common access with specific privileges to the software instance. With a multitenant architecture, a software application is designed to provide every tenant a dedicated share of the instance – including its data, configuration, user management, tenant individual functionality and non-functional properties.

2. Multi Tenant PBX

Multi Tenant PBX Portal Overview

Please visit **https://pbx.adoreinfotech.co.in/** and login with following credentials

Username : admin

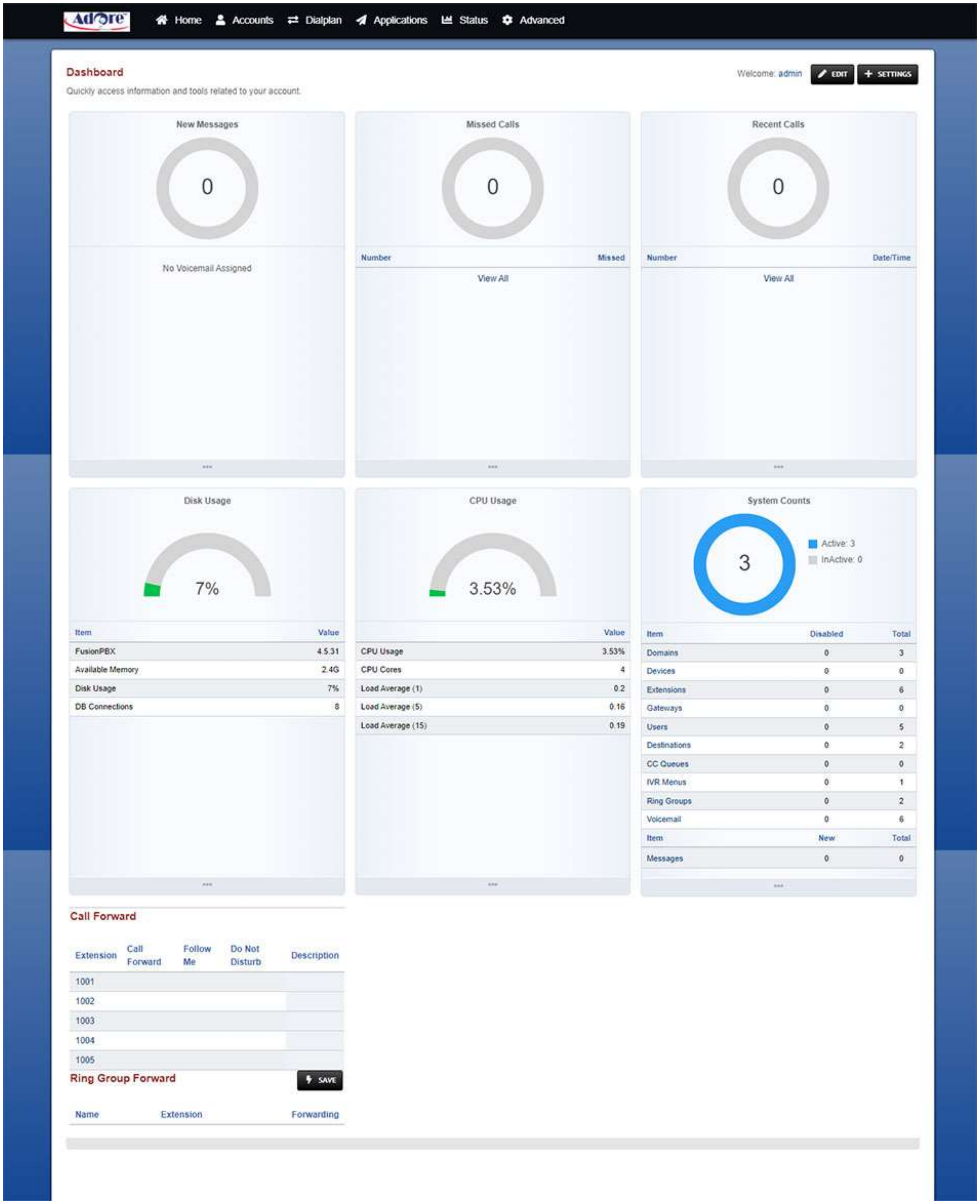
Password : 1@Qvvsfcs^vsvs!C

3. Portal Dashboard

After login you are on dashboard of Adore Multi Tenant PBX.

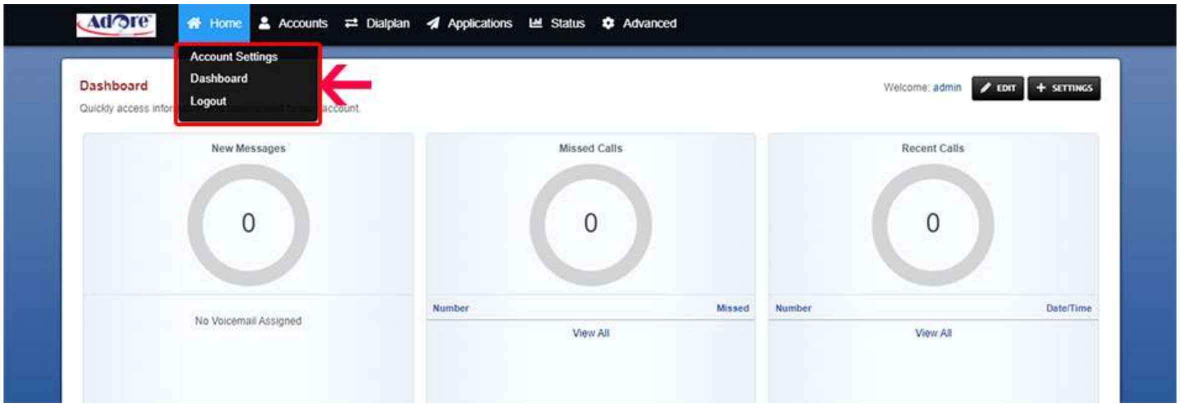
In Dashboard you can find following options:

- Home
- Accounts
- Dialplan
- Applications
- Status
- Advanced


Home
Accounts
Dialplan
Applications
Status
Advanced
Dashboard
Quickly access information and tools related to your account.
Welcome: admin
EDIT
SETTINGS
New Messages
0
No Voicemail Assigned
Missed Calls
0
Number
Missed
View All
Recent Calls
0
Number
Date/Time
View All
Disk Usage
7%
Item
Value
FusionPBX
4.5.31
Available Memory
2.4G
Disk Usage
7%
DB Connections
8
CPU Usage
3.53%
Value
CPU Usage
3.53%
CPU Cores
4
Load Average (1)
0.2
Load Average (5)
0.16
Load Average (15)
0.19
System Counts
3
Active: 3
InActive: 0
Item
Disabled
Total
Domains
0
3
Devices
0
0
Extensions
0
6
Gateways
0
0
Users
0
5
Destinations
0
2
CC Queues
0
0
IVR Menus
0
1
Ring Groups
0
2
Voicemail
0
6
Item
New
Total
Messages
0
0
Call Forward
Extension
Call Forward
Follow Me
Do Not Disturb
Description
1001
1002
1003
1004
1005
Ring Group Forward
SAVE
Name
Extension
Forwarding

4. Home

The **Home** menu gives access to Account Settings, Dashboard and the option to Logout.



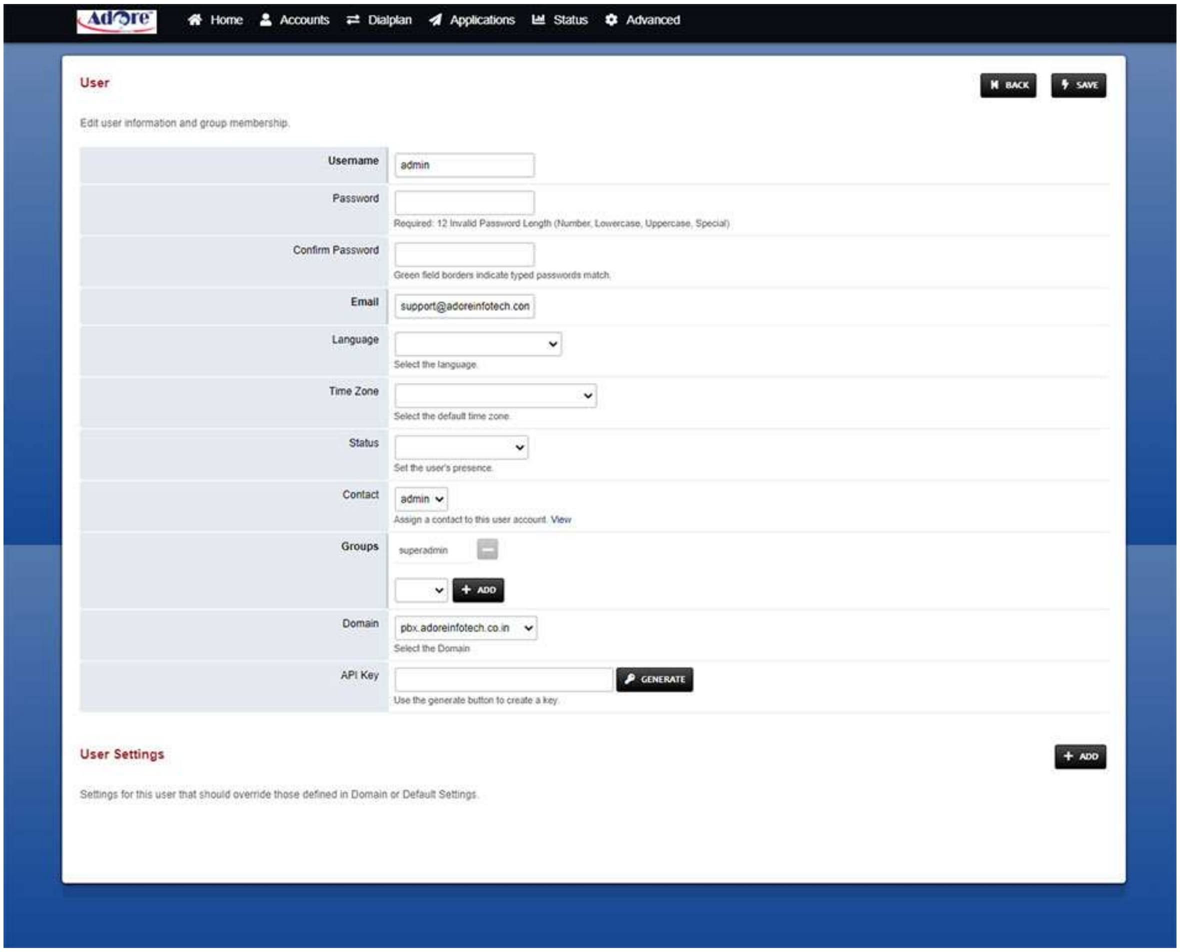
Account Settings

Go to **Home** - **Account Settings**

Here you can set your portal account settings

User Name	You can set here your portal login user name
Password	You can set or reset here your portal login password
Confirm Password	Must match the password.
Language	Choose the default language for the portal.
Time Zone	You can set here Time zone of the portal.
Status	Used for call center and operator panel.
Contact	The users contact. Is used in a phone directory or Apps > Contacts.
Groups	Group the user is in and relates to what the user can see and do in the menus.
Domain	Here you can set your portal Domain name.

Enabled	Here you can Enable or disable the account of portal



Dashboard Settings

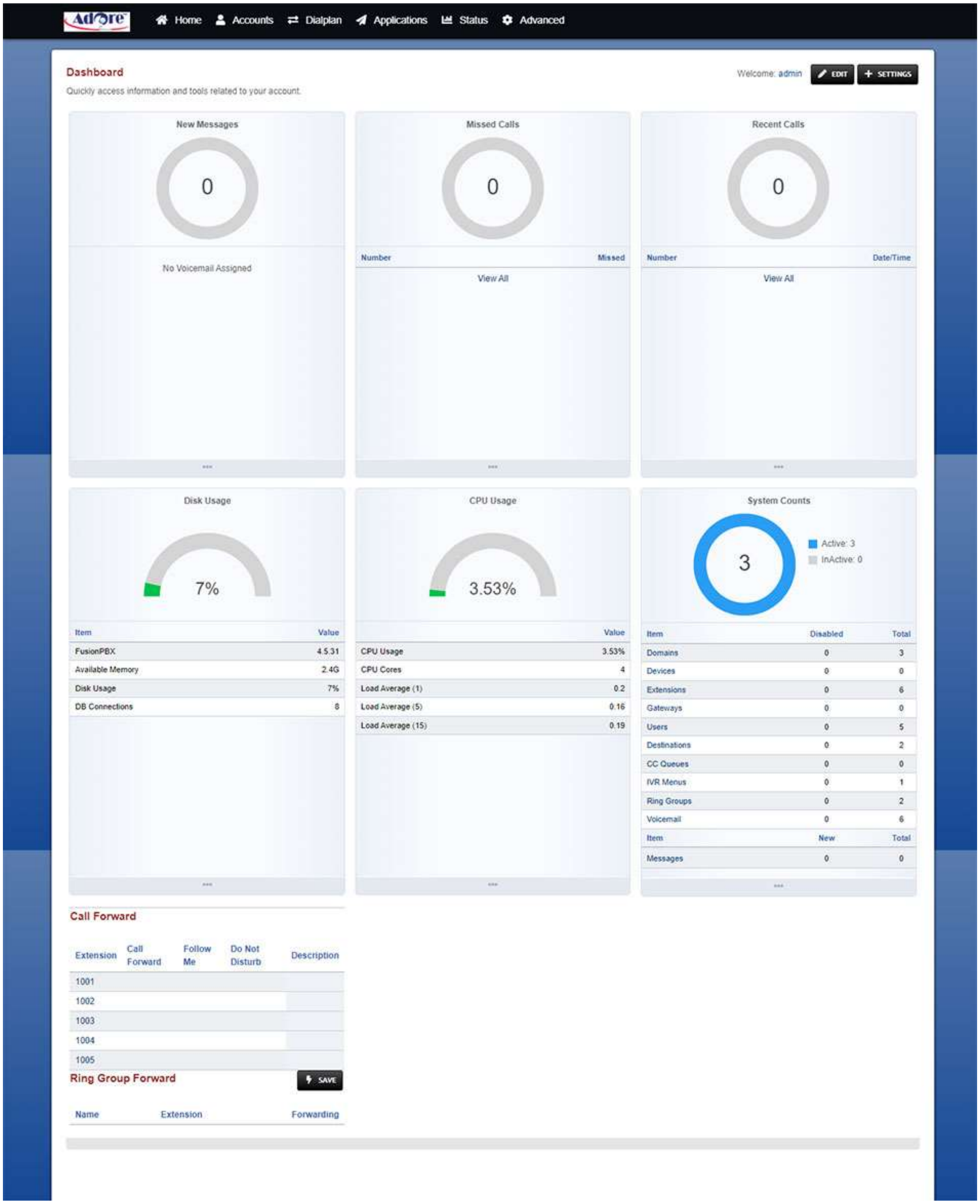
Go to **Home** - **Dashboard Settings**

Here you can set your portal Dashboard settings

Quickly access information and tools related to your account. Depending on the user permissions, the user may see less options on this screen.

Voicemail	Here you can see New and total voicemails related to the users voicemail box. A user can be assigned to more than 1

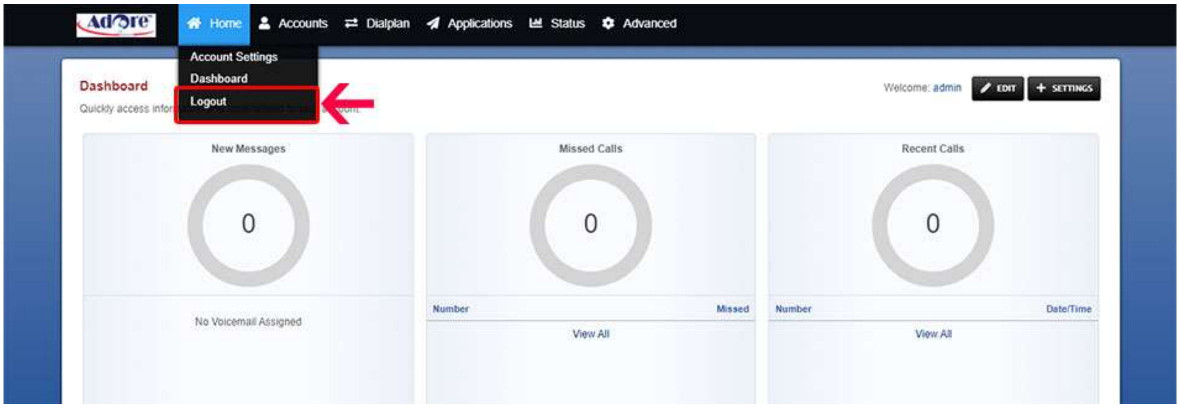
	voicemail box.
Missed Calls	Here you can see Missed calls for the user.
Recent Calls	Here you can get Number of calls in the last 24 hours.
System Status	Here you can get details of your system status.
Call Routing	Here you can See if call forward, follow me, do not disturb is set and a quick wait to edit those options if needed.
Ring Group Forward	Here you can See the name, extension number, if forwarding is enabled and what number it is forwarded to.
System Counts	Here you can see system Number of Domains, Devices, Extensions, Gateways, Users, Destinations, CC Queues, IVR Menus, Ring Groups, Voicemail and if they are disabled.



Logout

Go to **Home** - **Logout**

You can logout from the system.

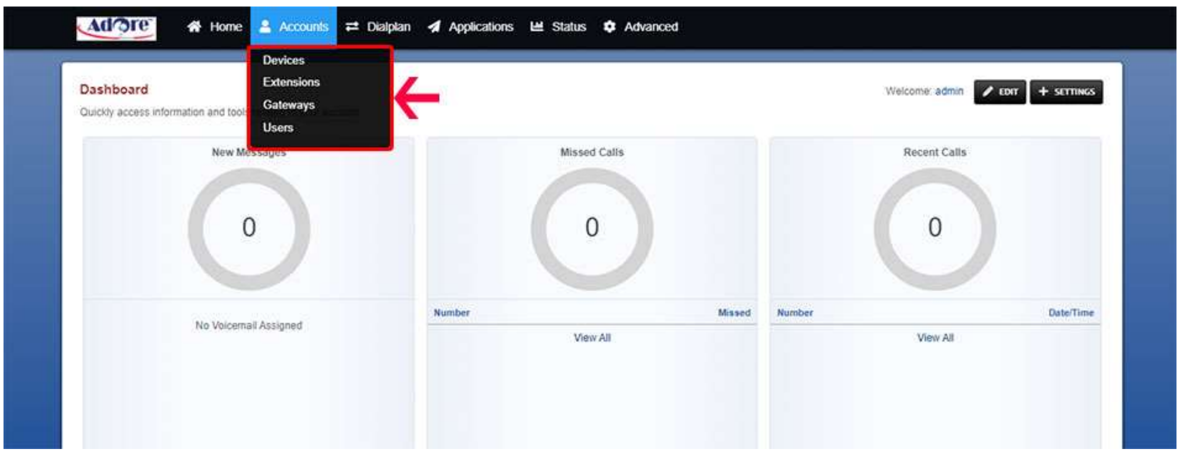


5. Accounts

Accounts

In the Accounts menu you have access to followings options

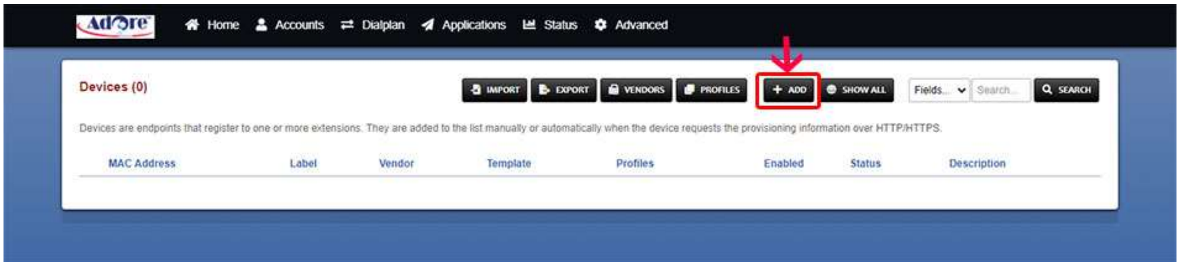
- **Devices**
- **Extensions**
- **Gateway**
- **Users / Provider**



Devices

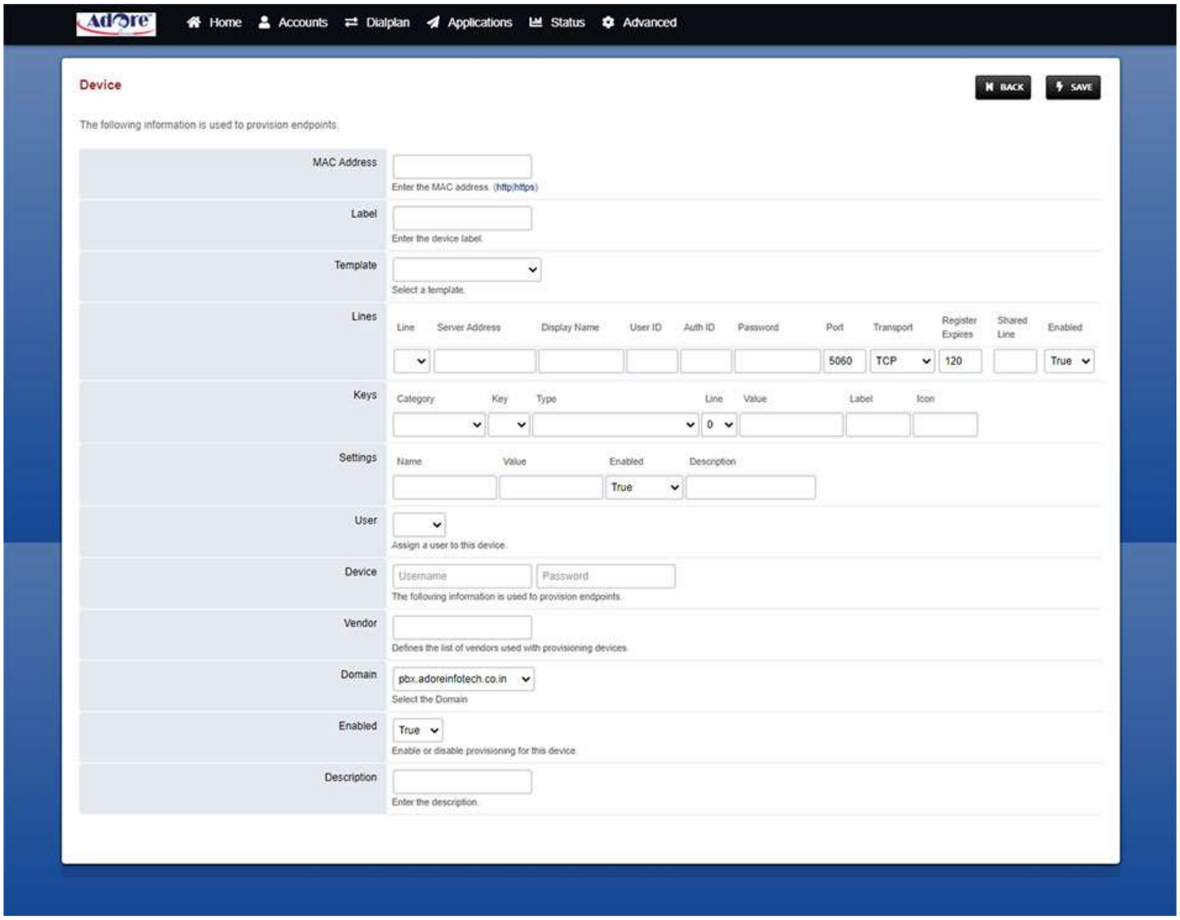
Here you can define the information needed to assign SIP accounts and keys to provision the devices.

Click the **Add** button to add a device.



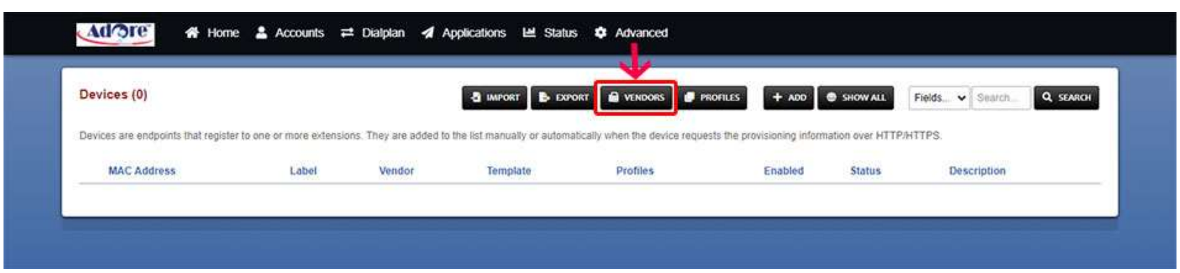
On click Add Button, following screen will open. please enter the required details and add device in to the system

- Enter the mac address of the phone.
- Add a label.
- Select from the drop down box the make/model.
- Populate the lines section.
- Populate the Key section.
- (Optional) Populate the Settings section. These settings are the same as the variables from Advanced > Default Settings > Provisioning and can be overridden in this settings section. Just set the variable for the device you are adding.
- Edit other fields as needed.
- Click Save



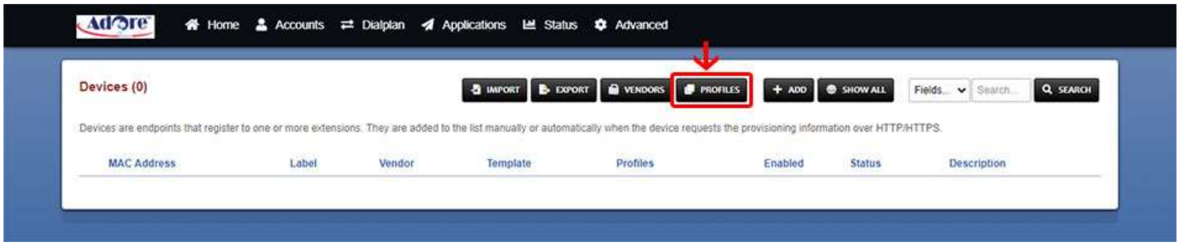
Device Vendor

Here Vendors can be added or removed to help fine tune the devices page when configuring specific vendor phones.

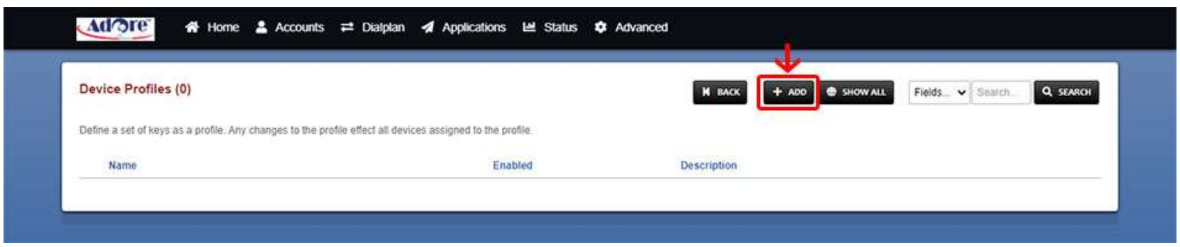


Profiles

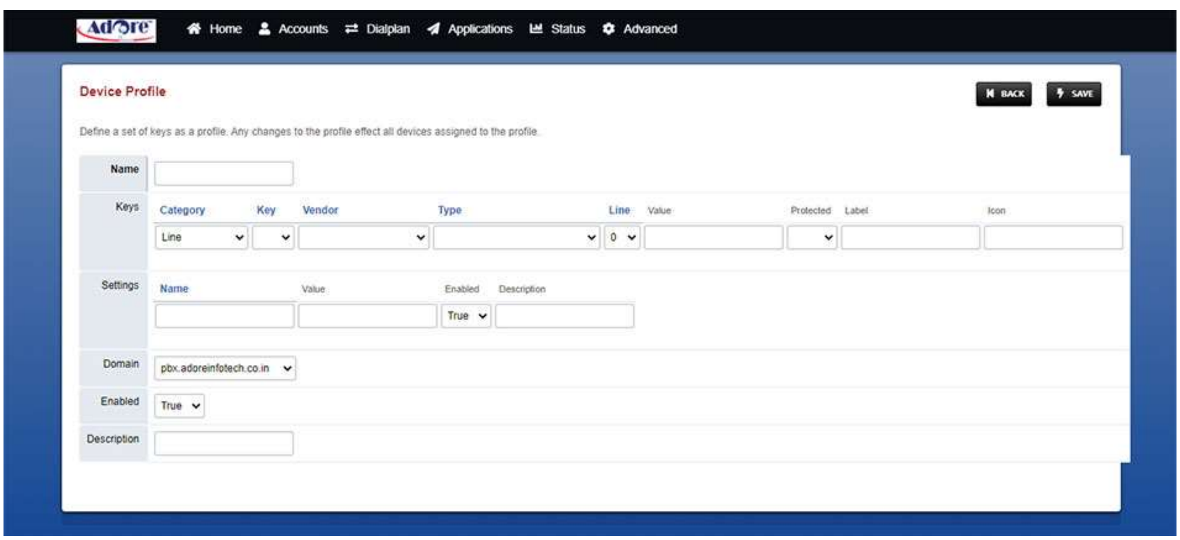
Define a set of keys as a profile. Any changes to the profile effect all devices assigned to the profile.



Click Add button to add profile of devices

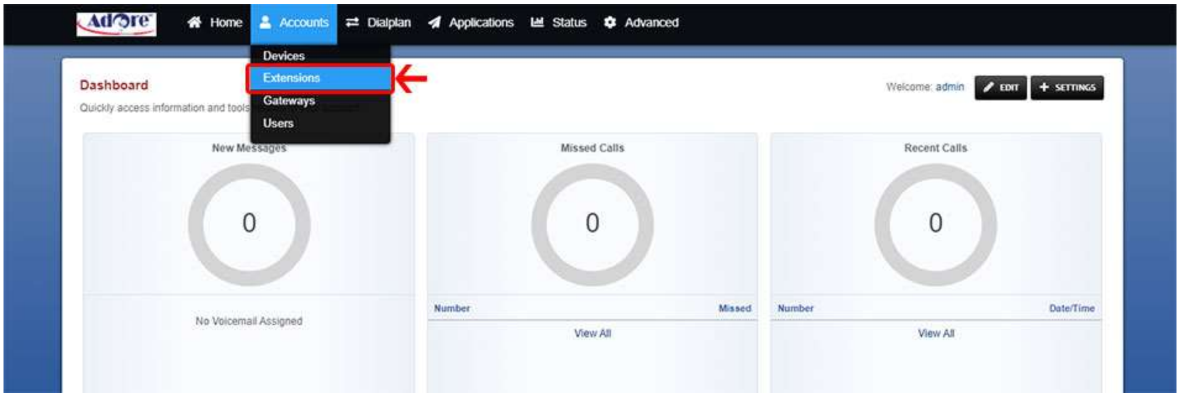


Add details of profile and save.

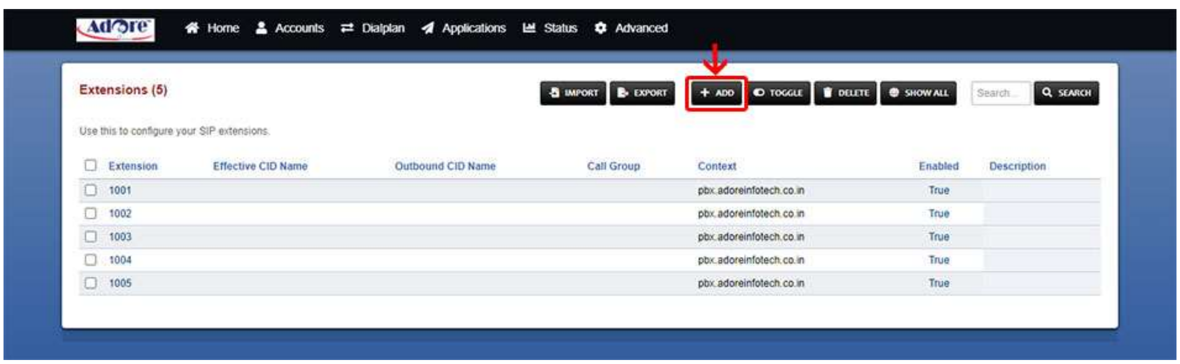


Extensions

Extensions define the information needed for an endpoint such as a hard phone, soft phone or some other device to connect to the SIP server. The extension is the **SIP username** and the **password** is the secret used for authentication. The domain name servers (DNS) to purposes it, locates the server to register to and is the realm that determines which domain the endpoint is registering to.



Click **Add** button to add new extension in to the system



Fill the Extension details and click save button

Basic Settings

Extension	Enter the alphanumeric extension. The default configuration allows 2 - 7 digit extensions.
Number Alias	If the extension is numeric then number alias is optional. The primary purpose of this field is when the extension is not a number then the number alias is required. Note a numeric extension and number alias does not currently work.

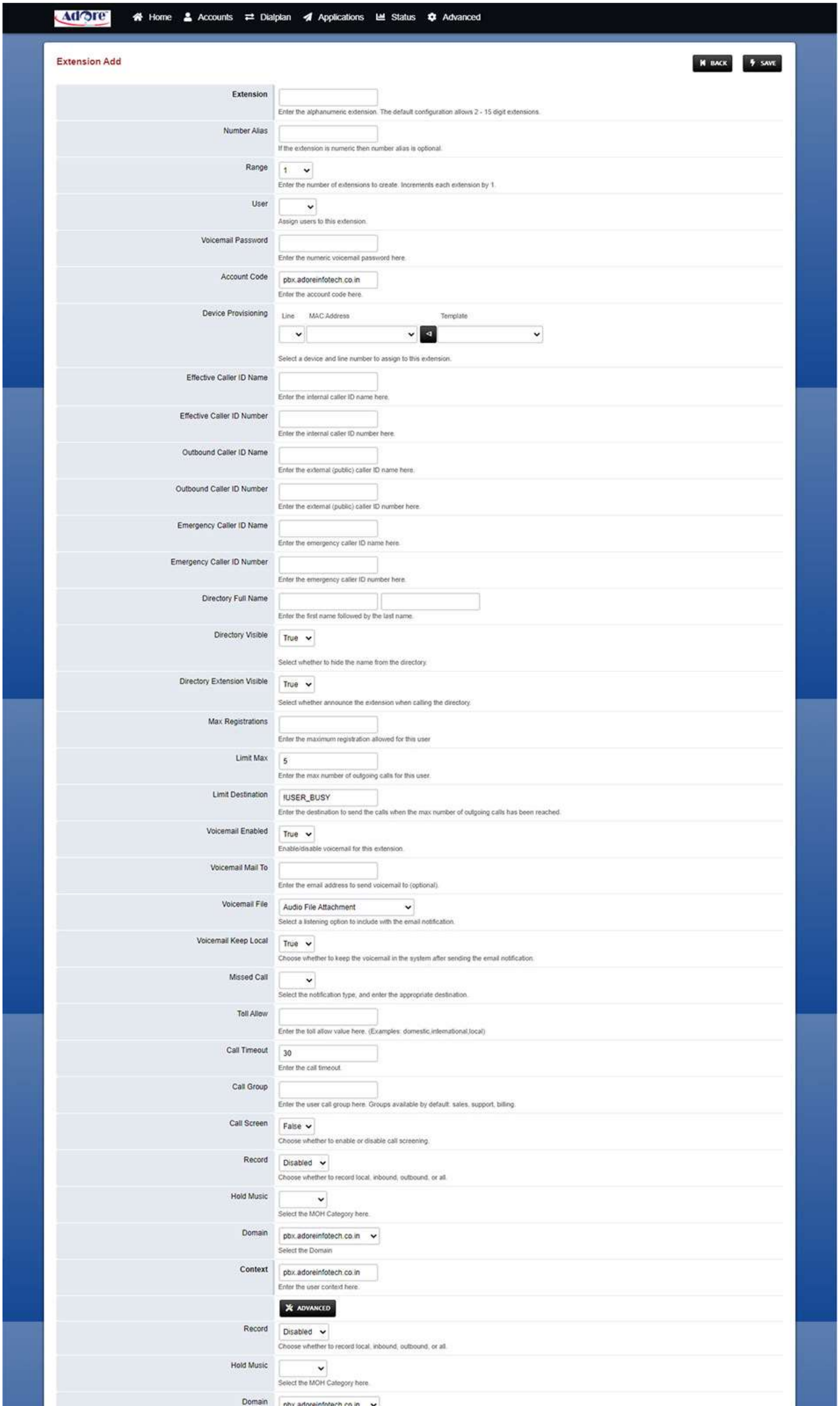
Range	Enter the number of extensions to create. Increments each extension by 1.
Voicemail Password	Enter the numeric voicemail password here.
Account Code	Used with billing systems if you don't have a billing system then its optional.
Effective caller ID Name	Enter Internal Caller ID name
Effective Caller ID Number	Enter Internal caller ID number usually set to the extension number.
Outbound Caller ID Name	Used by the outbound route for external caller ID name. Business or Organization typically is set here.
Outbound Caller ID Number	Used by the outbound route for external caller ID number here. Business or Organization number goes here.
Emergency Caller ID Name	This is used when calling out to an emergency service like 911.
Emergency Caller ID Number	This is used when calling out to an emergency service like 911.
Directory Full Name	The first and last name used in the directory. You can call that directory with *411
Directory Visible	Select whether to hide the name from the directory.
Directory Extension Visible	Select whether announce the extension when calling the directory.
Limit Max	Set max number of outgoing calls for this user.
Limit Destination	Set the destination to send the calls when the max number of outgoing calls has been reached.
Voicemail Enabled	Enable or disable voicemail for this extension.
Voicemail Mail To	The email address for sending voicemail to email.
Voicemail File	Select whether to send the voicemail as an attachment or as a link in the email.

Voicemail Keep Local	Choose whether to keep the voicemail in the system after sending the email notification.
Missed Call	Set the missed call to true and set the email address if you want to receive an email for missed calls that were routed through the dialplan to and was not answered by the extension.
Toll Allow	Enter the toll allow value here. (Examples: domestic, international, local) This can be set to any name you want it sets a variable that can be a condition on the outbound routes.
Call Timeout	Set the timeout for the call ringing.
Call Group	You can define any call group you want the following groups are examples: sales, support, billing. These are used for group intercept or calls can be sent to the call group.
Call Screen	If set will ask the caller to identify themselves. Their response will be recorded and offered to the person receiving the call.
Record	Whether to record local, inbound, outbound, or all calls that were sent directly to this extension.
Hold Music	Select music or ring tones that will be used for music on hold for this extension.
Context	The context is set by default to match the domain name or IP address. It is usually correct by default and doesn't need to be changed in most cases.
Enabled	Extension enabled or disabled.
Description	A description for the extension.

Advanced Settings

Advanced settings in extensions. Be sure to know what and why you are changing these settings or you will risk causing issues for the extention.

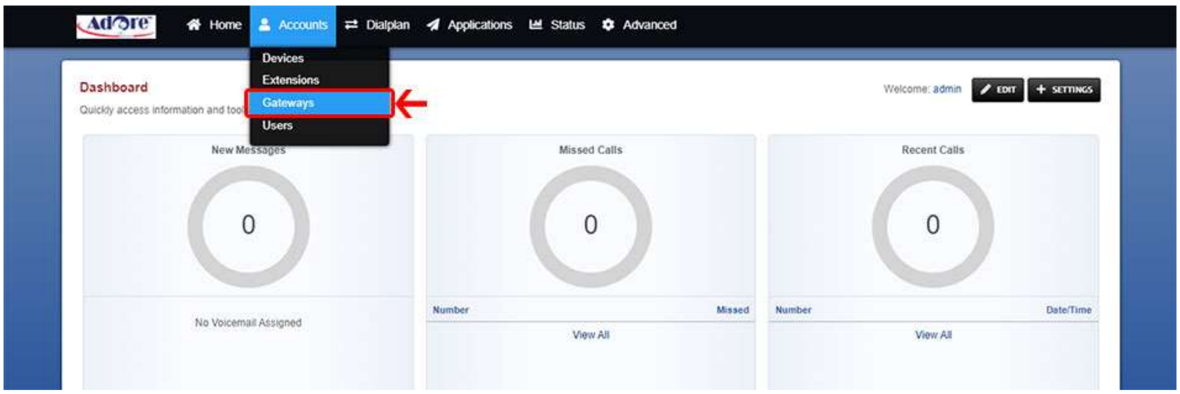
Auth ACL	Advanced auth acl uses.
CIDR	Advanced cidr uses.
SIP Force Contact	Choose whether to rewrite the contact port, or rewrite both the contact IP and port.
SIP Force Expires	To prevent stale registrations SIP Force expires can override the client expire.
MWI Account	MWI Account with user@domain of the voicemail to monitor.
SIP Bypass Media	Choose whether to send the media stream point to point or in transparent proxy mode.
Absolute Codec String	Absolute Codec String for the extension.
Force ping	Use OPTIONS to detect if extension is reacheable.
Domain	The domain the extension is currently saved on.
Dial String	Location of the endpoint.



Home
Accounts
Dialplan
Applications
Status
Advanced
Extension Add
BACK
SAVE
Extension
Enter the alphanumeric extension. The default configuration allows 2 - 15 digit extensions.
Number Alias
If the extension is numeric then number alias is optional.
Range
1
Enter the number of extensions to create. Increments each extension by 1.
User
Assign users to this extension.
Voicemail Password
Enter the numeric voicemail password here.
Account Code
pbx.adoreinfotech.co.in
Enter the account code here.
Device Provisioning
Line
MAC Address
Template
Select a device and line number to assign to this extension.
Effective Caller ID Name
Enter the internal caller ID name here.
Effective Caller ID Number
Enter the internal caller ID number here.
Outbound Caller ID Name
Enter the external (public) caller ID name here.
Outbound Caller ID Number
Enter the external (public) caller ID number here.
Emergency Caller ID Name
Enter the emergency caller ID name here.
Emergency Caller ID Number
Enter the emergency caller ID number here.
Directory Full Name
Enter the first name followed by the last name.
Directory Visible
True
Select whether to hide the name from the directory.
Directory Extension Visible
True
Select whether announce the extension when calling the directory.
Max Registrations
Enter the maximum registration allowed for this user
Limit Max
5
Enter the max number of outgoing calls for this user.
Limit Destination
!USER_BUSY
Enter the destination to send the calls when the max number of outgoing calls has been reached.
Voicemail Enabled
True
Enable/disable voicemail for this extension.
Voicemail Mail To
Enter the email address to send voicemail to (optional).
Voicemail File
Audio File Attachment
Select a listening option to include with the email notification.
Voicemail Keep Local
True
Choose whether to keep the voicemail in the system after sending the email notification.
Missed Call
Select the notification type, and enter the appropriate destination.
Toll Allow
Enter the toll allow value here. (Examples: domestic,international,local)
Call Timeout
30
Enter the call timeout.
Call Group
Enter the user call group here. Groups available by default: sales, support, billing.
Call Screen
False
Choose whether to enable or disable call screening.
Record
Disabled
Choose whether to record local, inbound, outbound, or all.
Hold Music
Select the MOH Category here.
Domain
pbx.adoreinfotech.co.in
Select the Domain
Context
pbx.adoreinfotech.co.in
Enter the user context here.
ADVANCED
Record
Disabled
Choose whether to record local, inbound, outbound, or all.
Hold Music
Select the MOH Category here.
Domain

Gateways

Gateways define the location and settings for other VoIP servers or Providers. After defining the Gateways use the Outbound routes to direct calls through the gateways. Required items are in bold. Its a good idea to start with the required items test it and then make adjustments as needed.

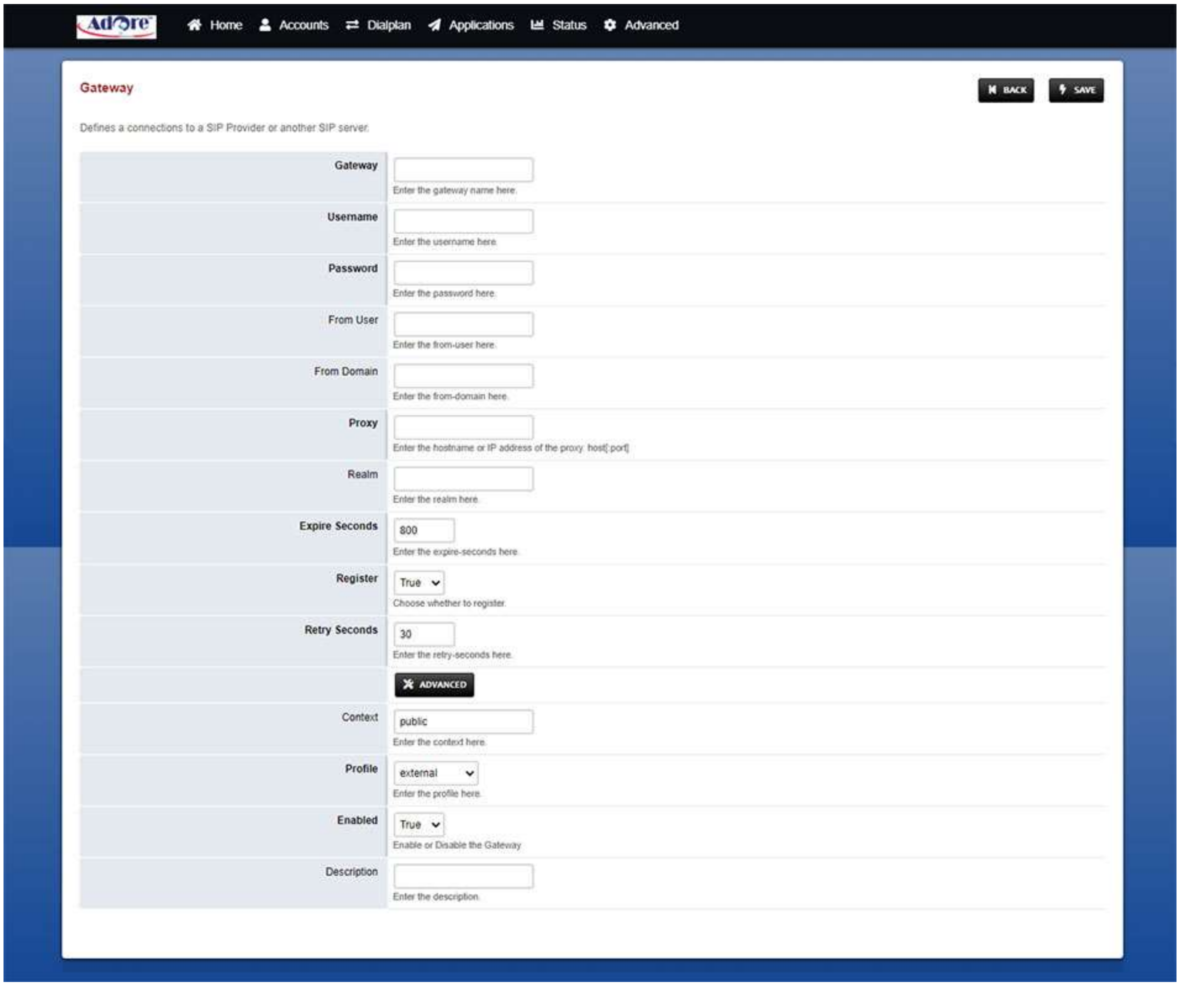


Gateways provide access into other voice networks. These can be voice providers or other systems that require SIP registration. Click on **Add** Button to add gateway in to the system

Basic Settings

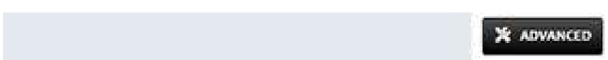
Gateway	The name of the Gateway. The company name or domain name of th VoIP provider is commonly used for the name.
Username	This is the username for SIP registration provided by the carrier.
Password	This is the password for SIP registrations it is provided by the

	carrier.
From User: (Optional*)	Set a specific SIP From User
From Domain: (Optional)	Sets a specific SIP From Domain.
Proxy:	Proxy server address used by the carrier. This will vary by carrier.
Realm : (Optional)	Required by some carriers
Expire Seconds: (Optional)	The time until the registration with carrier expires.
Register:	Set to true if the carrier uses a username and password. Set to false if the carrier uses IP authentication. If false, you will need to specify all of the carrier IP's in the **Advanced > Access Controls.**
Context:	Default is set to public and usually the correct value.
Profile:	The SIP profile used by default external is used. If you disable the external profile make sure to change the SIP profile to one that is enabled.
Hostname:	This should usually be left empty. When the hostname is set the gateway will only start on the matching server with same hostname. If the hostname is left blank the gateway will start regardless of the server's hostname.
Enabled:	If the gateway is enabled or disabled.
Description:	It is helpful to provide a good description for the gateway.



Advanced Settings

Most settings in the Advanced Gateway Settings can remain the same. Some carriers will require slight changes in this section to help with outbound caller ID.



Distinct To:	As per you.
Auth Username:	set as per you
Extension:	Usually used for testing and not for production. Hard codes a set number and all calls would be hard coded to that number for inbound calls from that gateway.

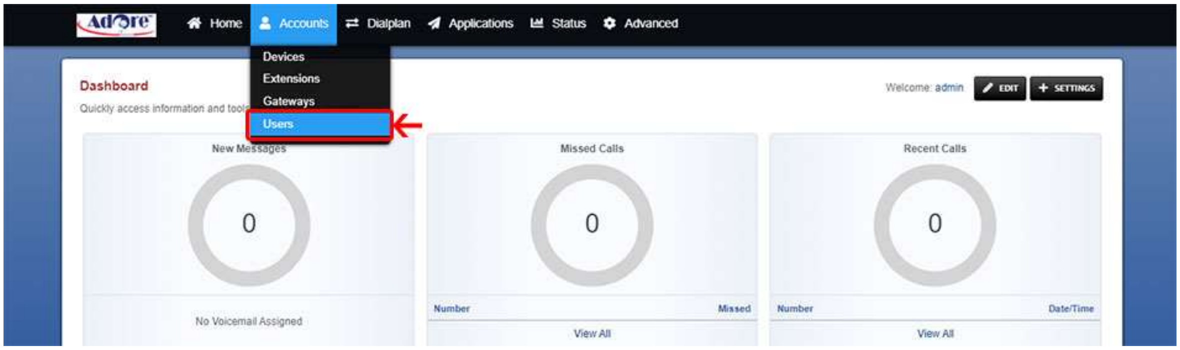
Register Transport:	Tells the switch to use SIP with TCP, UDP or TLS.
Register Proxy:	Enter the hostname or IP address of the register proxy. host[:port].
Outbound Proxy:	Enter the hostname or IP address of the outbound proxy. host[:port].
Caller ID In From:	If you caller ID isn't working setting this to true will often fix the problem.
Supress CNG:	Set this value to true to disable comfort noise.
Sip CID Type:	The SIP caller id type: none, pid, and rpid.
Codec Preferences:	Enter the codec preferences as a list. Ex: PCMA,PCMU,G722,OPUS
Extension In Contact:	Option to set the Extension In Contact.
Ping:	If your server is behind NAT then the ping option can be used to keep the connection alive through the firewall. The ping interval is in seconds.
Domain:	If the gateway will be used on a specific domain or global to all tenants.

NOTE:

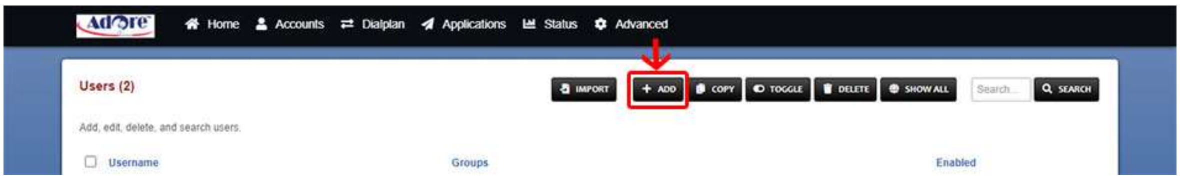
To see which Gateway a call is using. Advanced > Command and in the switch command section type show channels as xml and then press the execute button. In the output that is returned, look for the string sofia/gateway/ and the gateway name. This is the gateway your call is using.

Users

Here you can Define the users information to login to the web interface.



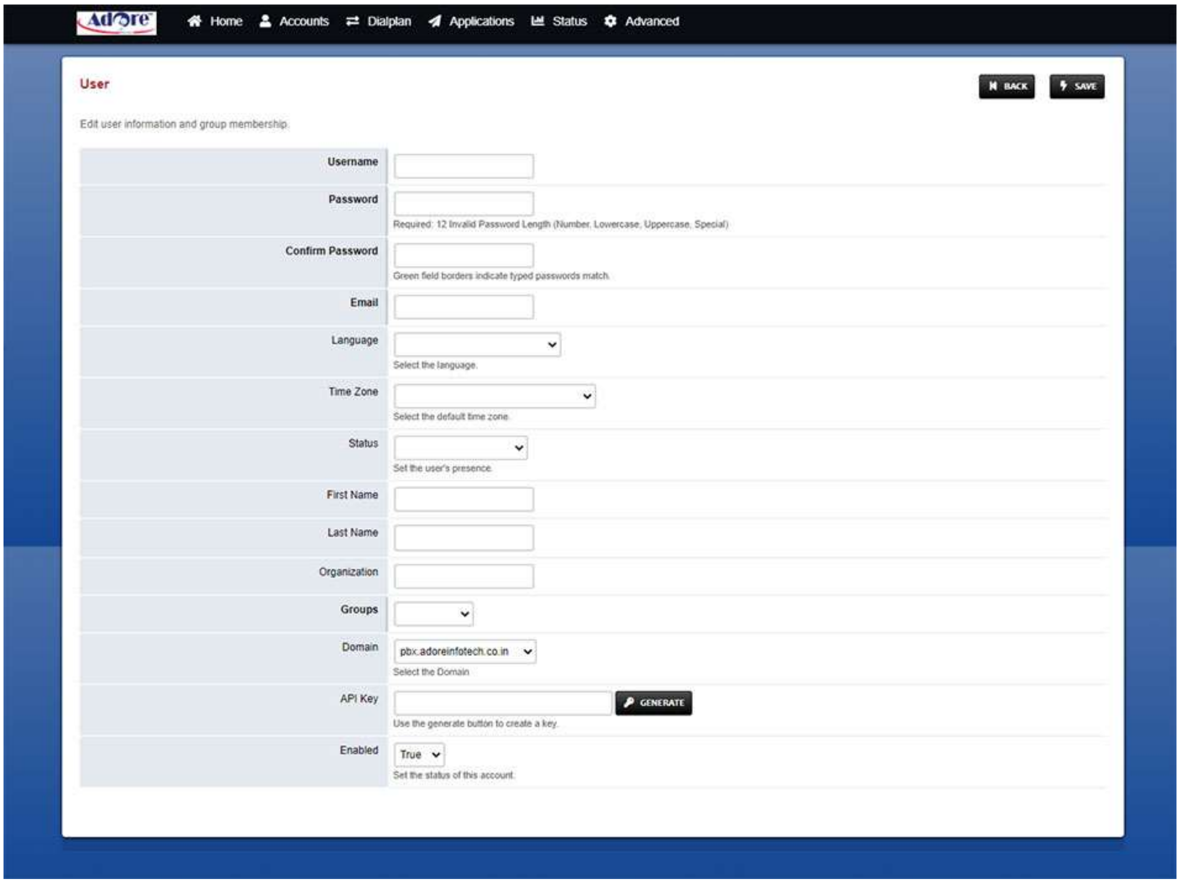
Click on **Add** button to add user in to the system



Please fill required field to add user for system

Username	Enter User id to be used to login.
Password	Enter Secret password used to login.
Confirm Password	Enter Same Password
Email	Enter email as per your wish
Language	Select language as per your wish, **NOTE: Per user language to override the domain or global language.**
Time Zone	Select Time Zone as per your wish, **NOTE : Per user time zone only needed if it needs to be different from the global time zone.**
Status	Set the user's presence as per your wish.
First Name	Enter Your First Name for new User.
Last Name	Enter Your Last Name for new user.
Organization	Enter your organization name
Group	Your can select group.
Domain	Select your Domain Name which user you are create

API Key	Generates an API Key
Enabled	You can enabled or disabled user from here



After create user you can also edit if you required.

6. Dialplan

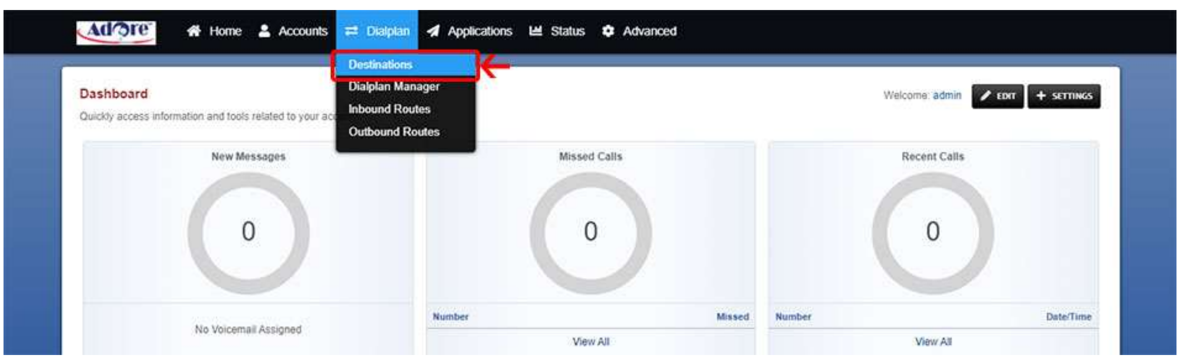
DialPlan

In the Dialplan menu you can set following options:

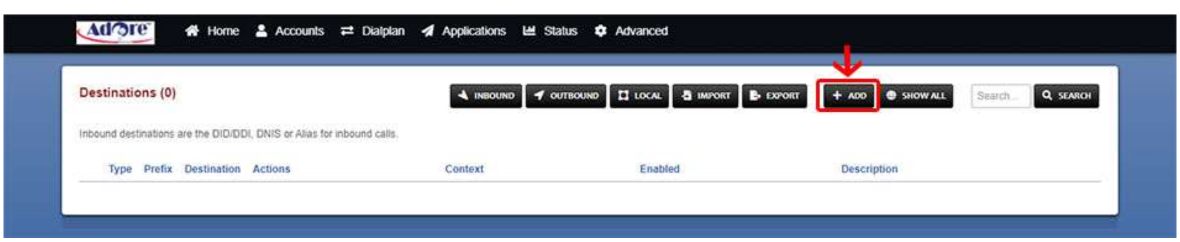
- **Destinations**
- **Dialplan Manager**
- **Inbound Routes**
- **Outbound Routes**

Destinations

Here you can add Destinations.

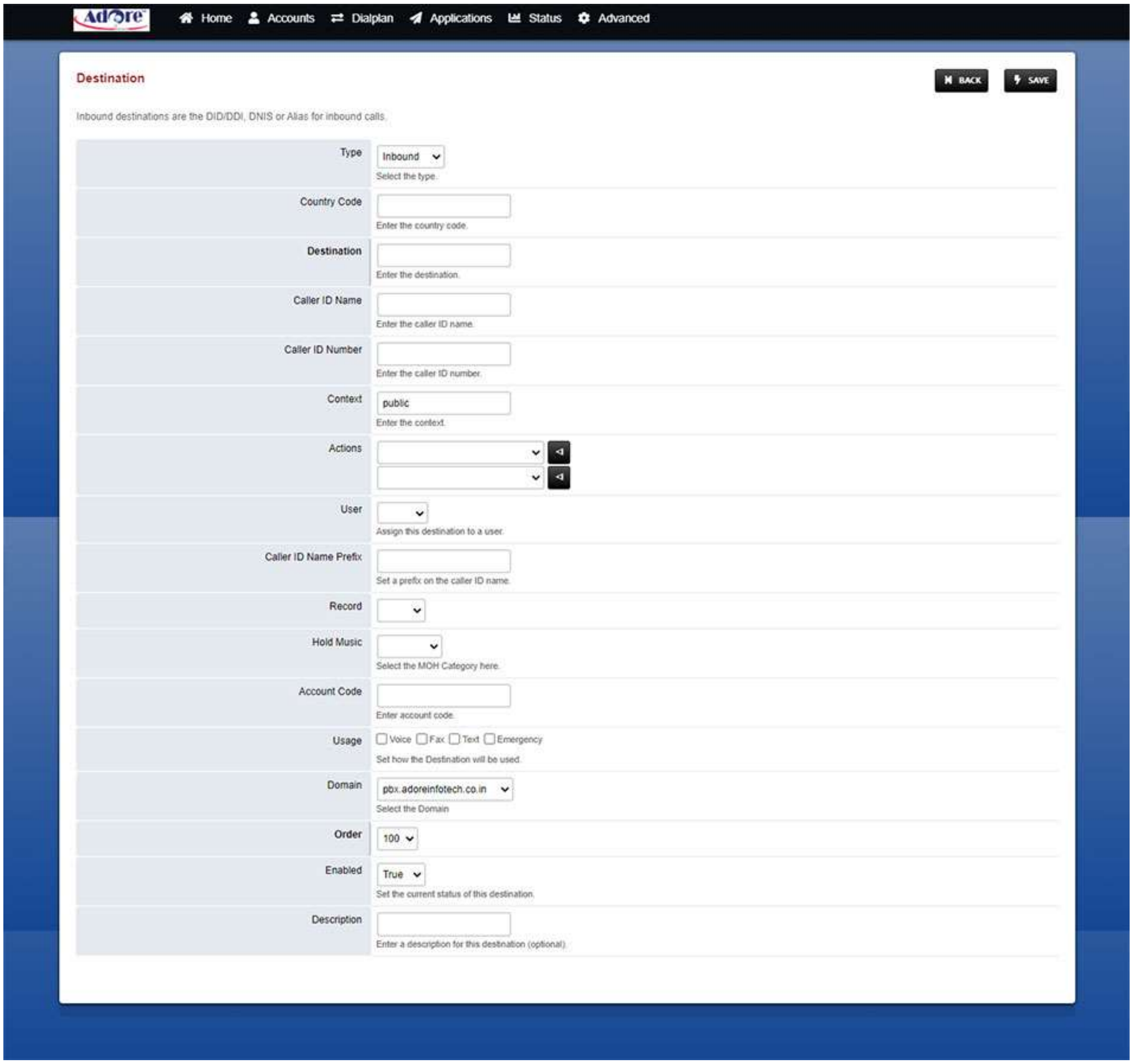


To add a destination click on the **Add** button.



On Click Add button following page will appear, Enter the route information below and Click Save once complete.

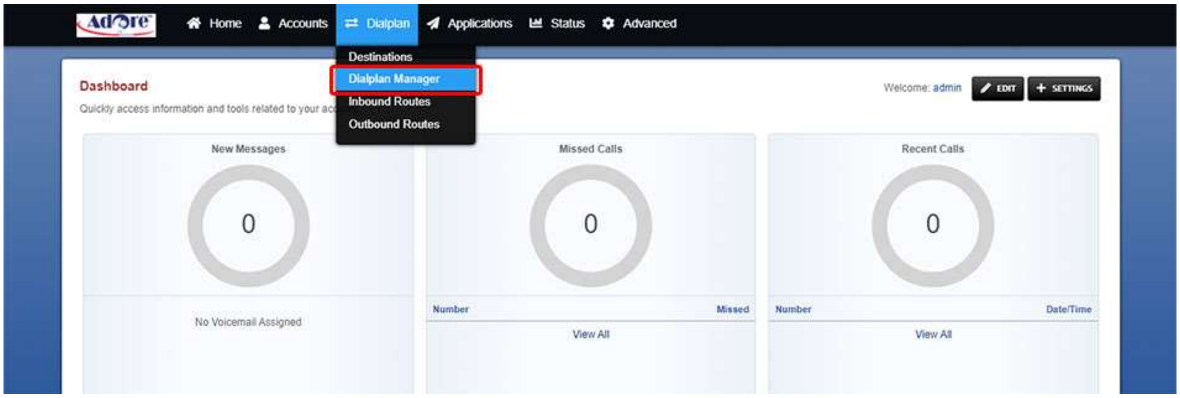
Type	Inbound or Outbound. Choose if this is an inbound destination or outbound destination.
Country Code	Enter your destination Country Code
Destination	This is usually the DID a caller will call.
Caller ID Name	Enter your Caller ID Name as per your wish
Caller ID Number	Enter your Caller ID Number as per your wish
Context	This will usually be public.
Actions	Choose where the call will go after it enters system.
User	Select User which you want to add call plan destination
Caller ID Name Prefix	Adds a name to the Caller ID that will display to the endpoint and call detail records.
Record	Record all calls made to the destination.
Hold Music	You can select Music On Hold
Account Code	Used in some billing systems
Usage	Select Always Voice
Domain	The domain can be global to all domains or domain specific
Order	As per your wish
Enabled	Enabled will enable the destination or Disabled to disable the destination.
Description	A way to label and organize what the destination is for.



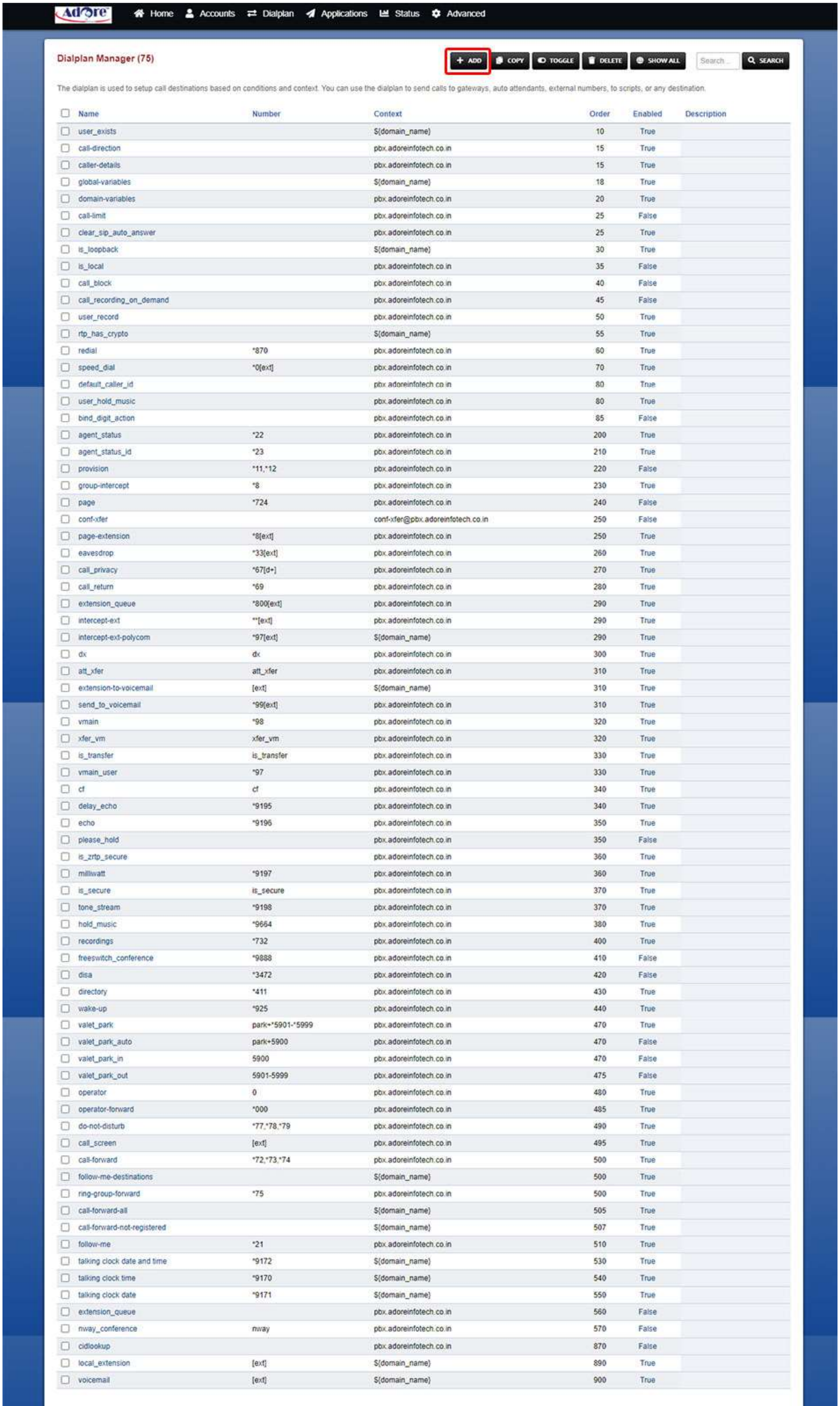
Dialplan Manager

The dialplan is used to setup call destinations based on conditions and context. You can use the dialplan to send calls to gateways, auto attendants, external numbers, to scripts, or any destination.

Click on **Dialplan** - **Dialplan Manager**



Dialplan Manager is internal setting so before any changes make sure you you have experience.



Dialplan Manager (75)

The dialplan is used to setup call destinations based on conditions and context. You can use the dialplan to send calls to gateways, auto attendants, external numbers, to scripts, or any destination.

Name	Number	Context	Order	Enabled	Description
user_exists		${domain_name}	10	True	
call-direction		pbx.adoreinfotech.co.in	15	True	
caller-details		pbx.adoreinfotech.co.in	15	True	
global-variables		${domain_name}	18	True	
domain-variables		pbx.adoreinfotech.co.in	20	True	
call-limit		pbx.adoreinfotech.co.in	25	False	
clear_sip_auto_answer		pbx.adoreinfotech.co.in	25	True	
is_loopback		${domain_name}	30	True	
is_local		pbx.adoreinfotech.co.in	35	False	
call_block		pbx.adoreinfotech.co.in	40	False	
call_recording_on_demand		pbx.adoreinfotech.co.in	45	False	
user_record		pbx.adoreinfotech.co.in	50	True	
rtp_has_crypto		${domain_name}	55	True	
redial	*870	pbx.adoreinfotech.co.in	60	True	
speed_dial	*0[ext]	pbx.adoreinfotech.co.in	70	True	
default_caller_id		pbx.adoreinfotech.co.in	80	True	
user_hold_music		pbx.adoreinfotech.co.in	80	True	
bind_digit_action		pbx.adoreinfotech.co.in	85	False	
agent_status	*22	pbx.adoreinfotech.co.in	200	True	
agent_status_id	*23	pbx.adoreinfotech.co.in	210	True	
provision	*11,*12	pbx.adoreinfotech.co.in	220	False	
group-intercept	*8	pbx.adoreinfotech.co.in	230	True	
page	*724	pbx.adoreinfotech.co.in	240	False	
conf-xfer		conf-xfer@pbx.adoreinfotech.co.in	250	False	
page-extension	*8[ext]	pbx.adoreinfotech.co.in	250	True	
eavesdrop	*33[ext]	pbx.adoreinfotech.co.in	260	True	
call_privacy	*67[d+]	pbx.adoreinfotech.co.in	270	True	
call_return	*69	pbx.adoreinfotech.co.in	280	True	
extension_queue	*800[ext]	pbx.adoreinfotech.co.in	290	True	
intercept-ext	**[ext]	pbx.adoreinfotech.co.in	290	True	
intercept-ext-polycom	*97[ext]	${domain_name}	290	True	
dx	dx	pbx.adoreinfotech.co.in	300	True	
att_xfer	att_xfer	pbx.adoreinfotech.co.in	310	True	
extension-to-voicemail	[ext]	${domain_name}	310	True	
send_to_voicemail	*99[ext]	pbx.adoreinfotech.co.in	310	True	
vmain	*98	pbx.adoreinfotech.co.in	320	True	
xfer_vm	xfer_vm	pbx.adoreinfotech.co.in	320	True	
is_transfer	is_transfer	pbx.adoreinfotech.co.in	330	True	
vmain_user	*97	pbx.adoreinfotech.co.in	330	True	
cf	cf	pbx.adoreinfotech.co.in	340	True	
delay_echo	*9195	pbx.adoreinfotech.co.in	340	True	
echo	*9196	pbx.adoreinfotech.co.in	350	True	
please_hold		pbx.adoreinfotech.co.in	350	False	
is_zrtp_secure		pbx.adoreinfotech.co.in	360	True	
milliwatt	*9197	pbx.adoreinfotech.co.in	360	True	
is_secure	is_secure	pbx.adoreinfotech.co.in	370	True	
tone_stream	*9198	pbx.adoreinfotech.co.in	370	True	
hold_music	*9664	pbx.adoreinfotech.co.in	380	True	
recordings	*732	pbx.adoreinfotech.co.in	400	True	
freeswitch_conference	*9888	pbx.adoreinfotech.co.in	410	False	
disa	*3472	pbx.adoreinfotech.co.in	420	False	
directory	*411	pbx.adoreinfotech.co.in	430	True	
wake-up	*925	pbx.adoreinfotech.co.in	440	True	
valet_park	park+*5901-*5999	pbx.adoreinfotech.co.in	470	True	
valet_park_auto	park+5900	pbx.adoreinfotech.co.in	470	False	
valet_park_in	5900	pbx.adoreinfotech.co.in	470	False	
valet_park_out	5901-5999	pbx.adoreinfotech.co.in	475	False	
operator	0	pbx.adoreinfotech.co.in	480	True	
operator-forward	*000	pbx.adoreinfotech.co.in	485	True	
do-not-disturb	*77,*78,*79	pbx.adoreinfotech.co.in	490	True	
call_screen	[ext]	pbx.adoreinfotech.co.in	495	True	
call-forward	*72,*73,*74	pbx.adoreinfotech.co.in	500	True	
follow-me-destinations		${domain_name}	500	True	
ring-group-forward	*75	pbx.adoreinfotech.co.in	500	True	
call-forward-all		${domain_name}	505	True	
call-forward-not-registered		${domain_name}	507	True	
follow-me	*21	pbx.adoreinfotech.co.in	510	True	
talking clock date and time	*9172	${domain_name}	530	True	
talking clock time	*9170	${domain_name}	540	True	
talking clock date	*9171	${domain_name}	550	True	
extension_queue		pbx.adoreinfotech.co.in	560	False	
nway_conference	nway	pbx.adoreinfotech.co.in	570	False	
cidlookup		pbx.adoreinfotech.co.in	870	False	
local_extension	[ext]	${domain_name}	890	True	
voicemail	[ext]	${domain_name}	900	True	

Dialplan Name	Dialplan Number
caller-details	
• *Details about the caller.*	
not-found:	
• *Used to help trigger fail2ban from bogus calls.*	
call-limit:	
• *Limit calls based on number of calls and more.*	
speed_dial:	*0[ext]
• *Uses LUA for extension speed dial.*	
agent_status:	*22
• *Agent login to call center.*	
page-extension:	*8[ext]
• *Password protected paging of an extension.*	
eavesdrop:	*33[ext]

• *Password protected evesdropping on extensions.*	
send_to_voicemail:	*99[ext]
• *Sending an active call to an extensions voicemail.*	
cf:	cf
echo:	*9196
• *Real time echo test.*	
milliwatt:	*9197
• *Plays a milliwatt test tone.*	
recordings:	*732
• *Password protected way to record audio that can be used in other applications like IVR.*	
directory:	*411
• *Directory of users.*	
user_exists:	
• *Determines if a user exists on the switch.*	
caller-details:	
• *Logic to decifer caller details.*	
call-direction:	
• *Determines the direction of the call.*	

variables:	
• *Set variables on a domain level.*	
is_local:	
• *Can be used to evaluate calls as local.*	
call_block:	
• *Block calls from reaching endpoints.*	
user_record:	
• *Used to record calls.*	
redial:	*870
• *Dial the last number that was dialed.*	
default_caller_id:	
• *Caller ID that can be set per domain.*	
agent_status_id:	*23
• *Status of the agent.*	
provision:	*11,*12
• *Used with devices.*	
clear_sip_auto_answer:	
nway_conference	nway
cidlookup:	
group-intercept:	*8
• *Intercepts a call from a defined group.*	
page:	*724
• *Password protected paging defined set of extensions.*	
conf-xfer:	

call_privacy:	*67[d+]
• *Send a privacy header to the carrier to hide caller id.*	
call_return:	*69
• *Call the last number that called the endpoint.*	
extension_queue:	*800[ext]
intercept-ext:	**[ext]
• *Password protected intercept of an extension.*	
dx:	dx
• *Direct transfer.*	
att_xfer:	att_xfer
• *Attended transfer.*	
extension-to-voicemail:	[ext]
• *Used for extension to voicemail.*	
vmain	*98
• *Main menu to access any voicemail using a pin number.*	
xfer_vm	xfer_vm
• *Transfer to voicemail.*	
is_transfer	is_transfer
• *Used for call transfering.*	
vmain_user	*97
• *Endpoint's voicemail using a pin number.*	
delay_echo	*9195
• *Play back an echo with a 5 second delay.*	
please_hold	

• *Plays an audio file when on hold.*	
is_zrtp_secure	
is_secure	is_secure
tone_stream	*9198
• *tones that stream and sound like Tetris music.*	
hold_music	*9664
• *Play music on hold. Good for testing on an endpoint.*	
freeswitch_conference	*9888
• *An easy way to join the Cluecon Weekly call.*	
disa	*3472
• *Call in to a phone number and provide a pin to dial out.*	
wake-up	*925
• *Schedule date and time for an automated call.*	
extension_queue	
valet_park	park+*5901-*5999
• *Default range to valet park calls.*	
valet_park_in	park+*5900
• *Default number to send valet calls to.*	
valet_park_out	park+*5901-*5999
• *Default range to retreive valet parked calls.*	
operator	0
• *Configurable option for an operator.*	
operator-forward	*000
• *Uses dial_string.lua.*	

do-not-disturb	*77,*78,*79
• *Turn on, toggle, turn off do not disturb.*	
call-forward	*72,*73,*74
• *Turn on, toggle on/off and turn off call forwarding.*	
follow-me	*21
• *Forwards call to defined list of phone numbers or extensions.*	
bind_digit_action	
call_screen	[ext]
• *Play an audio file and give options to the caller to record a short message for the call recipient. Call recipient can then accept or reject the call.*	
local_extension	[ext]
• *Examines to see if the extension is local.*	
voicemail	[ext]
• *Voicemail for extensions.*	

Inbound Routes

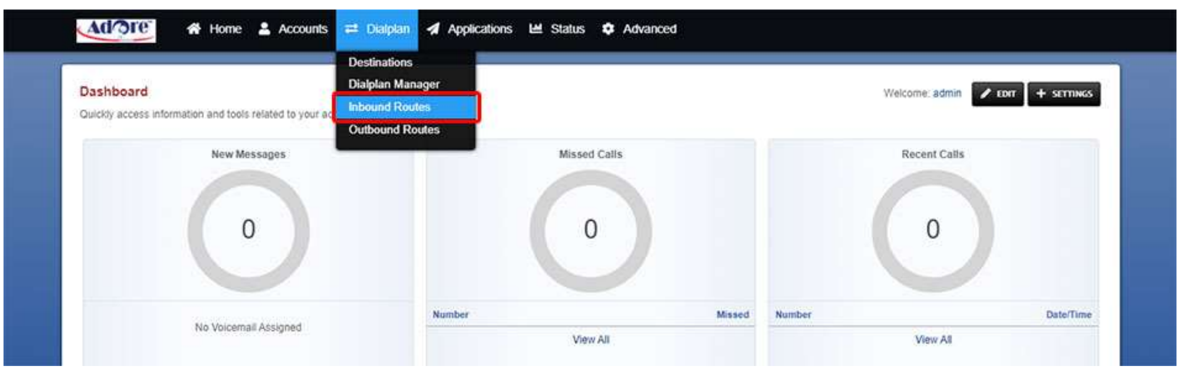
Route incoming calls to destinations based on one or more conditions. It can send incoming calls to:

- IVR Menu
- Call Group
- Extension
- External Number
- Script

Directs public inbound calls to an internal destination on the system. Note that the only difference between the inbound route dial plan and the normal dial plan is that the inbound route dial plan works on all calls that are in the public context whereas the normal dial plan works on the domain context.

Inbound Call Routing is used to route incoming calls to destinations based on one or more conditions and context. It can send incoming calls to an auto attendant, huntgroup, extension, external number, or a script. Order is important when an anti-action is used or when there are multiple conditions that match.

Inbound routes can be used for advanced reasons. Dialplan > Destinations will create and configure the Inbound Route for you.



Click Add button to **add** Inbound Routing

Name: The name of the Inbound Route.

Number: The Number (DID) an outside caller will call.

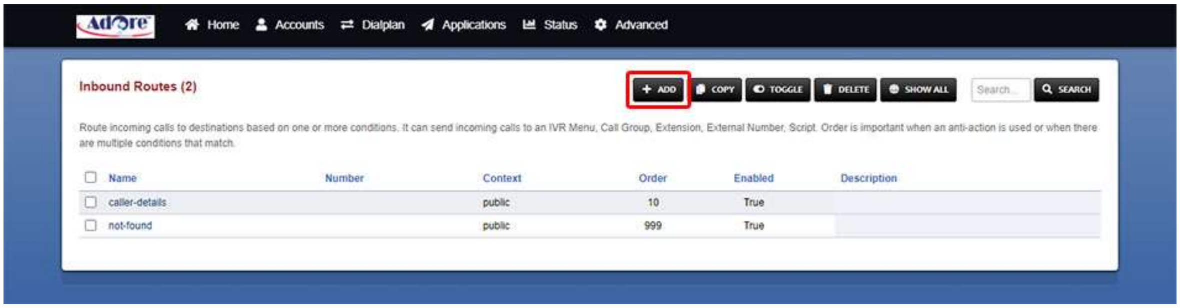
Context: Context of the Inbound Route. Usually will be public.

Hostname: Usually blank, otherwise for advanced use.

Order: Order where the inbound route will be used in the dialplan.

Enabled: If the Inbound Route is enabled or disabled.

Description: A way to organize what the inbound route is used for.



Edit / Add Inbound Rating

Name: The name of the Inbound Route.

Number: The Destination Number (DID) an outside caller will call.

Context: Context of the Inbound Route. Usually will be public.

Order: Order where the inbound route will be used in the dialplan.

Domain: Can be global to all domains or specific to one domain.

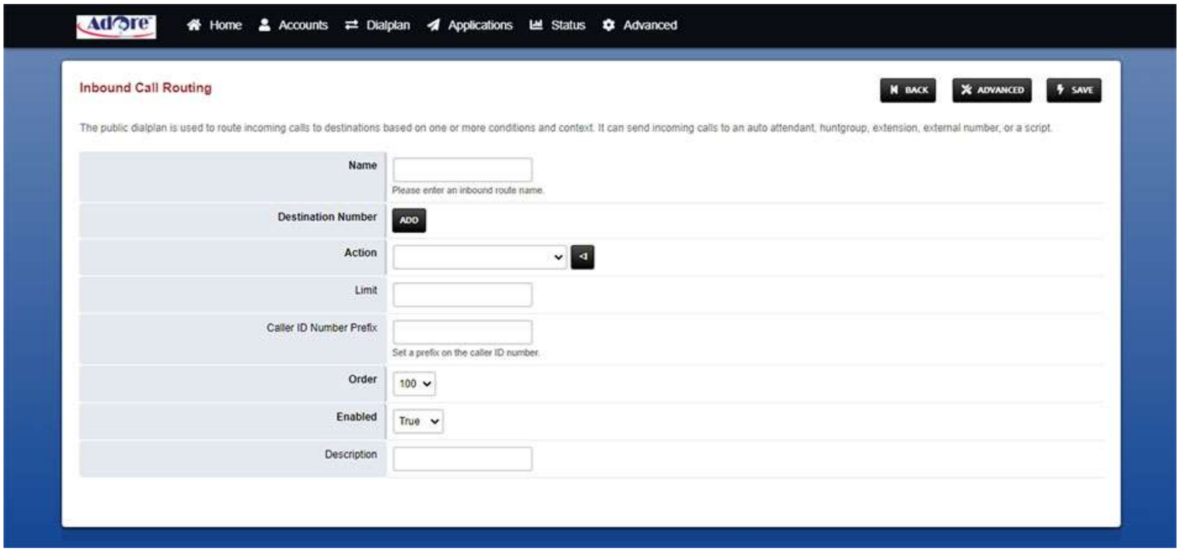
Continue: If you want the call to continue through the order of the remaining dialplans. This is usually set as false.

Enabled: If the Inbound Route is enabled or disabled.

Description: A way to organize what the inbound route is used for.

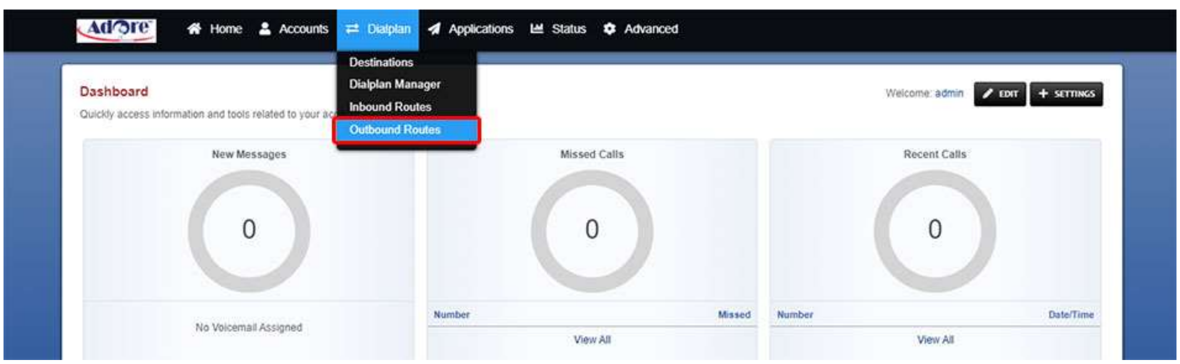
Destination Number

Add button allows creating an inbound route. The list of destinations for the Destination Number select list is populated by the list from the Destinations tool. This list can be found by navigating to Menu -> Dialplan -> Destinations. Note: It is recommended to use Destinations tool and select where to route the call and have it build the inbound routes. There are many benefits to the destination select tool. Manually creating an inbound route is not recommended unless you need an advanced inbound route.



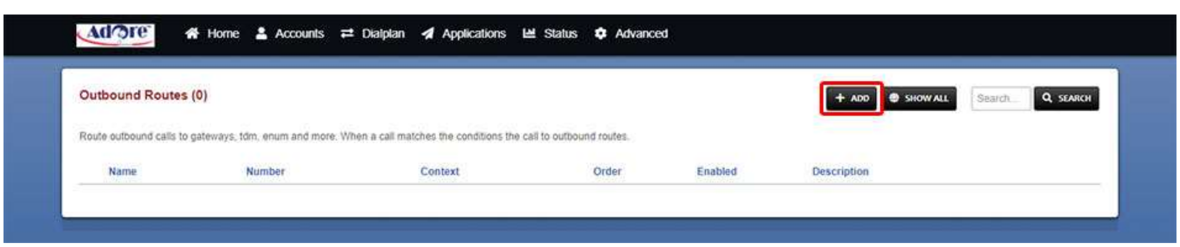
Outbound Routes

Route outbound calls to gateways, tdm, enum and more. When a call matches the conditions the call to outbound routes.

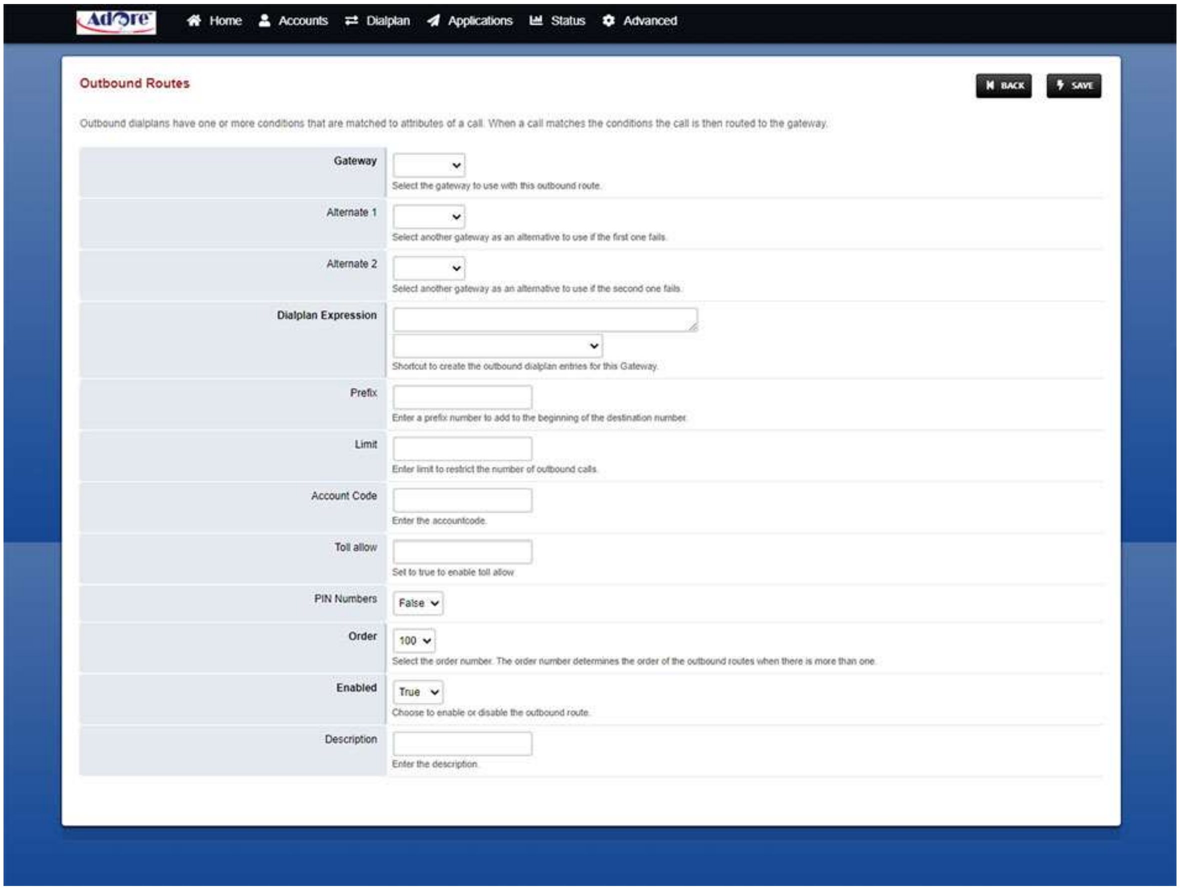


Configuring an Outbound Route.

- Select Dialplan from the drop-down list and then click Outbound Routes .
- Click the plus button on the right. Enter the route information below and Click Save once entry is complete.



Outbound dialplans have one or more conditions that are matched to attributes of a call. When a call matches the conditions the call is then routed to the gateway.



7. Applications

In the **Applications** menu (Apps) section you will get following options:

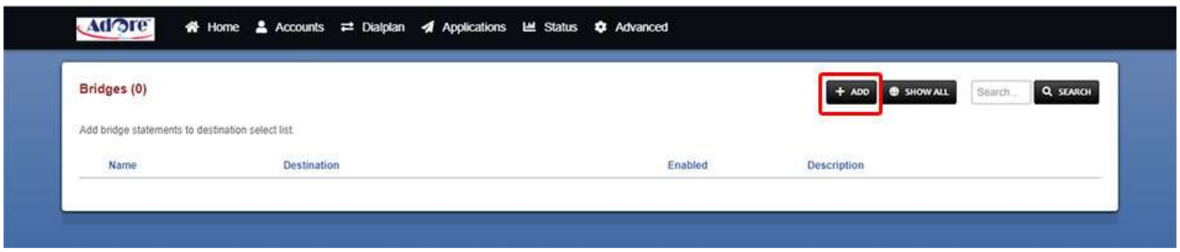
- **Bridges**
- **Call Block**
- **Call Broadcast**
- **Call Center**
- **Call Detail Records**
- **Call Flows**
- **Conference Center**
- **Conference Controls**
- **Conference Profiles**
- **Contacts, Fax Server**
- **Follow Me**
- **GS Wave**
- **IVR Menu**
- **Music on Hold**
- **Operator Panel**
- **Phrases**
- **Queues**
- **Recordings**
- **Ring Groups**
- **Streams**
- **Time Conditions**
- **Voicemail**

Bridges

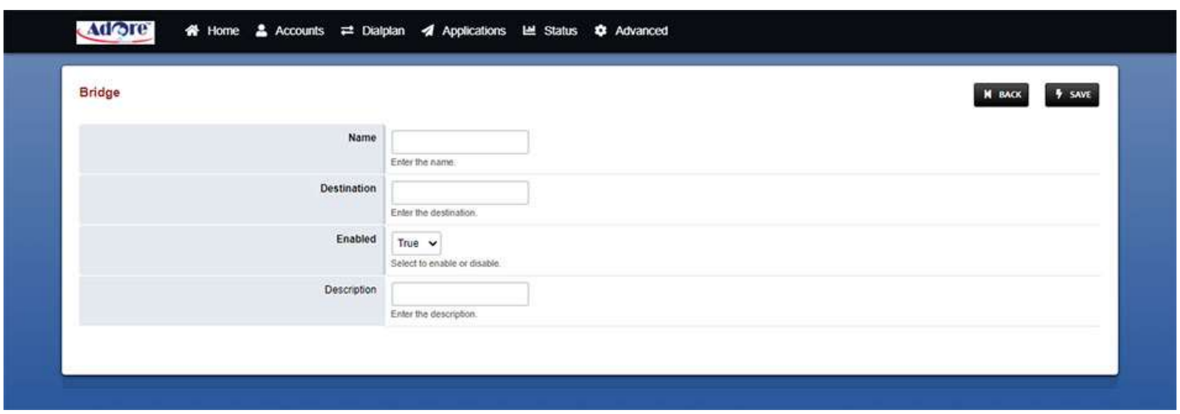
Bridge are used to send calls directly to other destinations like another PBX, Carrier or External SIP to TDM Gateway and more. The bridge are added to destination select list.

Go to **Applications** -> **Bridges**

Click the **Add** icon to add a bridge



Fill all the details and save to add bridge



Call Block

A list of numbers from which to block calls.

Go to **Applications** -> **Call Block**

To block a call click on the **Add** icon

Fill out the fields with pertinent information

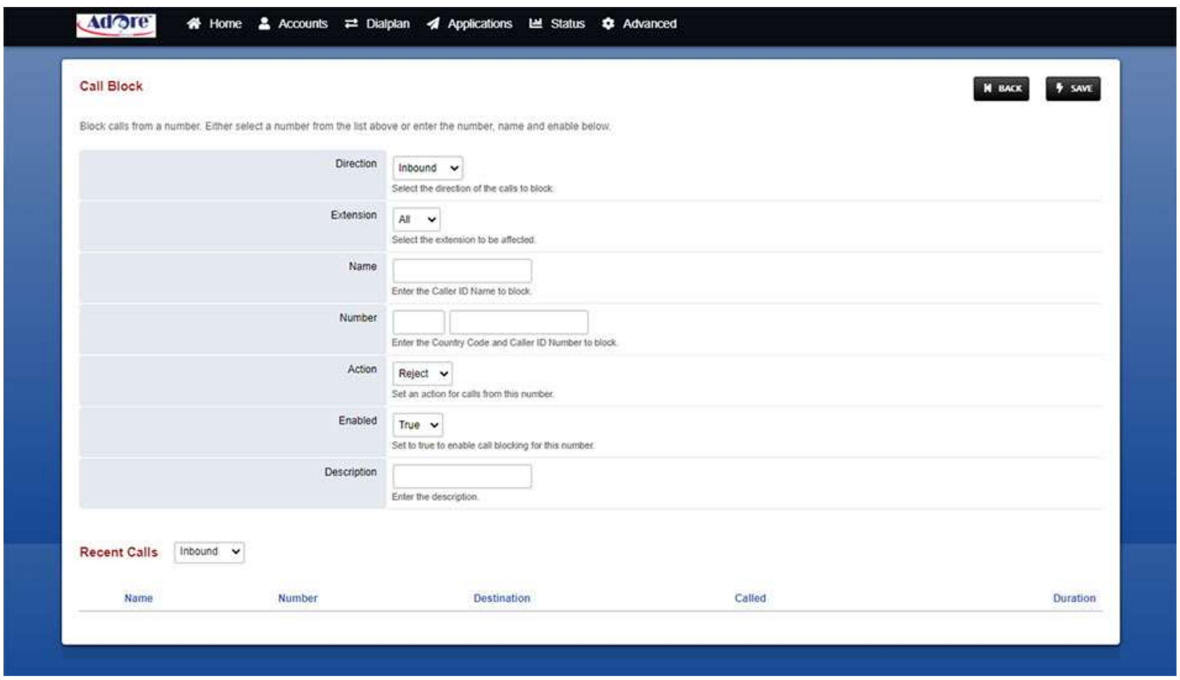
Action:

Reject- Will reject the call

Busy- Will send a busy signal

Hold- Will put the call on hold

Voicemail- Will send the call to the specified voicemail box

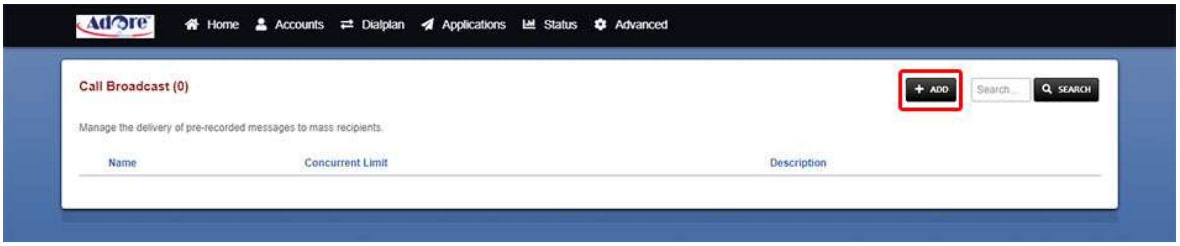


Call Broadcast

Go to **Applications - Call Broadcast**

Broadcast calls (a light dialer) to a defined list of phone numbers.

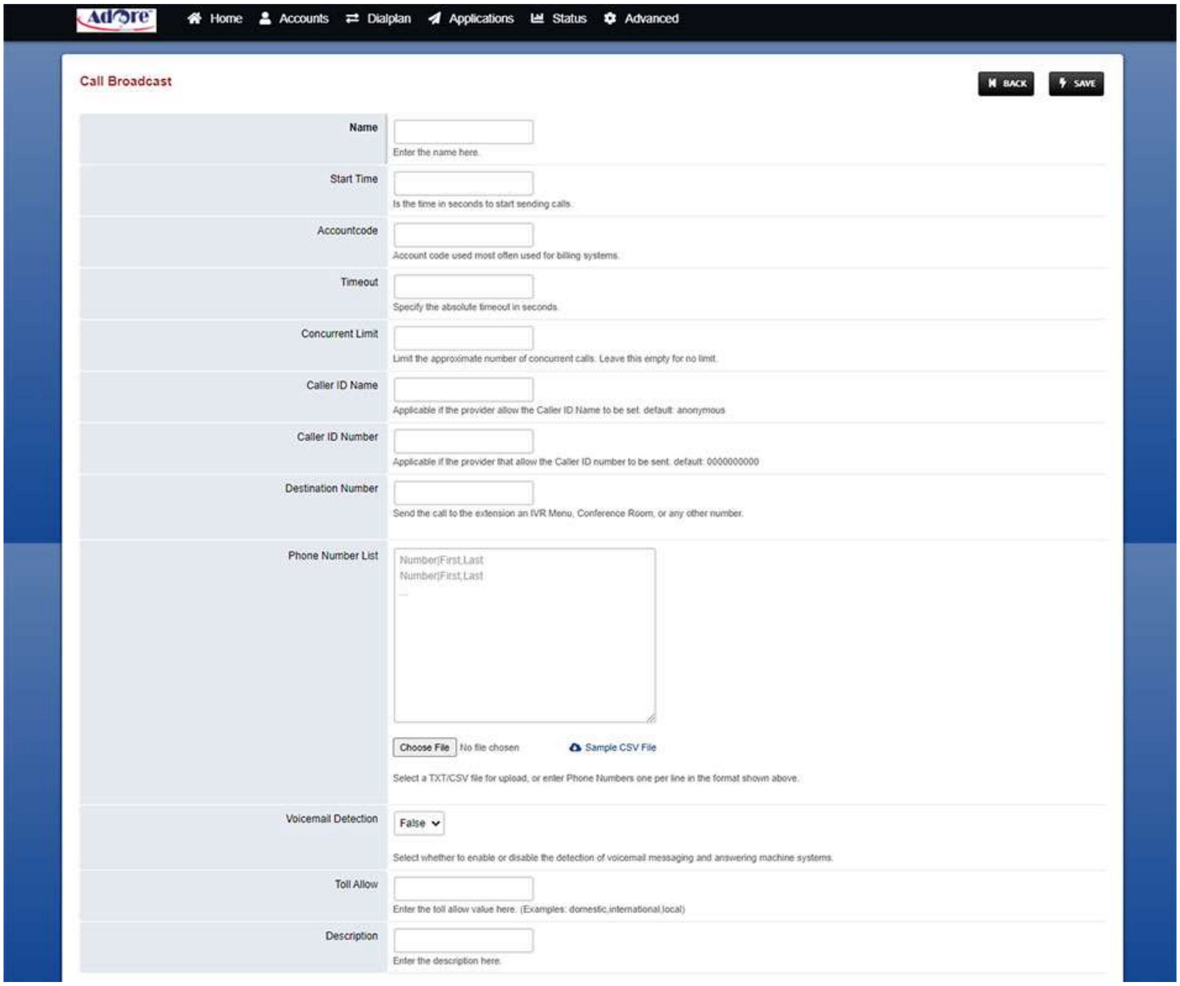
To create **Call Broadcast** click **Add** button



Fill in the following fields

Name	Name for the Call Broadcast.
Start Time	The time in seconds to start sending calls
Accountcode	Accountcode Used by some billing systems.
TimeOut	Amount of time till hangup.
Concurrent Limit	Used to pace the calls calls if the timeout was 60 and the concurrent. (limit is 100 then we would schedule 100 calls every 60 seconds.)
Caller ID Name	Name that will be used on outbound caller id.
Caller ID Number	Number that will be used on outbound caller id.
Destination Number	This is the internal number to call. Send the call to an IVR Menu or some other number. If sending to a conference room make sure the room has a pin number or something that requires user input you don't want to add voiemail messages into the conference room.
Phone Number List	Phone Number List- List of phone numbers to call in the call broadcast. This is the external number to call. Set a list of phone numbers one per row in the following format: 123-123-1234|Last Name, First Name 5551231234|example1 5551231234|example2 5551231234|example3
Voicemail Detection	Set True or false to detect an answering machine.
Toll Allow	Enter the toll allow value here. (Examples: domestic, international, local)

Description	Description Help organize and label what the call broadcast is for.



Once you have everything filled out click the **Save** button, Call Broadcast name you just created. On the top right click the Send Broadcast button to start the call broadcast. To stop the call broadcast click **STOP BROADCAST** on the top right.

Call Centers

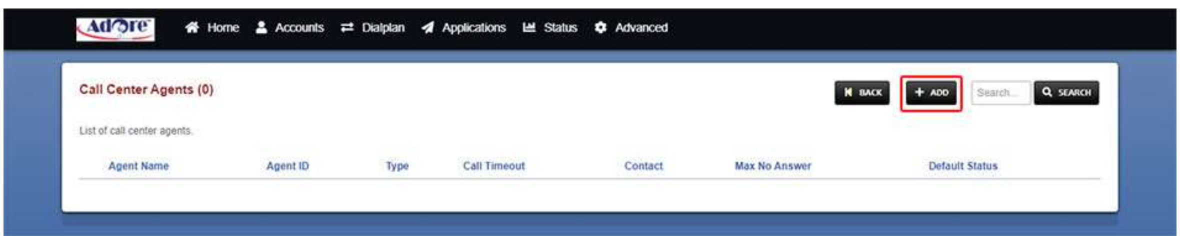
List of queues for the call center.

Go to **Application** -> **Call Centers**

Call Center Agents

From **Application** -> **Call Center** click **Agents** at the top to access Call Center Agents

Click the **Add** button on the top right to add agents (make sure to set Agent ID!)

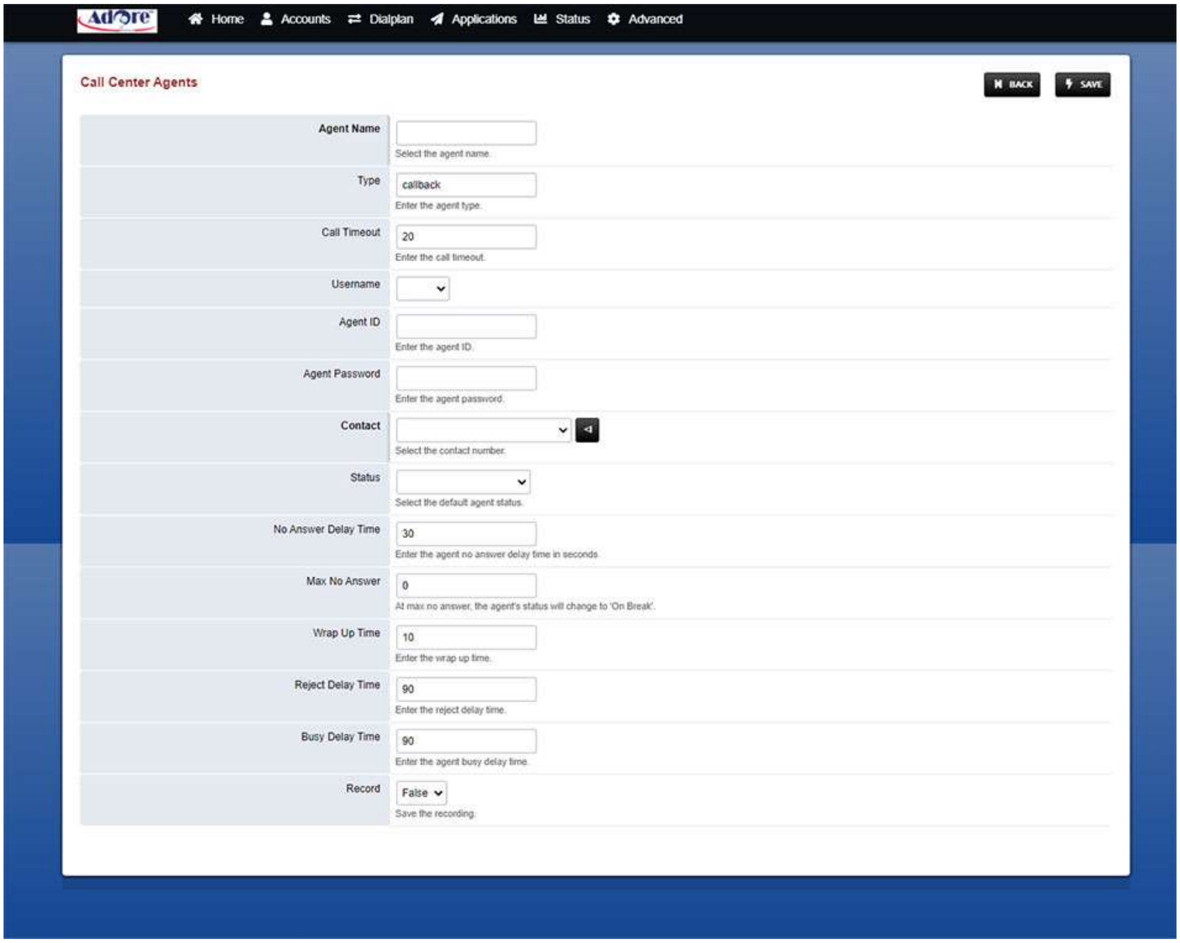


Set the Agent Password, or add agent_authorized=true to the dialplan for *22 if you do not want to require a PIN to log in

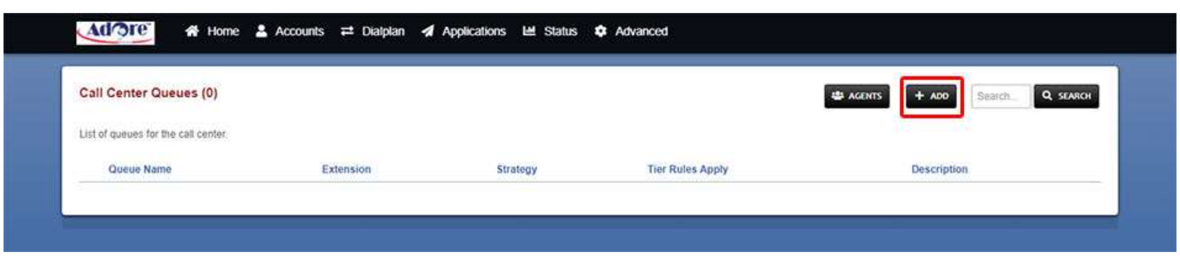
If you want to enable Follow Me or Call Forwarding for an Agent, set the contact string to loopback/<extension>1

Agent Name	Name of the agent. When adding agents to the Call Center, this is what you will see to describe the agent
Type	2 types supported, callback and uuid-standby. callback will try to reach the agent via the contact fields value. uuid-standby will try to directly bridge the call using the agent uuid

Call Timeout	Time to ring the agent before deeming them unavailable
Username	Assign a system user with this call center agent
Agent ID	An ID that can be used to log the agent in and out of the call center
Agent Password	A password to log the agent into the call center. This is not used if you have added agent_authorized=true to the dialplan for *22
Contact	A dropdown to select which extension should be used to contact the agent
Status	The default status that the agent in the call center
No Answer Delay Time	The time the system will wait to attempt a call to the agent again if they did not answer within the Call Timeout
Max No Answer	Max attempts to call the agent. For example, when set to 1, if the agent does not answer within the first Call Timeout, they will not get another chance to answer the call. If set to 2, the agent will have two attempts to answer the call



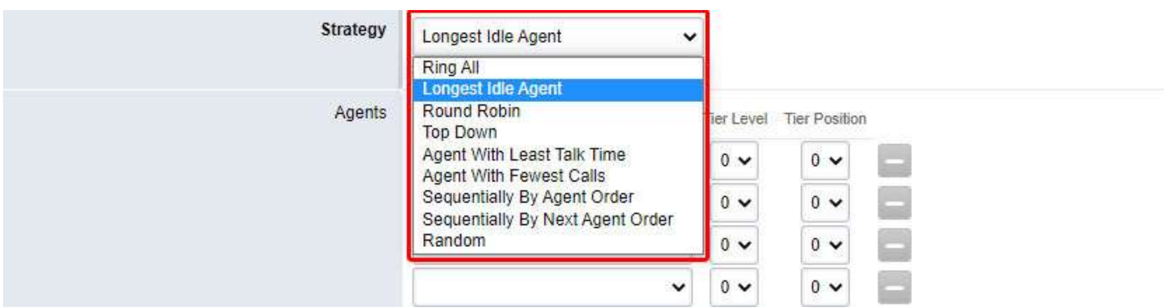
- To add a Call Center Queue click the **Add** button in the top right
- Once a Queue is created click the name of the queue to edit it. Once in edit mode, at the top right you can view, stop, start, restart and save the queue as well as make changes.



Fill the desired fields and click Save button to add call center queues

Call Center Strategies

- **Agent With Least Talk Time**: Rings the Agent will ring that has the least time talking.
- **Agent With Fewest Calls**: Agent will ring that has the least calls.
- **Longest Idle Agent**: The agent will ring who idles the longest depending on their tier level.
- **Ring All**: All agents ring simultaneously.
- **Random**: Rings Agents will ring randomly in not particular order.
- **Ring Progressively**: Agents will ring the same as top-down and will progress until each agent ends up ringing.
- **Round Robin**: Will ring the next agent available in line.
- **Sequentially By Agent Order**: Agents will ring in a sequence by the tier and the tiers order.
- **Top Down**: Agent rings in order starting from one.



Others setting option for Call Center Agents, Please read carefully and set as per you.

Agents

Select agents from the drop down list and specify tier level and tier position.

Music On Hold

Select the desired hold music. Music on hold, streams and ringtones can be used.

Record

Save the recording

Time base score

Queue: Caller in queue time will start. If the caller goes to another queue the time will start over.

System: Caller in queue will have their wait calculated as soon as they enter the system. If a caller chooses the wrong queue, when they get to the correct queue the timer won't start over again.

Time base score - Seconds

This field is left blank by default which means the option will not be added to the XML Dialplan. If you populate the field with a number then the time base score will be set in the dialplan when entering the call center. This can be used to prioritize one call center queue over another.

Example 1: You may have two calls come into the system. Caller 1 entered before Caller 2. Caller 2 however has entered the "**VIP**" call center queue. FS will deliver the call that has the longest "**time base score**" to the agent. By setting the "**time base score - seconds**" you can tell FS that Caller 2 has "**waited**" longer that Caller 1 even if it isn't true. This will allow the "**VIP**" queue to be answered first.

Example 2: Similar to the example above, you may want to prioritize one queue over another however you may want a threshold at which the two then become equal. For example, if Caller 1 is waiting for an agent more than 5 minutes, their call should be equal in priority to Caller 2. In this case, set the "time base score - seconds" of the "**VIP**" queue to be 300 (5 min). This will mean that the "**VIP**" queue will get only a 5min head start on the regular queue.

Max Wait Time

A value of 0 is the default and equals an infinate amount of time. Any other numeric value is calculated in seconds.

Max Wait Time with No Agent

Enter the max wait time with no agent. Multi Tenant PBX sets the default to 90 seconds and the Timeout Action will be used if there are no agents available.

Max Wait Time with No Agent Time Reached

Enter the max wait time with no agent. Multi Tenant PBX sets the default to 30 seconds and the Timeout Action will be used if there are no agents available.

Timeout Action

Set the action to perform when the max wait time is reached.

Tier Rules Apply

True: Set the tier rule rules apply to true. The defined tiers will be used.

False: Set the tier rule rules apply to false. All tiers will be used.

Tier Rule Wait Second

30 seconds is default. Enter the tier rule wait seconds.

Tier Rule Wait Multiply Level

True: The amount of seconds the caller waits until the next tier. This value will increase(multiply) if Tier Rule Wait Multiply Level is marked true.

False: Tier Rule Wait Multiply Level is marked false then after the set amount of seconds pass the tiers in order will execute with no wait.

Tier Rule No Agent No Wait

True: Setting is enabled.

False: Setting is disabled.

Discard Abandoned After

Default is 900 seconds. Sets the discard abandoned after seconds.

Abandoned Resume Allowed

True: Setting is enabled. Permits a call to resume their position in the queue but only in the amount of seconds set in discard abandoned after .

False: Setting is disabled.

Caller ID Name Prefix

Set a prefix on the caller ID name.

Announce Sound

A sound to play to a caller every announce sound seconds. Needs the full path to the .wav file.

Announce Frequency

How often the announce sound is played in seconds.

Exit Key

Keys to quit the current queue waiting.

Description

Enter a description to help organize and define what the queue is for.

Agent Call Center Login

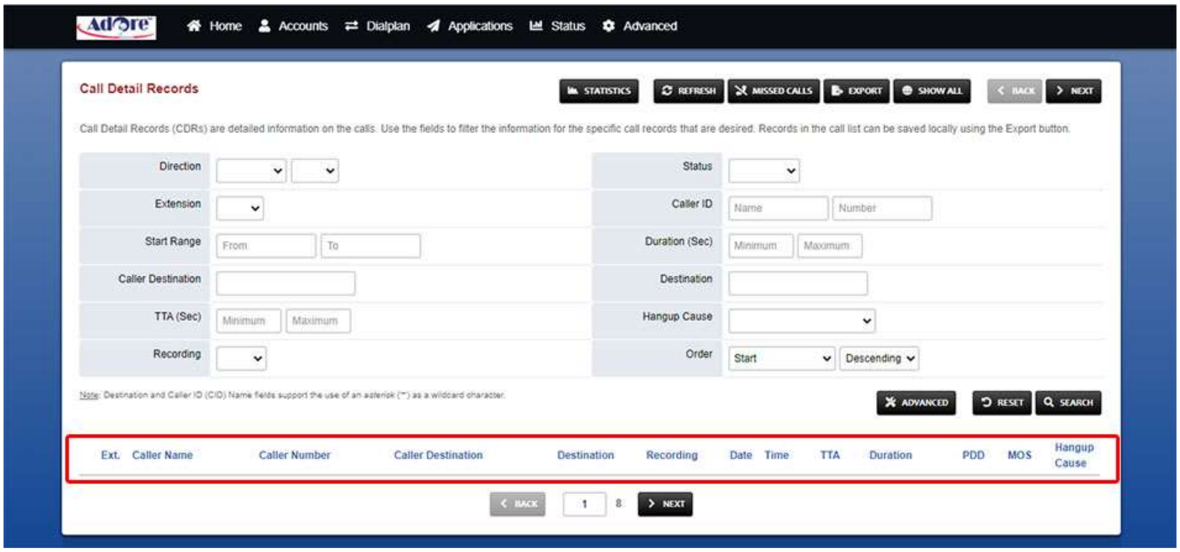
Agents can login to call center with *22 from the phone or via the Multi Tenant PBX web interface. Admin and Super Admin accounts can also log other agents in or out.

Login then go to **Status** > **Agent Status**

Call Details Records

Call Detail Records (CDRs) are detailed information on the calls. Use the fields to filter the information for the specific call records that are desired. Records in the call list can be saved locally using the Export button.

Go to **Applications** -> **Call Details Record**



CID Name - Caller ID Name

Source - Where the call came from

Destination - Where the call went to

Recording - A link will appear if the call recorded

Start Time - the call entered the system

TTA Time - To Answer the call

Duration - How long the call was

PDD - Post Dial Delay

MOS - Mean Opinion Score is a measure of voice call quality

Hangup Cause - Details about the entire calls. Usually will be “Normal Clearing”

Call Detail Records are detailed information on the calls. The information contains source, destination, duration, and other useful call details. Use the fields to filter the information for the specific call records that are desired. Then view the calls in the list or download them as comma seperated file by using the CSV button.

Note that this page makes use of XML CDR for reporting.

Post Dial Delay (PDD)

Post Dial Delay (PDD) is experienced by the sender as the time from the sending of the final dialed digit to the point at which the sender hears ring tone or other in-band information. In other words, the **PDD** would be the time from when the sender sends the **INVITE** to receiving the first ringing response.

That said, **PDD** does not take into account the time it takes the receiver to hear the call coming in due to the various factors on how they are setup for inbound calls. For example, call forwarding may affect the time it takes the receiver to know that someone is calling because of call forwarding. The sender might hear a ring tone almost instantly from the time it dials the final digit because they sent out an **INVITE**, but the receiver of the call might have setup inbound calls to be forwarded to their cell phone, in which now the call must travel through their phone system, to their phone system's gateway carrier to deliver the sender's call to the receiver's cell phone carrier network in order for the cell phone carrier to deliver the sender's call to the receiver's cell phone.

Recordings

Any calls which have the entry in the name column underlined (ie. the name is a link) have a recording available. Clicking on the name will playback the recording in a new window. In such cases the number entry will also be a link - clicking on this link will download the recording to your computer as a wav file.

Possible issues

No records showing up under Apps-Call Detail Records

Possible causes:

1. The module is disabled

Make sure the XML CDR module is enabled and running in the **Menu** -> **Advanced** -> **Modules**.

2. Wrong xml_cdr.conf.xml config

check <param name="url" value="http://127.0.0.1/app/xml_cdr/v_xml_cdr_import.php"/> and adapt it to your situation.
Compare your version (advanced-script editor-files-autoload_configs-xml_cdr.conf.xml) with the current default one that is included in Multi Tenant PBX (advanced-php editor-files-includes-templates-conf-autoload_configs-xml_cdr.conf.xml). If it is different copy the default one over yours.
Then edit the line <param name="url" value="http://{v_domain}/mod/xml_cdr/v_xml_cdr_import.php"/> and replace {v_domain} with the domain or IP address of your Multi Tenant PBX server.
Then edit the line <param name="cred" value="{v_user}:{v_pass}"/> and replace {v_user} with a complex name of upper and lowercase and numeric characters so it is really ugly and secure, and do the same for v_pass.
Make each of them completely unique.

Be aware that these don't have to match anything else on your server at all. This is because Multi Tenant PBX does something very simple but clever here. The xml_cdr module uses this account when it does an http post to Multi Tenant PBX of the new data. Multi Tenant PBX looks at the same xml_cdr.conf.xml file that the module uses in order to check if the module is using a valid account and password. Since they both look at the same config file they are using the same account and password and will happily talk to each other!

Once you've made these changes you can save the file. You could restart your server, or you could reloadxml and then restart the xml_cdr module. Either is ok, it is up to you. Then your changes will have taken effect and you should no longer lose your menu bar when looking at CDR information.

XML CDR configuration

For more detailed configuration go to the XML editor (Menu -> Advanced -> XML Editor) and in autoload configs look at xml_cdr.conf.xml

Note:

By default only the a-leg of the call is logged therefore if you make a recording of the b-leg you won't be able to retrieve it using the Call Detail Records. If you want the b-leg as well you need to change log-b-leg=true in this config.

Harddrive space usage

Note :

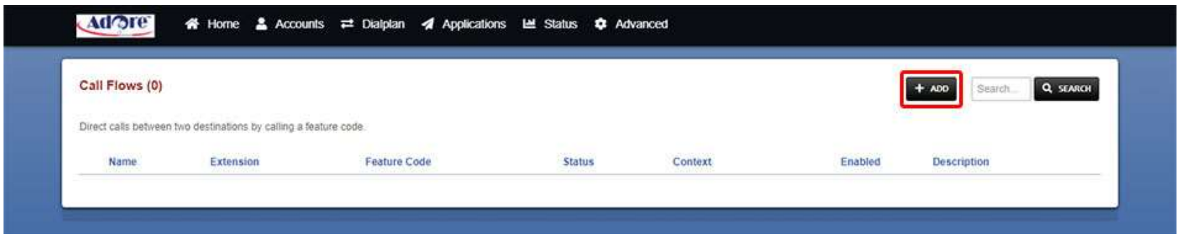
XML CDR data adds up fast, therefore you may need to clear this data at some point in the future. By default freeswitch keeps this in (source install) /usr/local/freeswitch/log/cdr-csv or (package install) /var/log/freeswitch/xml_cdr and inside that by year, month and day. Recordings also take up space and have to be manually deleted if you want the space back these are kept in (source install) /usr/local/freeswitch/recordings/{Domian_Name} or (package install) /etc/freeswitch/recordings/{Domain Name} and inside that by year, month and day.

Call Flows

Direct calls between two destinations by calling a feature code.

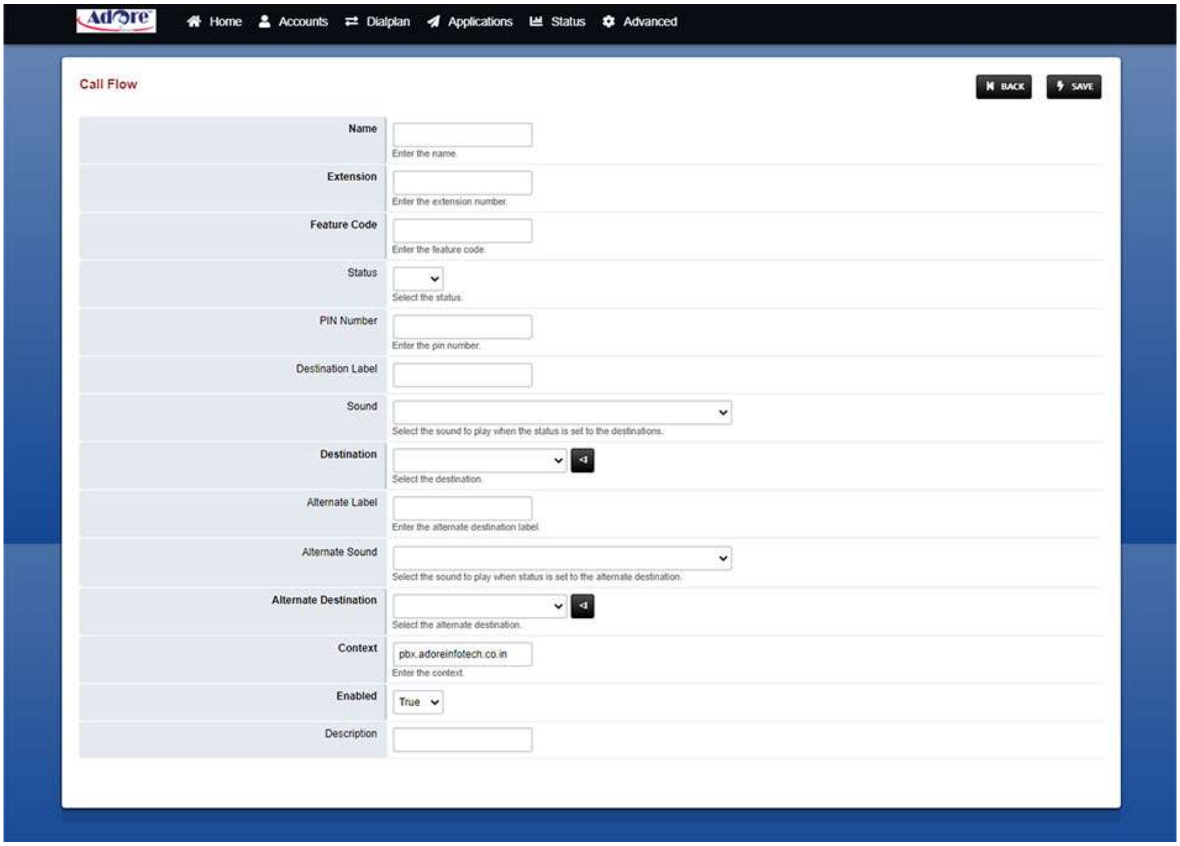
Go to **Applications** -> **Call Flows**

Click on **Add** button to add Call Flows in system



Name	Define the name of the call flow.
Extension	Define what extension to use. (This will make an extension not already created)
Feature Code	Define what * number to use
Status	Define what currently is in use.
Pin Number	Define a pin number in order to execute either mode.
Destination Label	Define where the call will go in the initial mode.
Sound	Define the sound that will play once mode is engaged
Destination	Define what the destination will be.
Alternate Label	Label that will show when alternative mode is in use.
Alternate Sound	Define the sound that will play once alternative mode is engaged.
Alternate Destination	Define where the call will go in the altenative mode.
Context	Domain context (typically leave as is)

Enabled	Select TRUE or FALSE
Description	Label what this call flow does.



Call Flow Example

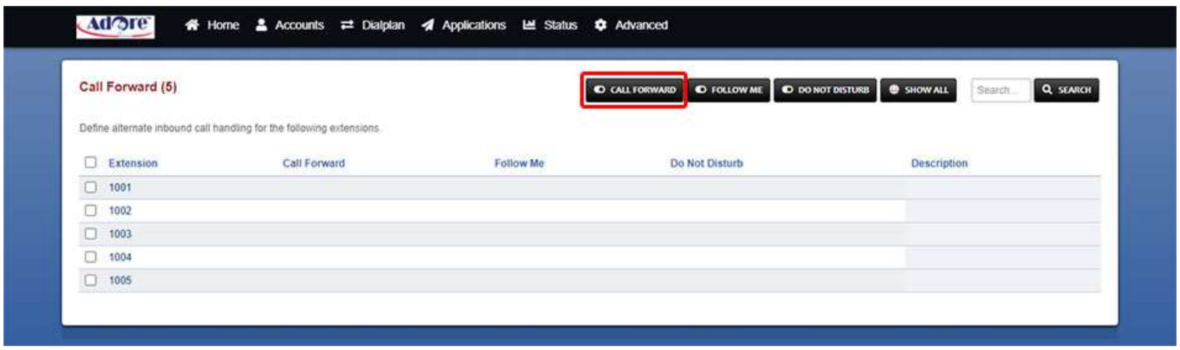
In the Call Flow example below we have the name as Call Flow. Make the Extension number 30 that didn't exist until now. Create the feature code as a *code with *30. Keep the context as-is with **pbx.adoreinfotech.co.in** . Select a Status to show which mode. Make a PIN to help secure the call flow. Make the destination label as Day Mode. Select a sound to auditorially indicate which mode is activated. Choose a destination for the alternative mode. Make the alternative detination label as Night Mode. Select an alternative sound to auditorially indicate which mode is activated. Choose a destination for the alternative mode. Finally, enter a description to describe what this call flow does.

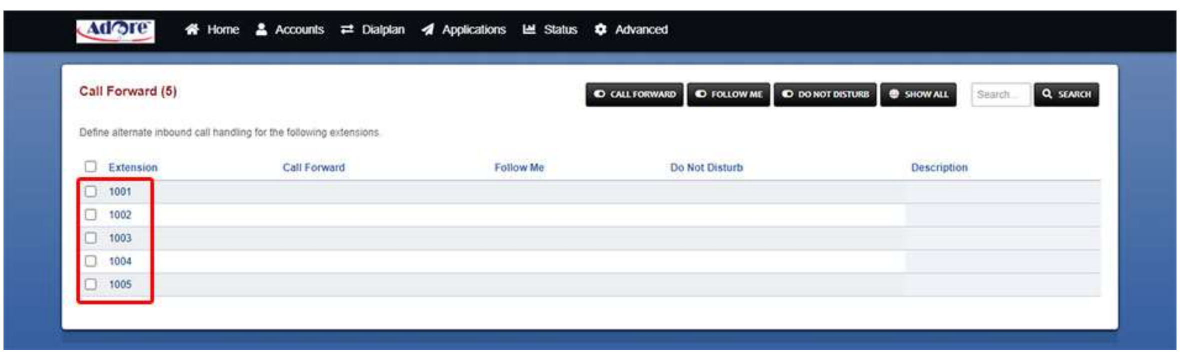
Call Forward

Define alternate inbound call handling for the following extensions.

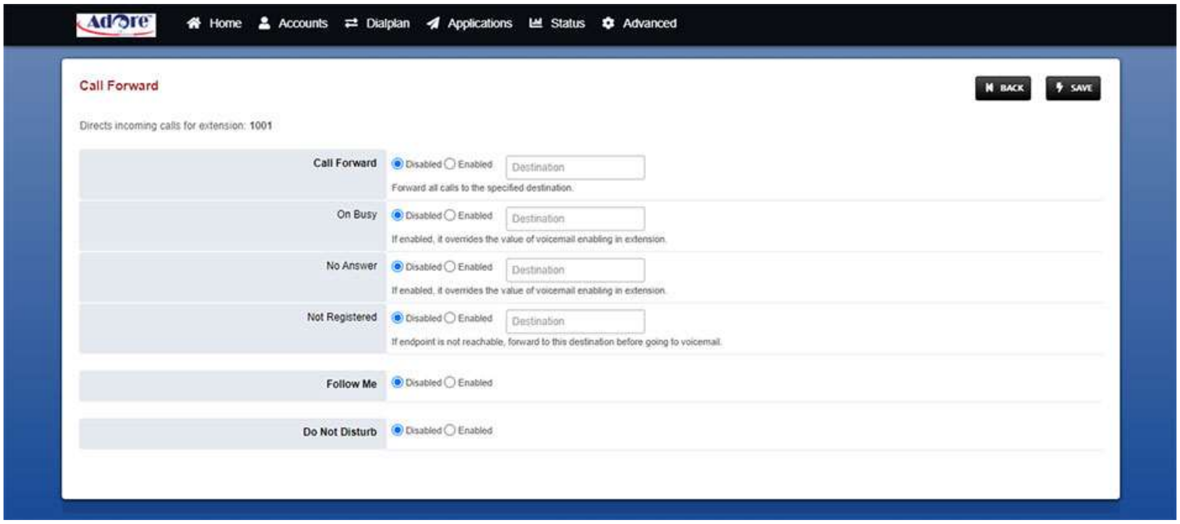
Go to **Applications** -> **Call Forward**

in this section you get all extension list, select extension from list and click Call forward option.





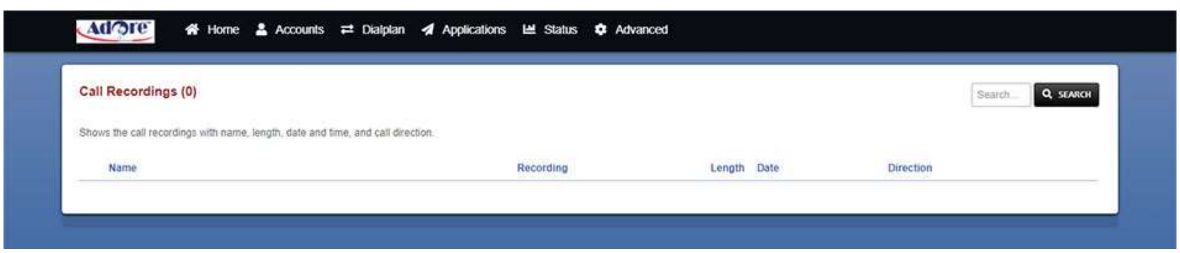
Here you can sett call forward option of each entension.



Call Recordings

Here shows the call recordings with name, length, date and time, and call direction.

Go to **Applications** -> **Call Recordings**

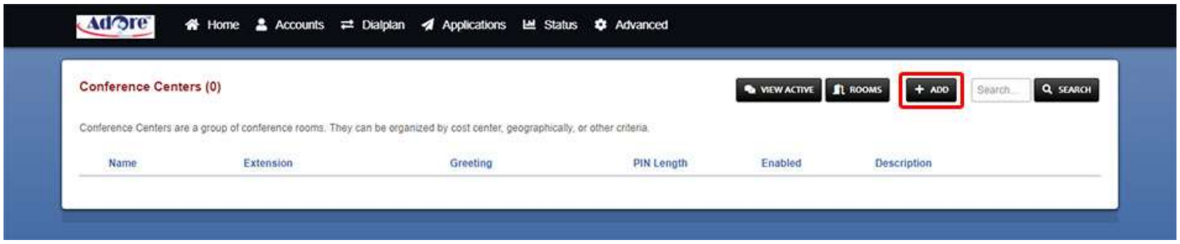


Conference Centers

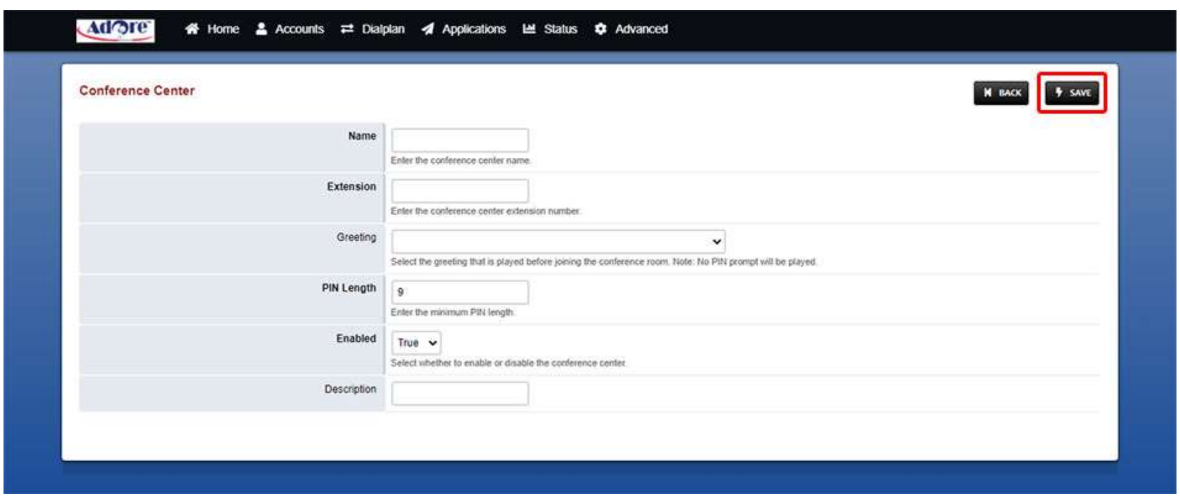
Conference Centers are a group of conference rooms. They can be organized by cost center, geographically, or other criteria.

Go to **Applications** -> **Conference Centers**

For adding Conference Centers , Please click on **Add** button



Name	Name of the Conference Center.
Extension	Extension of the Conference Center. (Be sure to not use an extension already in use)
Greeting	Choose a greeting to play.
Pin Length	Add a layer of security for entering the Conference Center.
Enabled	Enable or disable the Conference Center.
Description	A way to organize what the Conference Center is for.

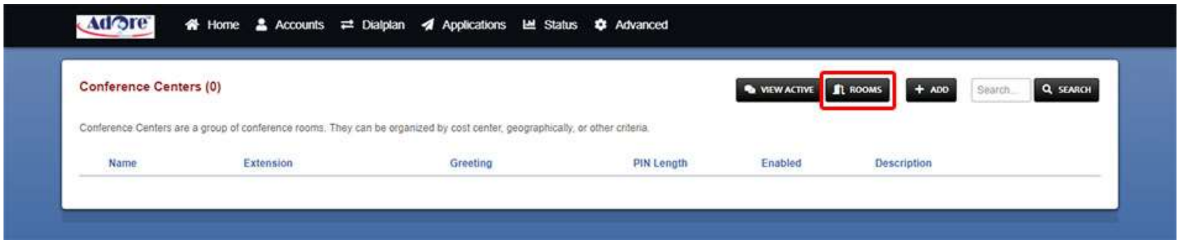


Conference Center Rooms

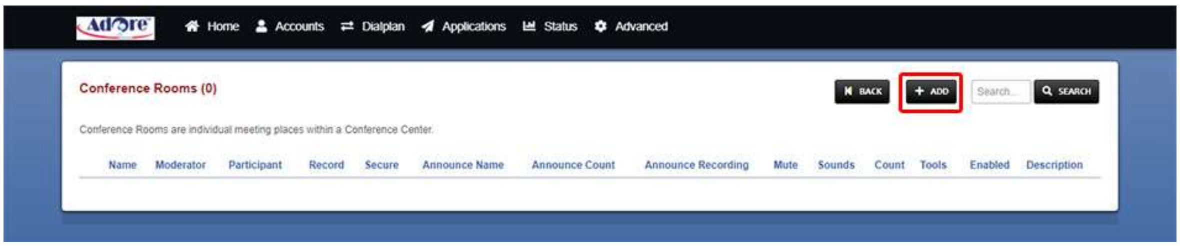
Applications -> **Conference Center** > Click **Rooms** at the top right. This will take you to the Conference Center Rooms. From here you can

- Create a Room

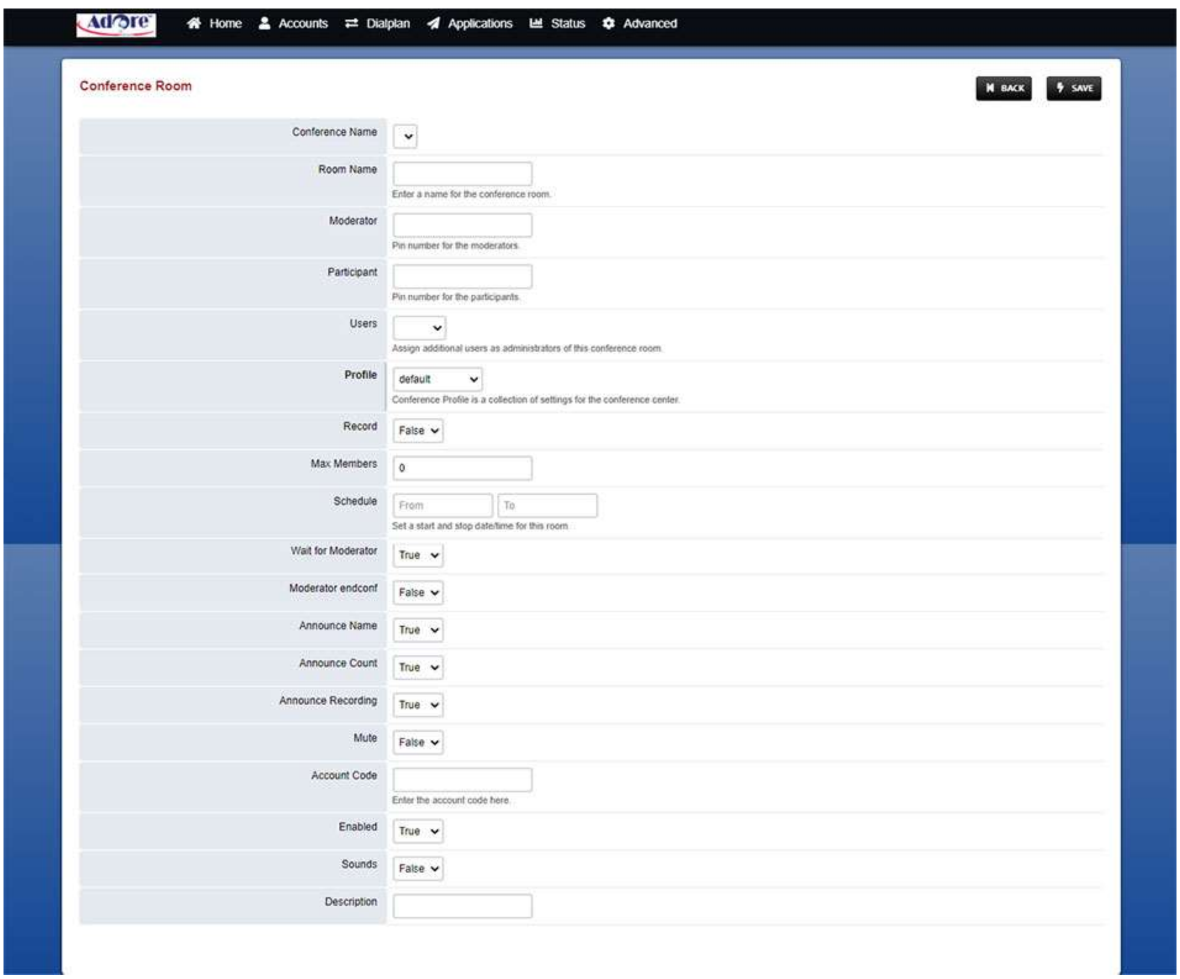
- Edit a Room



Click **Add** button to add **Rooms**



Add the desired details for creating Conference Center Rooms

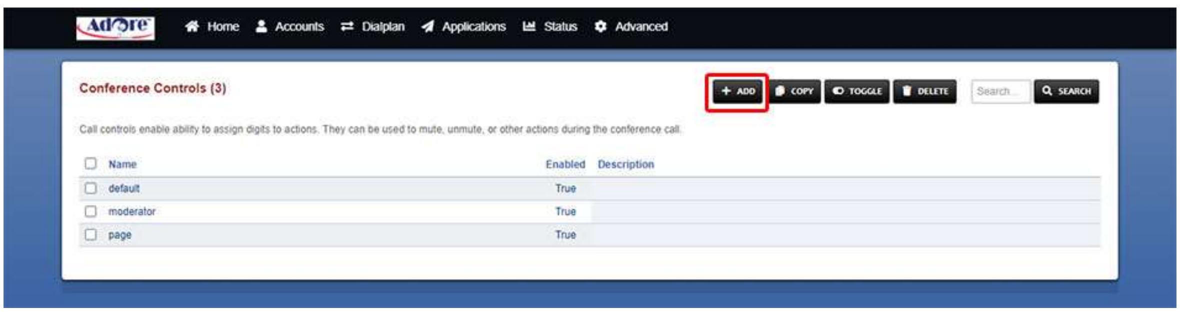


Conference Controls

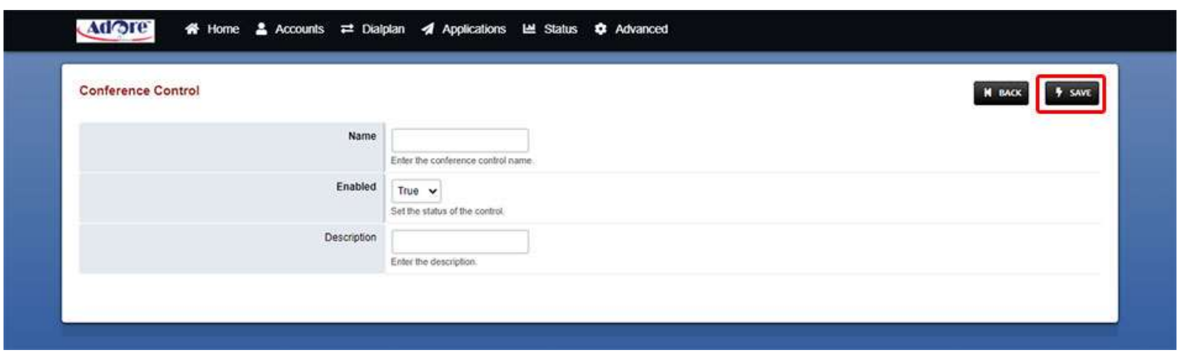
Call controls enable ability to assign digits to actions. They can be used to mute, unmute, or other actions during the conference call.

Go to **Applications** - > **Conference Controls**

Click **Add** button to add Conference Controls



Fill desired details and click save to save Conference Controls

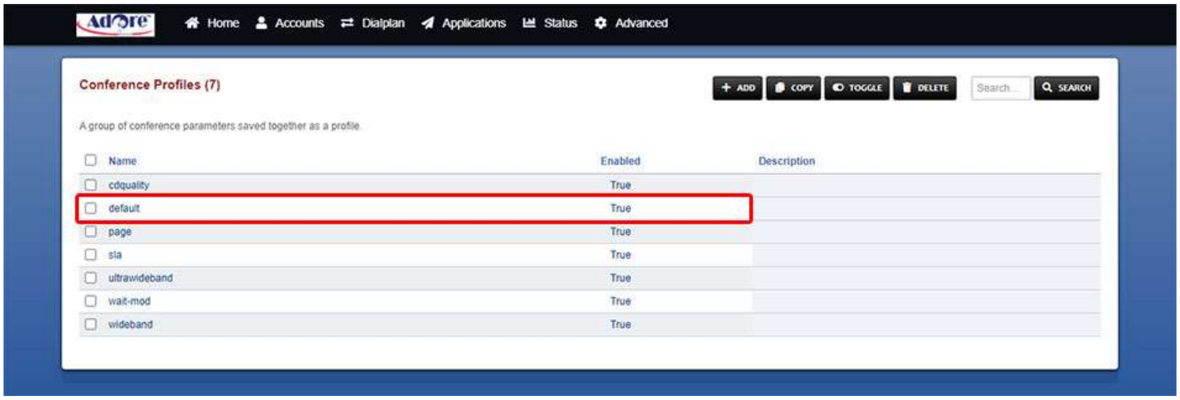


Conference Profiles

A group of conference parameters saved together as a profile.

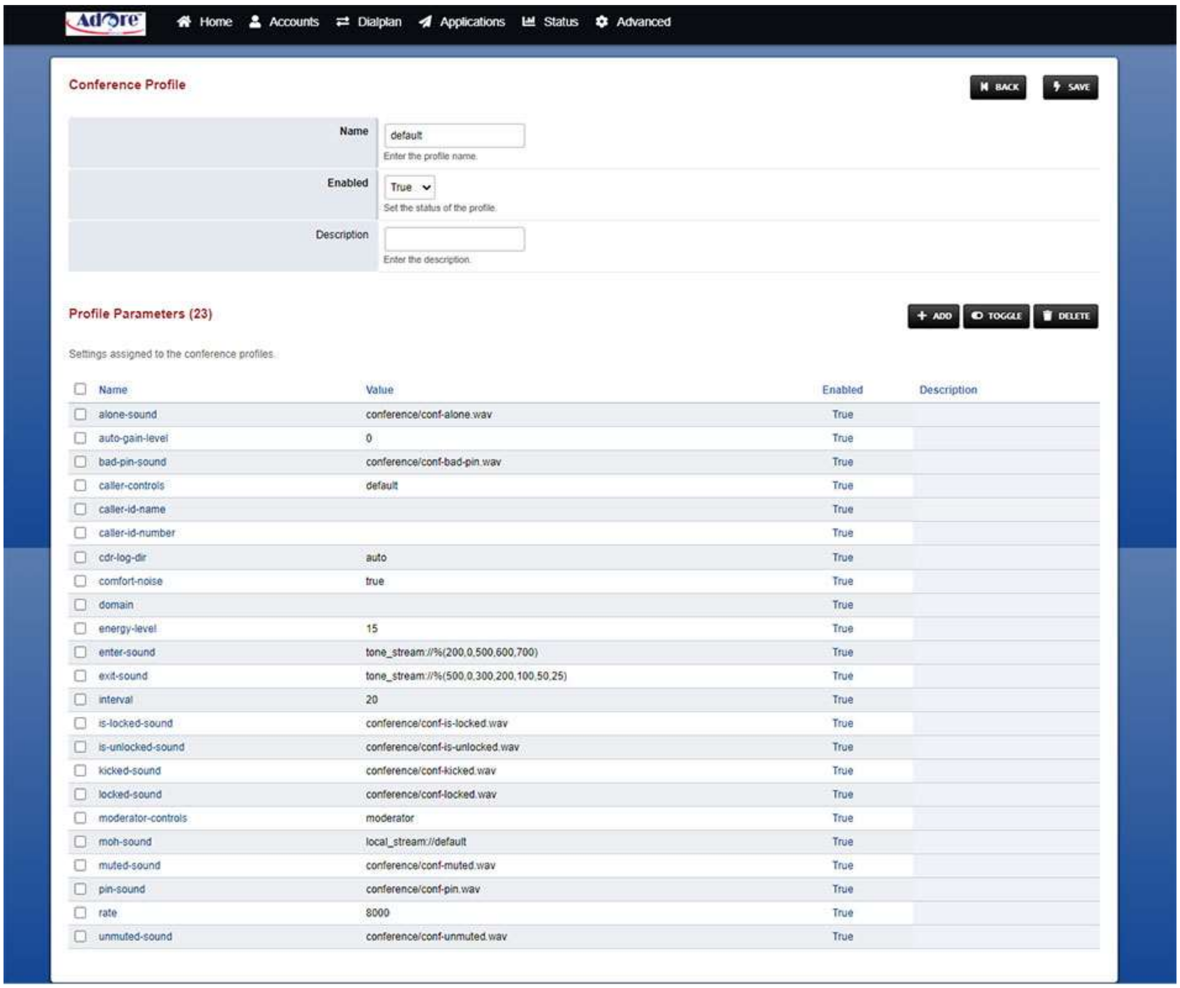
Go to **Applications** -> **Conference Profiles**

Click on **Default** setting, if you required any changes, if not please leave this settings



- **alone-sound**: conference/conf-alone.wav is the default.
- **auto-gain-level**: 0 is the default.
- **bad-pin-sound**: conference/conf-bad-pin.wav is the default.
- **caller-controls**: leave default setting
- **caller-id-name**:
- **caller-id-number**:
- **cdr-log-dir**: Set as auto. Could be set manually and is enabled.
- **domain**: enabled.
- **energy-level**: 15 is the default.
- **enter-sound**: tone_stream://%(200,0,500,600,700) is the default.
- **exit-sound**: tone_stream://%(500,0,300,200,100,50,25) is the default.
- **interval**: 20 is the default.
- **is-locked-sound**: conference/conf-is-locked.wav is the default.
- **is-unlocked-sound**: conference/conf-is-unlocked.wav is the default.
- **kicked-sound**: conference/conf-kicked.wav is the default.
- **locked-sound**: conference/conf-locked.wav is the default.
- **moderator-controls**: moderator is the default.

- **moh-sound**: local_stream://default is the default.
- **muted-sound**: conference/conf-muted.wav is the default.
- pin-sound: conference/conf-pin.wav is the default.
- rate: The rate in kHz. 8000kHz and is enabled.
- unmuted-sound: conference/conf-unmuted.wav is the default.

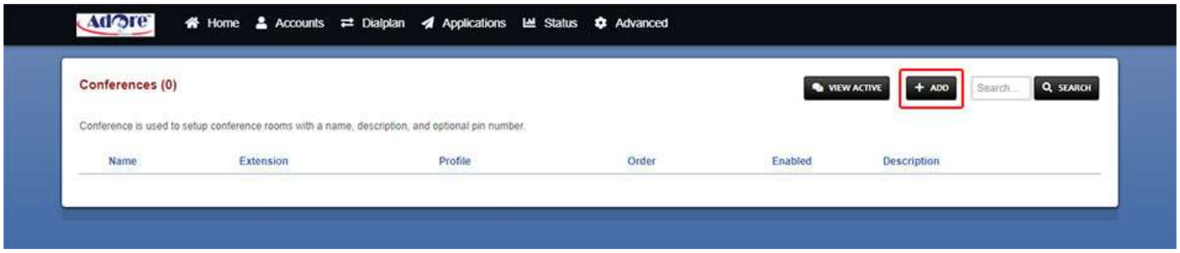


Conferences

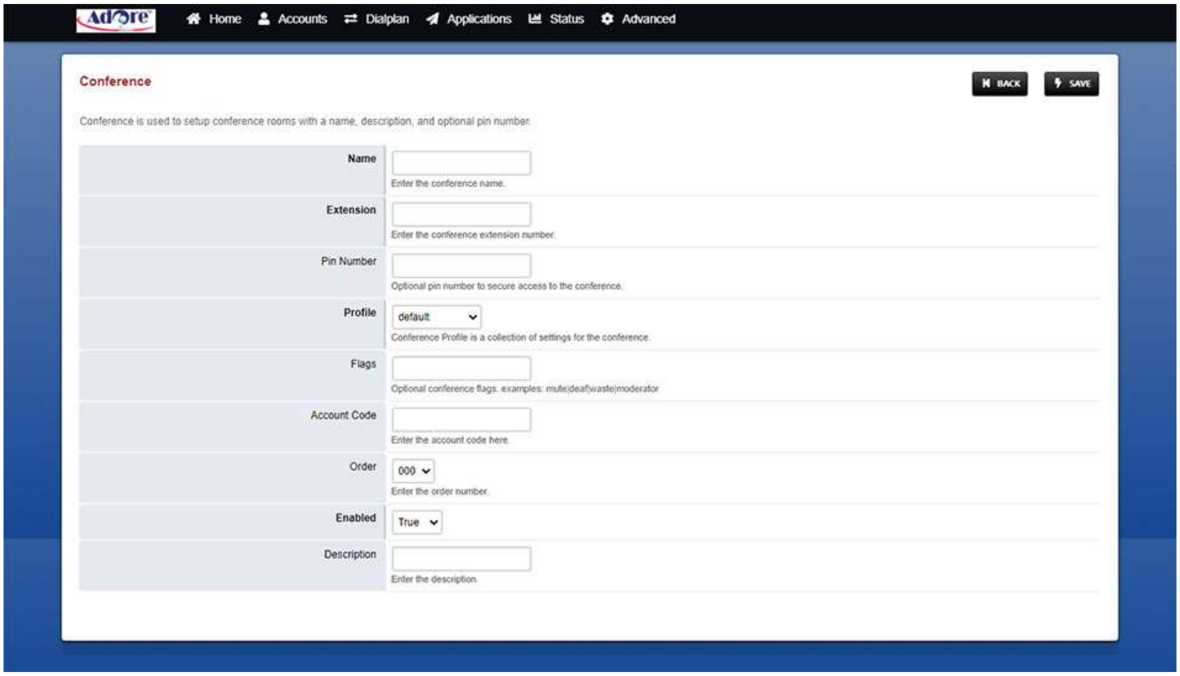
Conferences is used to setup conference rooms with a name, description, and optional pin number.

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

Click **Add** button to add conferences



- **Name**: Name for the conference.
- **Extension**: The number for the extension the user will dial.(Be sure it doesn't exist before creating it.)
- **Pin Number**: If you want to add a layer of security to enter the conference.
- **Profile**:
    - **Default**- The default audio quality rate and video.
    - **wait-mod**- Wait Mod setting.
    - **wideband- Wideband** audio quality rate and video.
    - **ultra-wideband**- Ultra wideband quality rate and video.
    - **cdquality**- CD Quality rate and video.
    - **page**- Page setting.
- **Flags**: mute|deaf|waste|moderator (Other values are available also)
- **Order**: The order of the conference.
- **Enabled**: If the conference is enabled.
- **Description**: A way to organize what the conference purpose is.



Enable Conferences

By default Conferences use to be hidden from the menu.

To add Conferences to the menu goto **Advanced** > **Menu Manager** and click the **default**

Then click the pencil edit icon on the right of Conferences

Select from the Groups dropdown list superadmin and click add then save

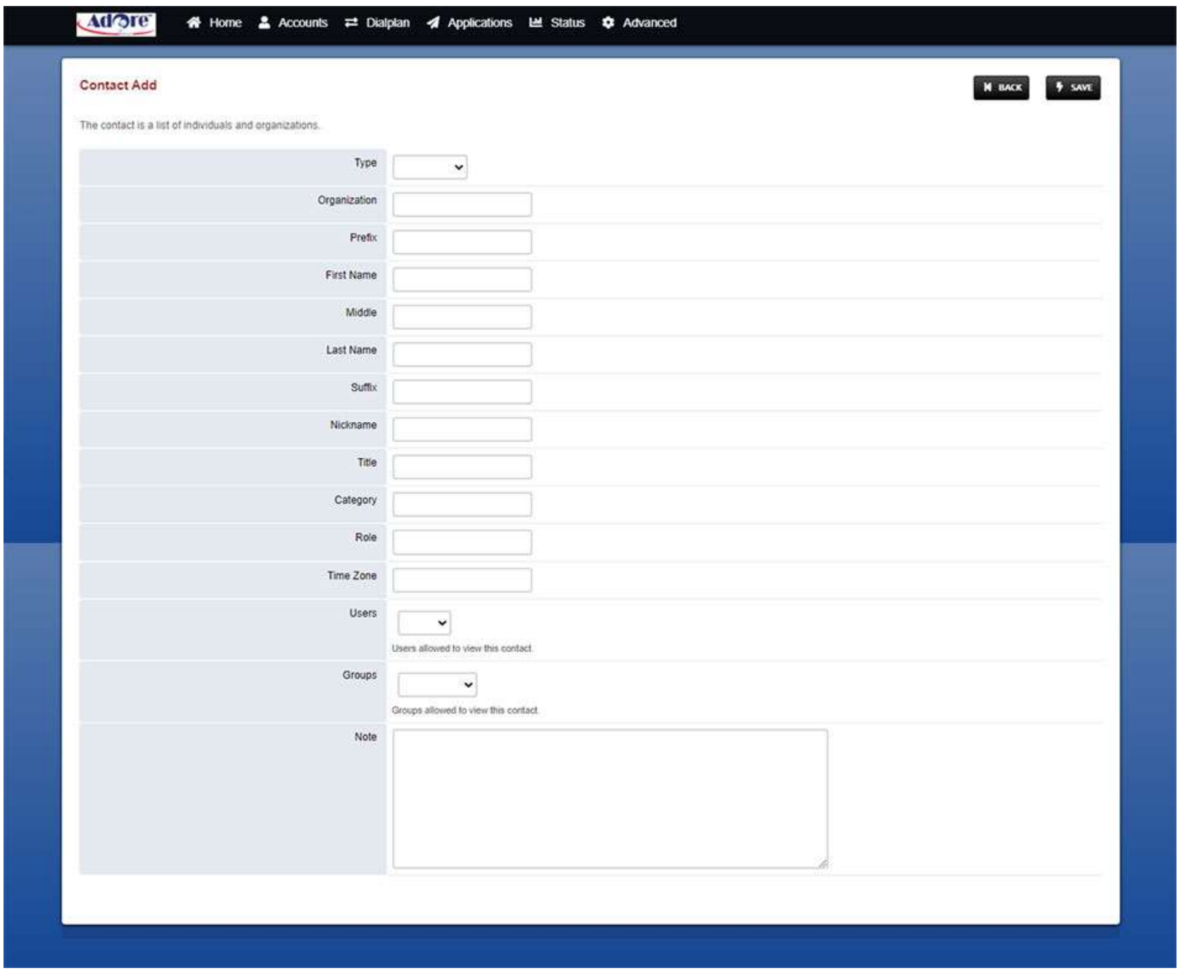
---

Contacts

Go to **Applications**- **Contacts**

Contacts is a list of individuals and organizations.

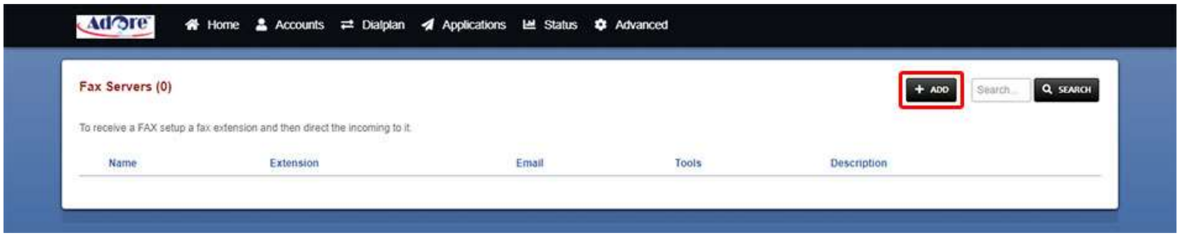
- Fill out the fields with pertinent information and click save.
- Users- Select the users that are allowed to view the contact
- Groups- Select the group that are allowed access to the contact.



Fax Server

Go to **Applications - Fax Server**

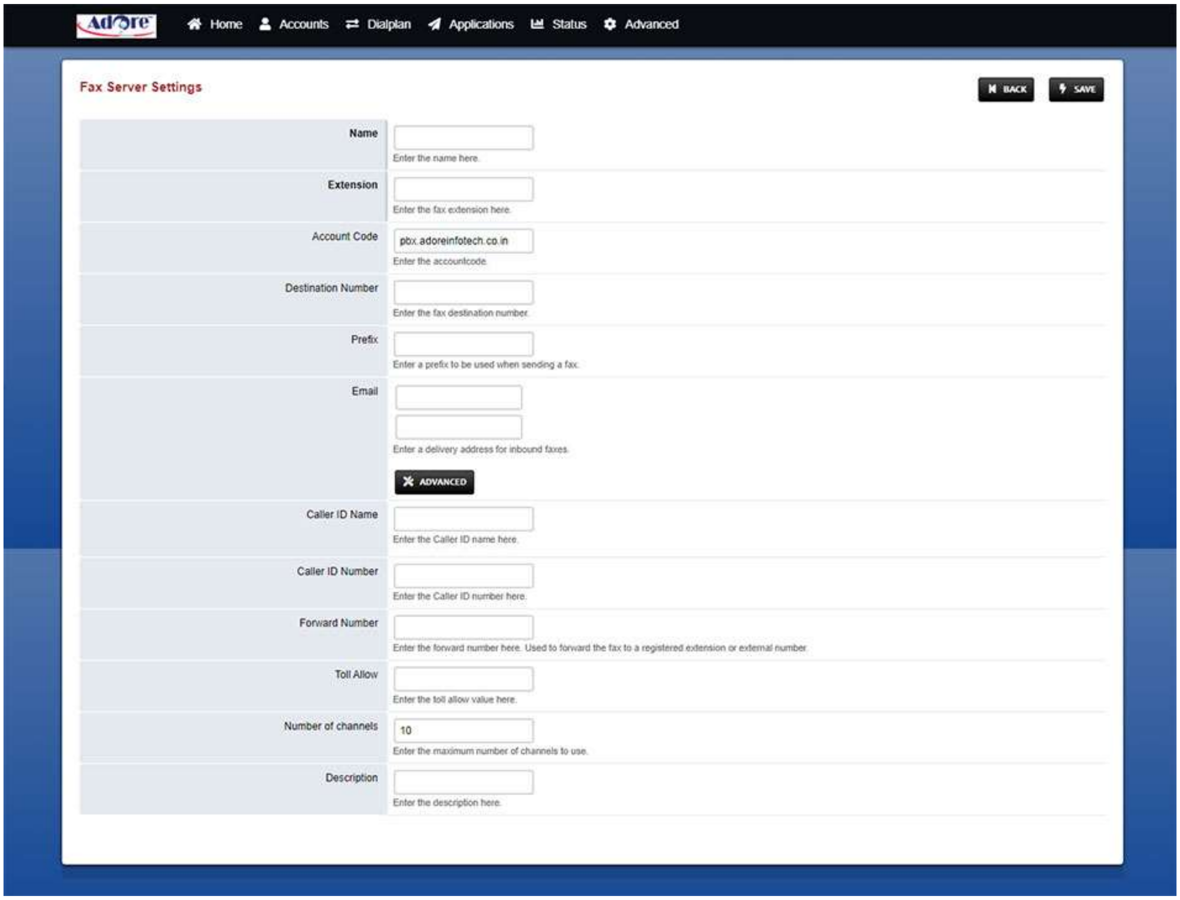
To receive a FAX setup a fax extension and then direct the incoming to it.



There are more settings for fax under Advanced > Default Settings then fax category.

- **To create a fax server goto App > Fax Server. Click the Add button on the top.**
    - **Leave the Destination Number** blank or faxing wont work.
- Destination Number is used in the Fax Server Dial Plan and is set based on the fax server internal extension number.
- Define the fields, the ones in bold are required. It is a good idea to organize so define the name thoughtfully.
- The extension you must use one that is not already created.
- Account Code should autofill. Again, leave the Destination Number blank.
- A prefix can be defined when sending a fax.
- Email is for inbound faxes and will be on the server and sent to the defines email.
- Define the Caller ID Name and Number.
- Leave the Forward Number and Greeting blank for normal settings.
- Number of channels define with a numerical value or keep blank for a default value.
- Keep organized by adding a Description.

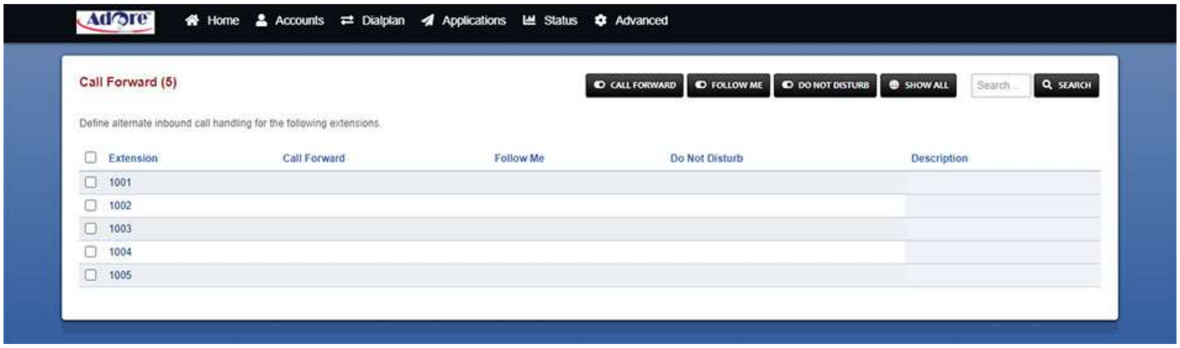
Once Fax Server Settings done , you are able to sent fax by using system.



Follow Me

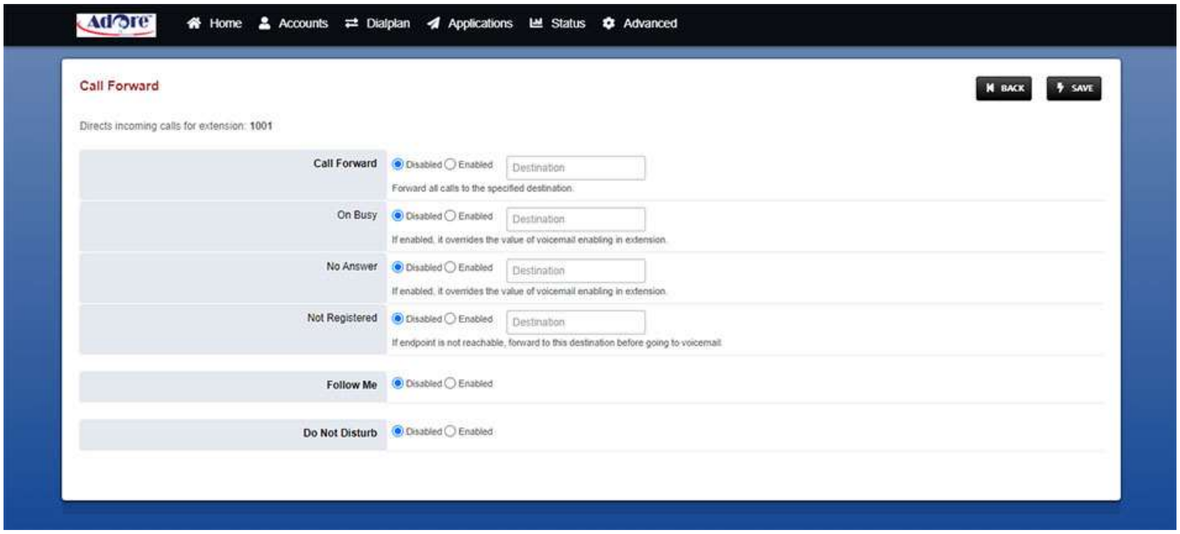
Go to **Applications** - **Follow Me**

Define alternate inbound call handling for the extensions.



- **Call Forward**- (Disabled or Enabled) Input the destination number
- **On Busy**- (Disabled or Enabled) If enabled, it overrides the value of voicemail enabling in extension
- **No Answer**- (Disabled or Enabled) If enabled, it overrides the value of voicemail enabling in extension
- **Not Registered**- (Disabled or Enabled) If endpoint is not reachable, forward to this destination before going to voicemail
- **Follow Me**- (Disabled or Enabled)
- **Destinations**- Can set Delay, Timeout and Prompt to accept the call.
- **Ignore Busy**- (Disabled or Enabled)
- **Do Not Disturb**- (Disabled or Enabled)

This example has both the extension 1001 itself and and external number to call. If you don't put the extension itself the extension wont ring when in Follow Me. This is due to the flexible nature of multi tenant PBX where if you didn't want that extension to ring like if you were out of the office on a business trip.

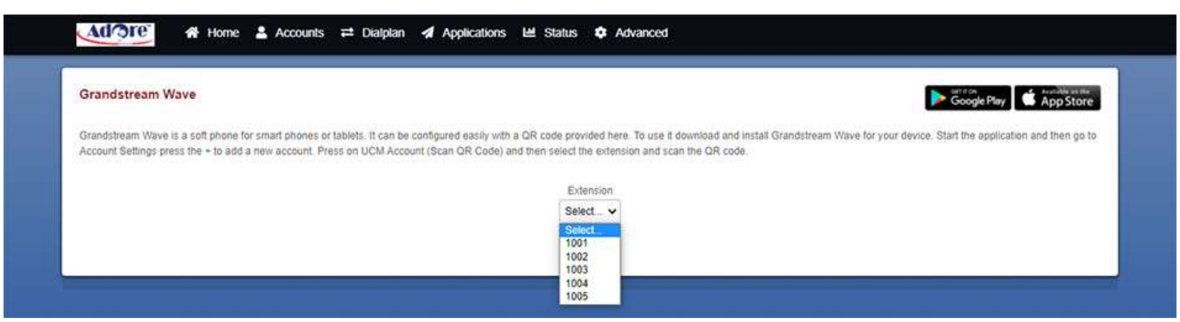


GS Wave

Go to **Applications** - **GS Wave**

Grandstream Wave is a soft phone for smart phones or tablets. It can be configured easily with a QR code provided in your Multi Tenant PBX installation.

- To use it download and install Grandstream Wave for your mobile device.
- Start the Grandstream Wave application on your mobile device.
- Then go to the Grandstream Wave Account Settings and press the plus+ to add a new account.

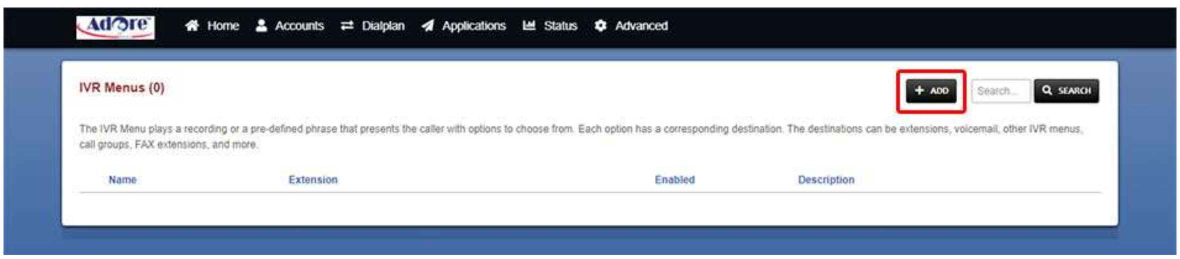


IVR Menu

Go to **Applications** - **IVR Menu**

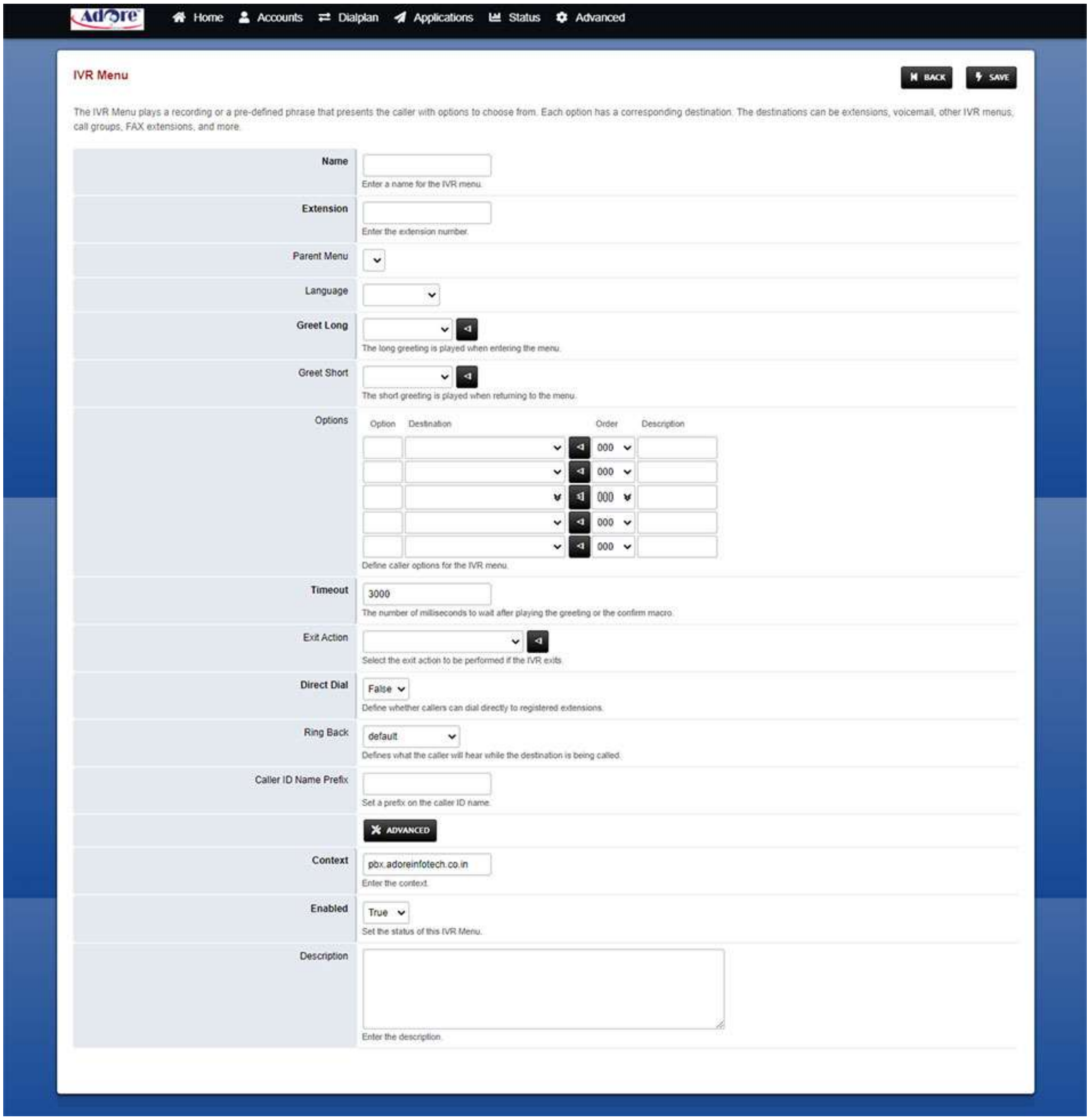
The IVR Menu plays a recording or a pre-defined phrase that presents the caller with options to choose from. Each option has a corresponding destination. The destinations can be extensions, voicemail, other IVR menus, call groups, FAX extensions, and more.

Here you can **ADD** IVR Menu



- **Name**: Enter a name for the IVR menu
- **Extension**: Enter the extension number (This must a new extension that isn't allready created)
- **Greet Long**: The long greeting when entering the menu.
- **Greet Short**: The short greeting is played when returning to the menu.
- **Options**: Define caller options for the IVR menu.
- **Timeout**: The number of milliseconds to wait after playing the greeting or the confirm macro.
- **Exit Action**: Select the exit action to be performed if the ivr exists.
- **Direct Dial**: Define whether the callers can dial directly to registered extensions.
- **Ring Back**: Defines what the caller will hear while the destination is being called.

- **Caller ID Name Prefix**: Set a prefix on the caller ID name.
- **Enabled**: set the status of the IVR Menu.



You can get very creative with IVR's and are almost limitless in possibilities. In the basic example below we;

- Name the IVR "IVR Main"
- Extension "200"
- Greet Long a phrase that was made from the phrase section under apps

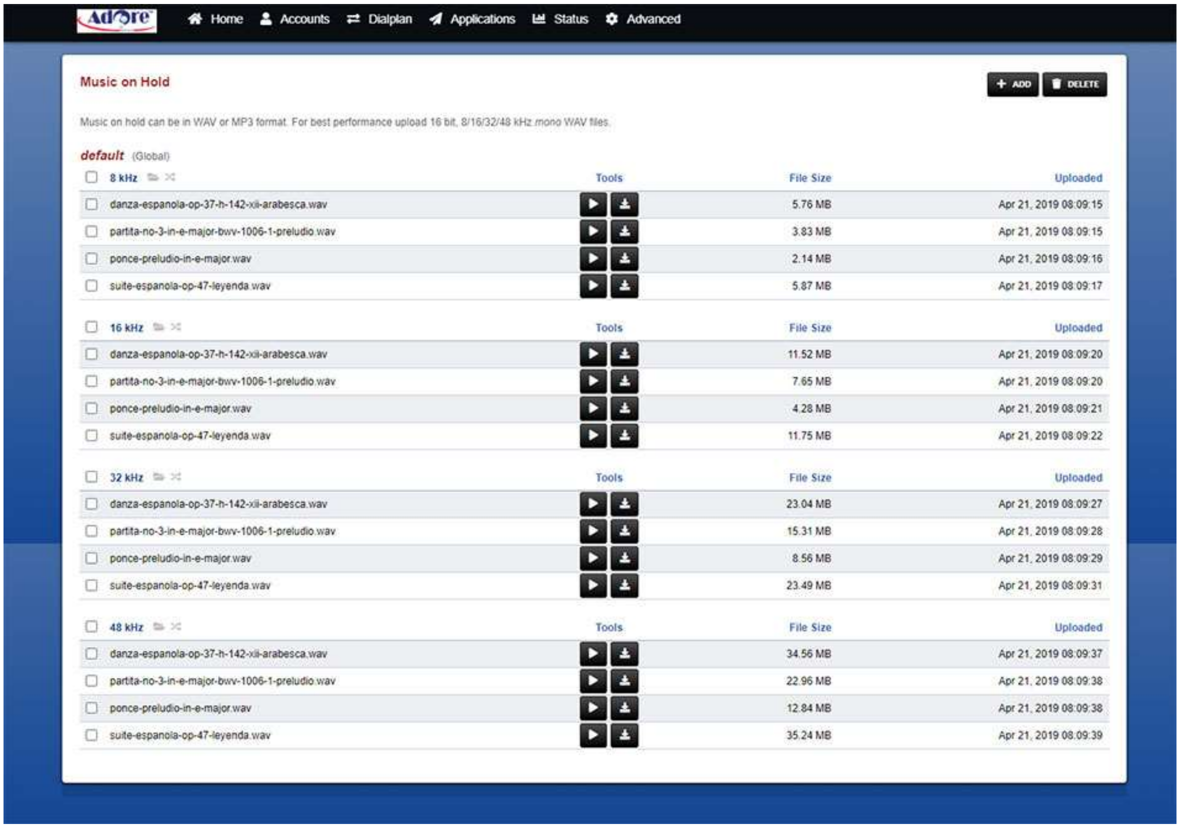
- Number entry in options, choose an extension for Destination and descriptions ie sales, billing, tech support, and after hours. timeout 3000 milliseconds
- Exit Action to the extension 109 (after hours)
- Direct Dial to False and Ring back to Default.

Once IVR add, You now have a list of IVR’s to go back to and edit or delete as needed.

Music On Hold

Go to **Applications** - **Music On Hold**

Music on hold can be in **WAV** or **MP3** format. To play an MP3 file you must have mod_shout enabled on the ‘Modules’ tab. You can adjust the volume of the MP3 audio from the ‘**Settings**’ tab. For best performance upload 16 bit, 8/16/32/48 kHz mono **WAV** files.



Music on Hold Tips

When a new music on hold category mod_local_stream will be restarted. If it is busy then it will not restart automatically. A manual restart of the module is required when it is not in use. The module can be restarted from the Menu -> Advanced -> Modules or from the console and fs_cli with following command.

Command : **reload mod_local_stream**

Each music on hold category is given a name. If the domain is set to global the name will be the name in the example below the protocol that is used is local_stream and the music on hold category is default and domain is set to global.

Command : **local_stream://default**

It is possible that a domain or tenant can have its own category of music. In this example the name is ‘custom’ and the domain was assigned automatically to the current domain.

Command : **local_stream://domain_name/custom**

Operator Panel

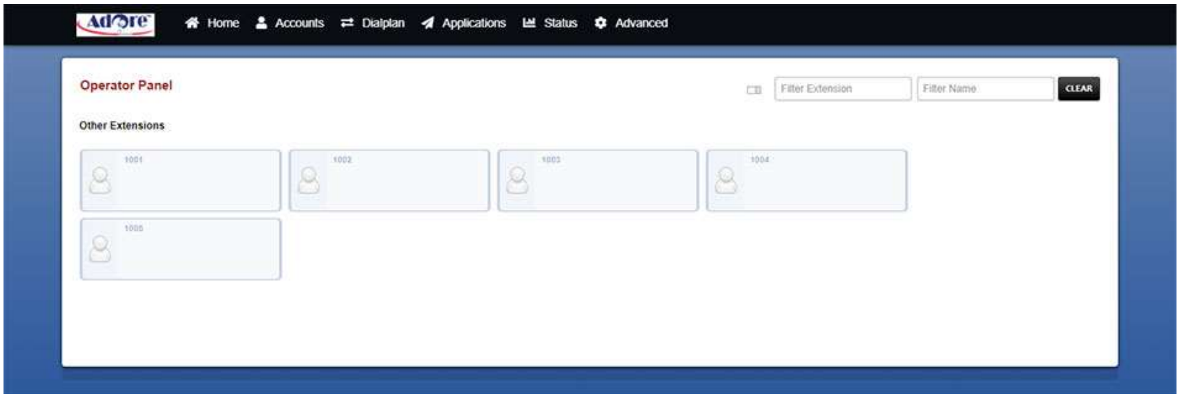
Go to **Applications** - **Operator Panel**

Operator Panel is a simple and easy way to use the Multi Tenant PBX web interface to:

- Make calls from.
- See who is on a call.
- Eavesdrop on a call.
- Hangup your own call.
- Drag and drop blind transfer an active call.
- Drag and drop calling to other users.
- Login and out of queues and call center.

You can see the status of other users also depending on what permissions are set to the user.

NOTE : Make sure in Accounts > Extensions that the extension is assigned to the user. This will enable Operator Panel for that user.



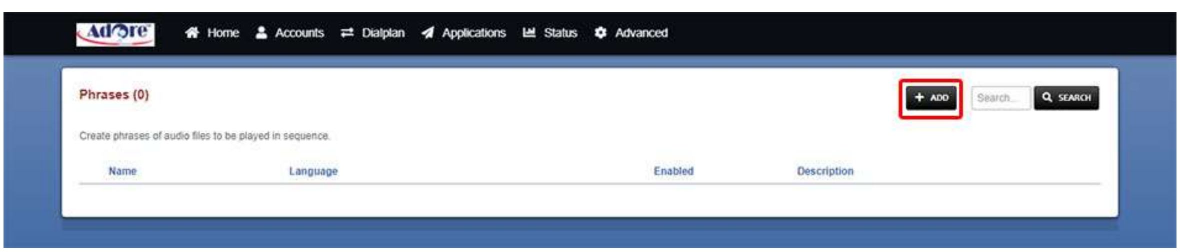
Operator Panel Status

- **Available**: The user will receive a call.
- **On Break**: The user won't receive a call but can still receive a call from other users that directly call.
- **Do Not Disturb**: The user won't receive any calls.
- **Logged Out**: The user won't receive any calls as they are logged out.

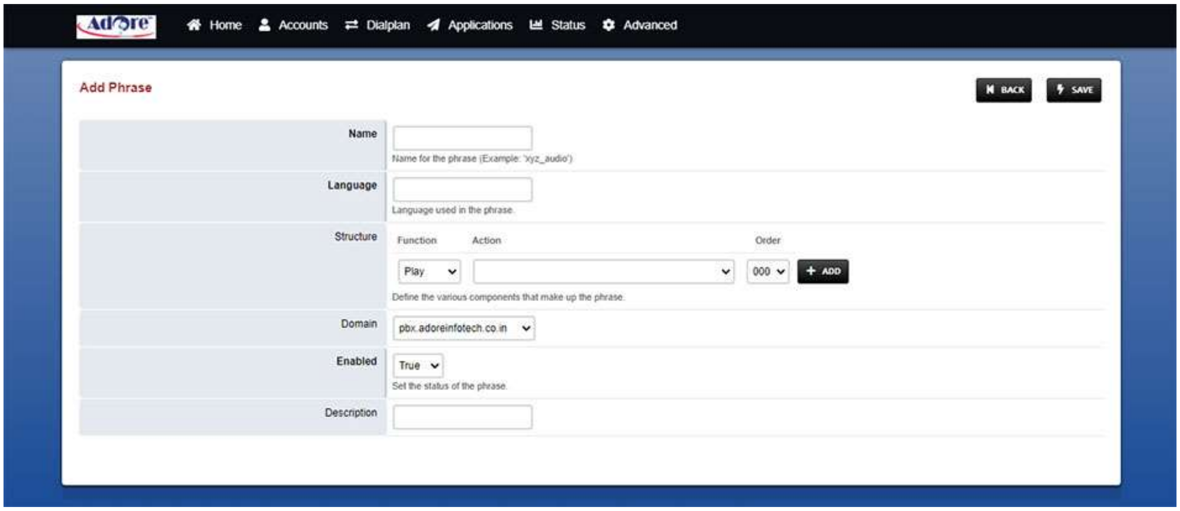
Phrases

Go to **Applications** - **Phrases**

Here you can create phrases of audio files to be played in sequence.



Fill the desired required fields and click save.



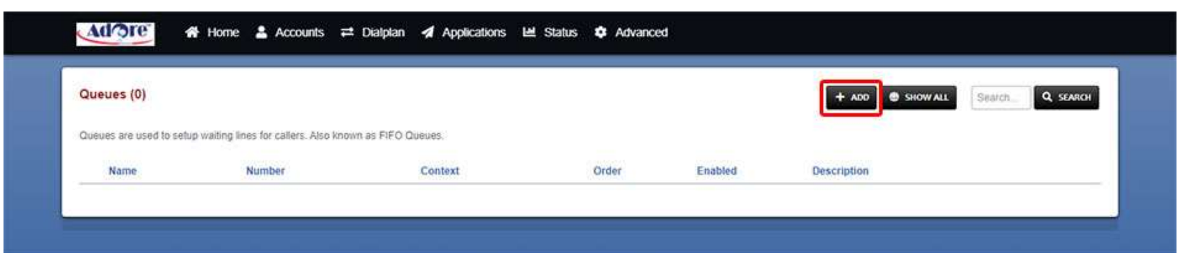
Queues

Go to **Applications** - **Queues**

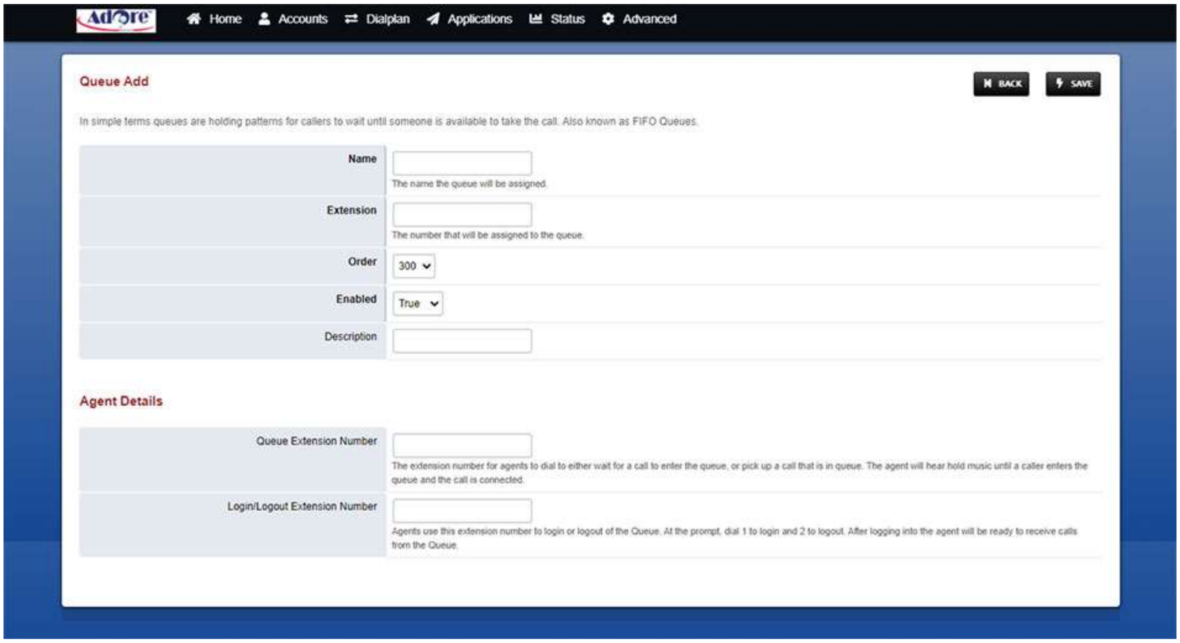
Queues are used to setup waiting lines for callers. Also known as FIFO Queues.

The Queues feature is rarely used for call center type work. When needed, Call Center is usually used instead.

Click **Add** button to add Queues



fill the desired field here and click save button.



Recordings

Go to **Applications** - **Recordings**

Create a Recording

- Dial extension (Ex:*1001) and wait for the voice prompt
- Enter the password (pin_number) followed by the pound sign# Enter at least a 3 digit number. This will label the recording file. (recording100.wav)
- start talking to make the recording after the voice prompt and press the pound key #
- Press 1 to accept the recording then hang up or press 2 to start over.

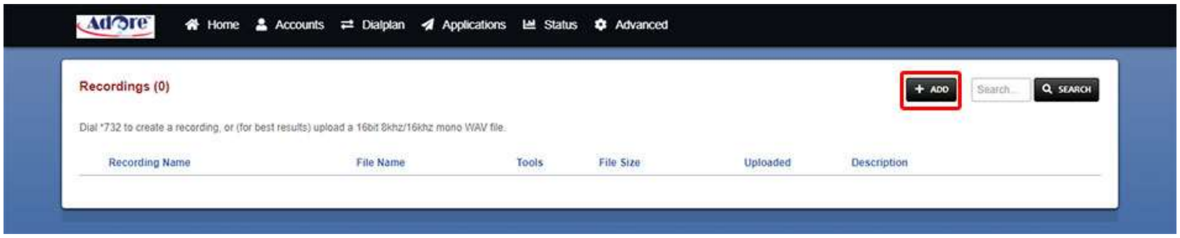
Edit Recording

- Click the edit pencil icon.
- Rename as needed.

- Click save to save the changes.

Applying Recordings

Once you have a recording made you can use the recordings in different area's of multi tenant PBX. Custom IVR's and phrases would be the typical uses.

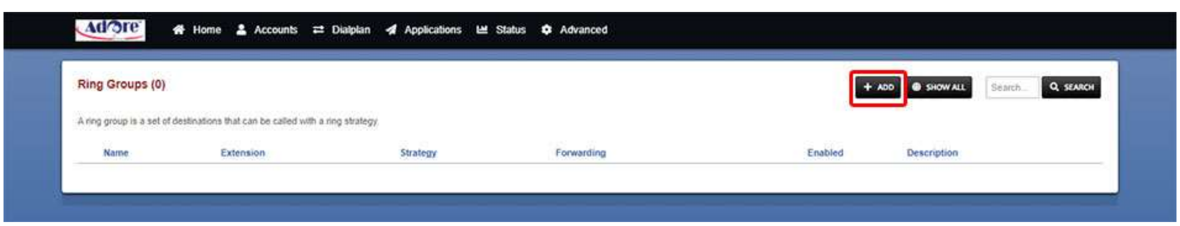


Ring Groups

Go to **Applications** - **Ring Groups**

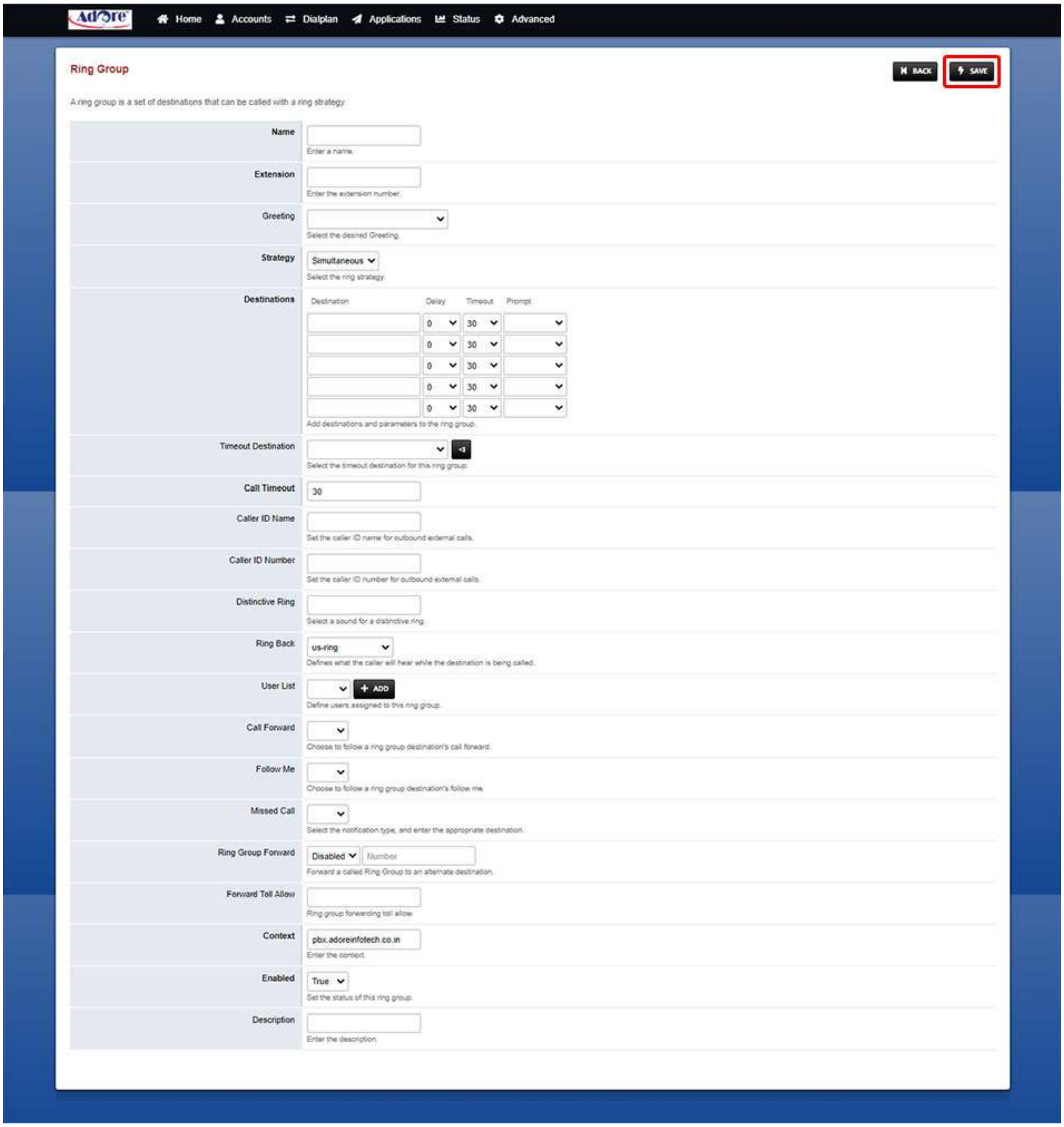
A ring group is a set of destinations that can be called with a ring strategy.

Click **Add** button to add **Ring Groups**



- **Name -** A meaningful name for this ring group. This name is used in th Destination select list.
- **Extension -** The extension number for this ring group.
- **Greeting -** Play a sound file upon calling the Ring Group extension.
- **Strategy -** The selectable way in which the destinations are being used.

    - **Simultaneous -** Rings all destinations. All destination share the same thread.
    - **Sequence -** Calls destinations in sequence where order that is lower goes first.
    - **Enterprise -** Ring all destinations. Each destination uses its own thread.
    - **Rollover -** Calls destinations in sequence and skips busy destinations.
    - **Random -** A random destination will ring.
- **Destinations -** The destination numbers are the numbers for the ring group to call. Destinations can only be local registered endpoints or external numbers.
- Extensions Local registered extensions.
- External numbers Destinations out to an external number.
- **Prompt -** Where you determine if the call must have a dial to confirm before a pickup event.
- **Caller ID Name Prefix -** The string that is added to the caller ID when it displays on the ringing extension.
- **Caller ID Number Prefix -** The Number that is added to the caller ID when it displays on the ringing extension.
- **Ring Back -** What the caller hears when they are waiting for the Destinations to answer. (ex. Music on Hold, us-ring)
- **Context -** The context defaults to the domain name.
- **Enabled -** True or False
- **Description** - As per you reference.



Ring Group Example

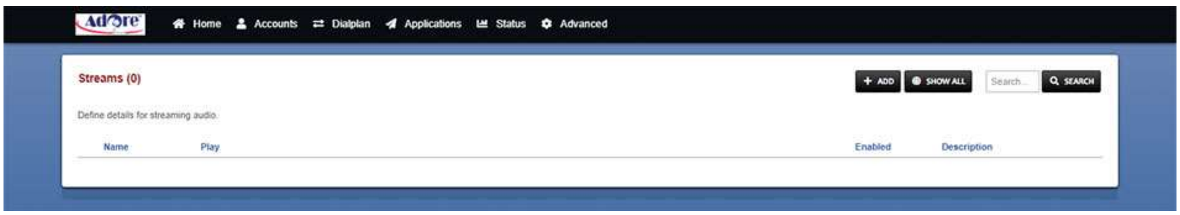
We have 4 extensions all ring at the same time until one of them pick up first. Click the + to create a ring group. Fill in the fields that are in bold. In the Extension box type a number that is NOT already created. This new extention won't be in the extension list. The **strategy** will be Simultaneous. Enter in the destination the 4 extensions 1001, 1002, 1003, 1004.

Streams

Go to **Applications** - **Streams**

Here you can define details for streaming audio.

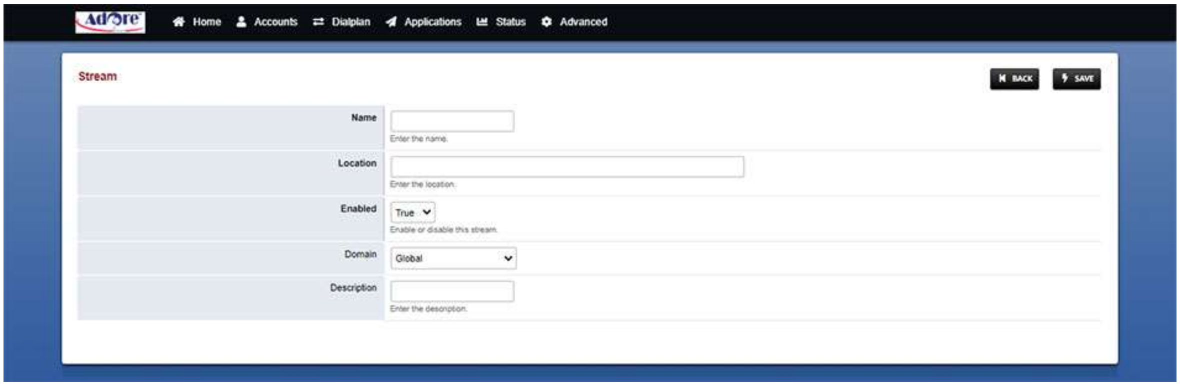
Click **Add** button



- Make sure **mod_shout** is installed and is started.
- Have a shoutcast url ready to use. (shout://domain.tld/path/to/)
- To add a stream click the plus icon on the right

Fill the fields:

- **Name**: Can be anything
- **Location**: Must start with shout://
- **Enabled**: If you want the stream enabled
- **Domain**: Choose a domain that will only have the stream. Choose Global for all domains
- **Description**: To help organize ;-)



Editing a stream path will result in having to update anything that is using the stream. For example, if you have extension 500 using stream "Local Weather" and you edit the shout:// path then you will have to go back to extension 500 and reset the music on hold for extension 500. This is by design.

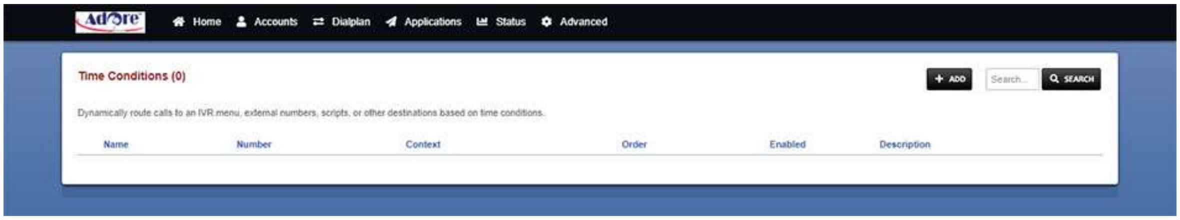
Please be aware of your countries copyright laws for streaming the content you are going to stream.

Time Conditions

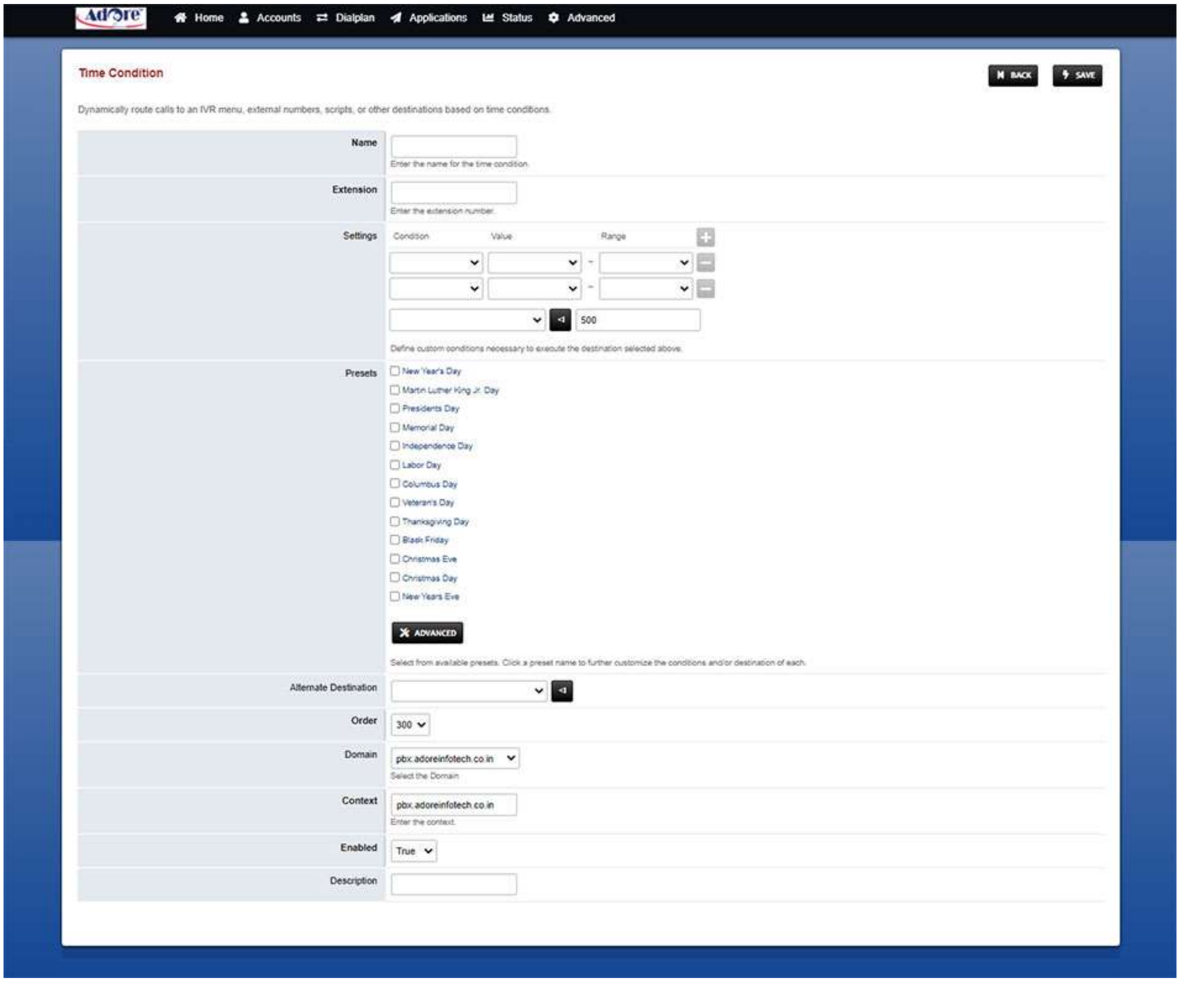
Go to **Applications** - **Time Conditions**

Here can do Dynamically route calls to an IVR menu, external numbers, scripts, or other destinations based on time conditions.

Click **Add** button here



- **Name -** Name of the Time Condition.
- **Extension -** Define an extension number that is NOT already created.
- **Presets -** US Holiday presets.
- **Alternate Destination -** If the condition doesn't match the call will go to the defined alternate destination.
- **Order -** Changes the order of which condition is evaluated first.
- **Domain -** Select Domain
- **Context** - Domain
- **Enabled -** If the ring group is enabled.
- **Description** - Description as you wish



Time Conditions Example

In our example we have an employee that will receive calls during a set time range and set days. Below is what the settings look like for Monday through Friday at 5:00pm to 11:00pm. If the employee doesn't answer the call will be directed to the Timeout Destination. Label the Name as Oncall and invent the Extension as 10011. In the Settings choose from the dropdown lists for Day of Week for the condition, Monday for the Value and Friday for the Range. Next set of dropdown list choose Time of Day for the condition, 5:00 PM for the value and 11:00 PM for the Range. If other options are needed just click the + to the right of Range.

Conditions

The most common conditions to use are Day of Week and Time of Day.

Time of Day

- Is a select list of every minute for the full 24 hour period of time.

Hour of Day

- Another alternative the Hour of Days. If you set a range of 9 - 4 it will include all of 4 until it changes to 5.

Day of Week

The day of week condition each day of the week is represented by a number. A valid range is from low to high. A valid range is like Monday to Friday (2-6).

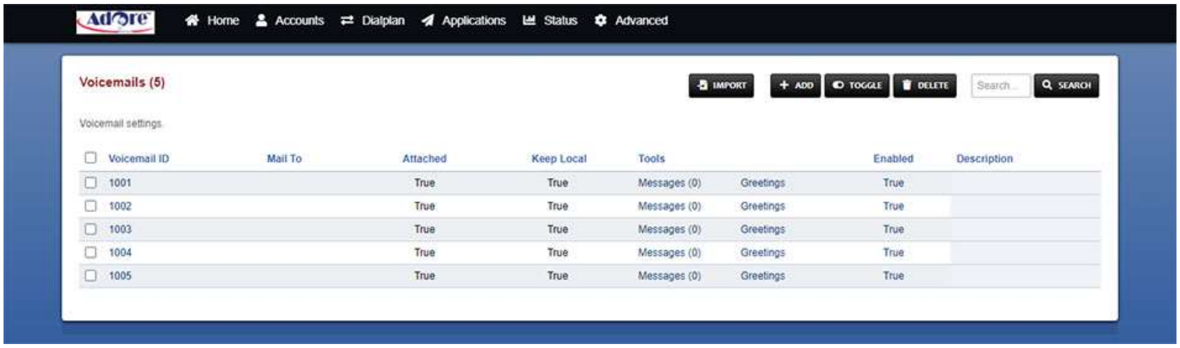
- 1 Sunday
- 2 Monday
- 3 Tuesday
- 4 Wednesday
- 5 Thursday
- 6 Friday
- 7 Saturday

An example of an invalid range would be Saturday to Sunday (7-1).

VoiceMail

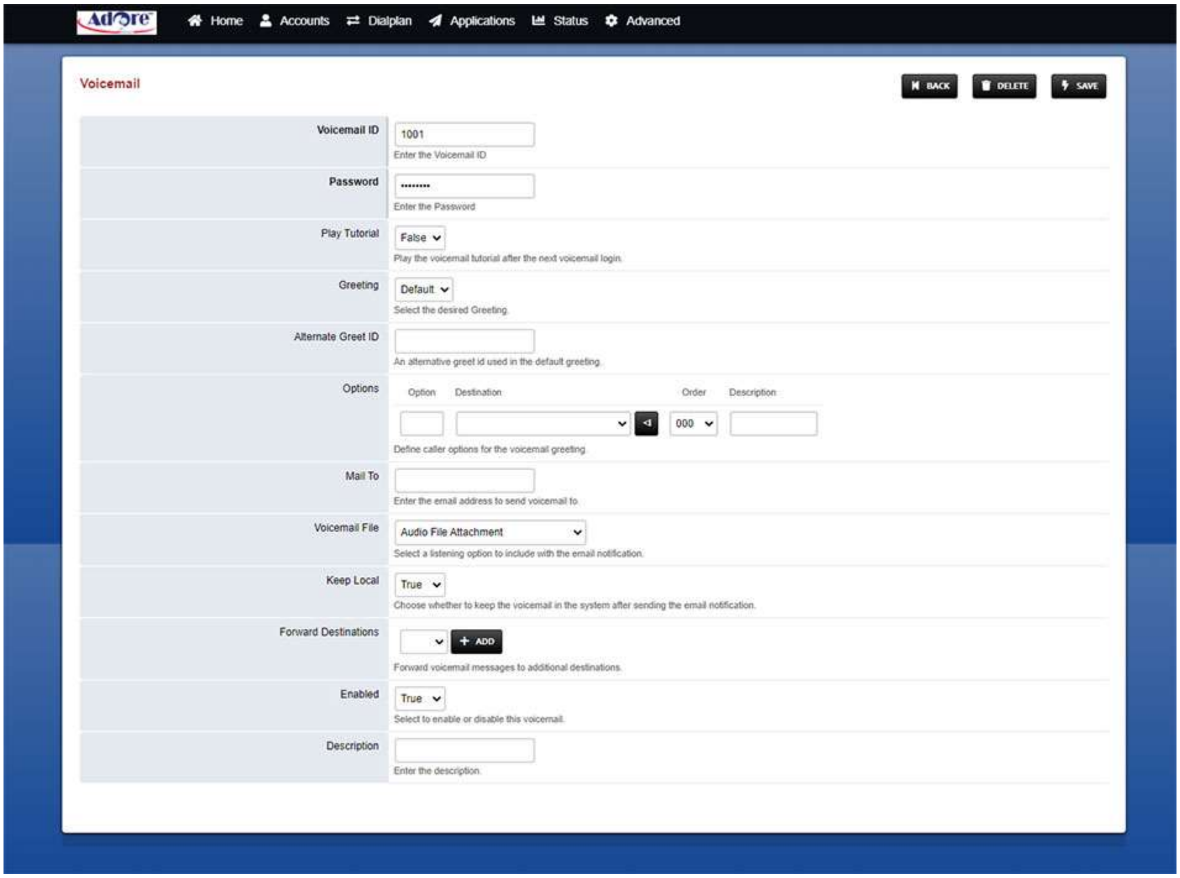
Go to **Applications** - **VoiceMail**

Here you can setup VoiceMail



Here you can edit voicemail settings.

- Play Tutorial- Play the voicemail tutorial after the next voicemail login
- Greeting- When you dial *97, record a greeting and set a number you can choose which greeting to use
- Alternate Greet ID- An alternative greet id used in the default greeting
- Options- Define caller options for the voicemail greeting
- Mail to- have voicemails emailed to this address
- Voicemail File- Select a listening option to include with the email notification
- Keep Local- Choose whether to keep the voicemail in the system after sending the email notification
- Forward Destinations- Forward voicemail messages to additional destinations
- Enabled- Enable or disable the voicemail box



Voicemail Options

To access an extensions voicemail **away** from the extension.

- Dial the extension and interupt the greeting with the ***star** key.

*97	To access **that** extensions voicemail **from the extension** or the voicemail button
*98	To access **any** extensions voicemail
*99[ext]	To access **a specific** extension voicemail

	Main Menu

press 5	For advanced options

	Advanced Options
press 1	Record a greeting
press 2	Choose a greeting
press 3	Record name
press 6	Change password
press 0	For main menu

Voicemail Variables

Using switch variables provides the ability to adjust Multi Tenant PBX Voicemail features. These variables can be set in either Dialplan -> global-variables or per domain with domain-variables dialplan.

Name	Value
skip_greeting	true or false
skip_instructions	true or false
voicemail_greeting_number	0-9
vm_disk_quota	0-3600 seconds
vm_message_ext	wav or mp3

voicemail_authorized	true or false
vm_say_caller_id_number	true or false
vm_say_date_time	true or false

Note: ‘wav’ format is the default voicemail message file type. A value of ‘mp3’ requires *mod_shout* be installed and running.

Not Found Message

When an extension is unavailable and no voicemail is configured, there is an option to play a message to the caller alerting them to this.

To enable/disable this, change the option for the **not_found_message** setting in **Advanced > Default Settings > Voicemail** category to suit your preference.

Please note that enabling this option means that the call must be answered in order to play the message to the caller and so the call will complete with a 200 OK rather than a 480 Unavailable or 486 Busy. In some jurisdictions this could potentially be illegal as it turns an otherwise toll free call into a chargeable one.

Voicemail Transcription

Multi Tenant PBX supports Voicemail Transcription, where emails will include a transcribed version of the voicemail the email was sent in regards to. To configure this feature, see applications/voicemail_transcription.rst.

8. Status

Status

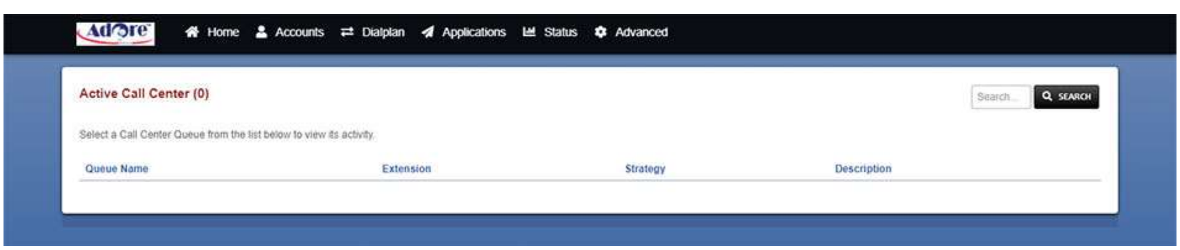
In the **Status** menu you have the following options :

- **Active Call Center**
- **Active Calls**
- **Active Conferences**
- **Active Queues**
- **Agent Status**
- **CDR Statistics**
- **Email Logs**
- **Extension Summary**
- **Registrations**

Active Call Centers

Go to **Status** -> **Active Call Centers**

Here you can Select a Call Center Queue from the list below to view its activity. From here you can view status, evesdrop on the call, transfer the call or click to call an available agent.



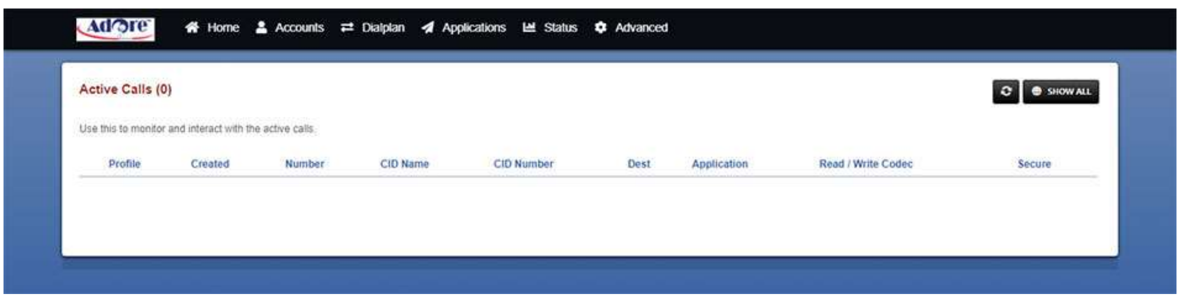
Active Calls

Go to **Status** - **Active Calls**

Here you can monitor and interact with the active calls in the system.

Here you can view the sip profile used, time the call was created, number, cid number, destination, application, Codecs used, and if the call is secure (encrypted)

- Once any call active you can click the **X** to end the call
- Click the **Show All** button to show calls in all domains.



Active Conferences

Go to **Status** - **Active Conferences**

Here list all the conferences that are currently active with one or more members.

When any conference call active you can use below option in details:

Red ball icon: If illuminated the conference is being recorded.

Lock: Can lock the conference from anyone else joining.

Unmute All: Unmute all the members.

End Conference: End the conference.

CID Name: Caller ID Name

CID Number: Caller ID Number

Capabilities: Icons show what capabilities each member have like hear/mute, talking, and video.

Joined: How long ago the member joined.

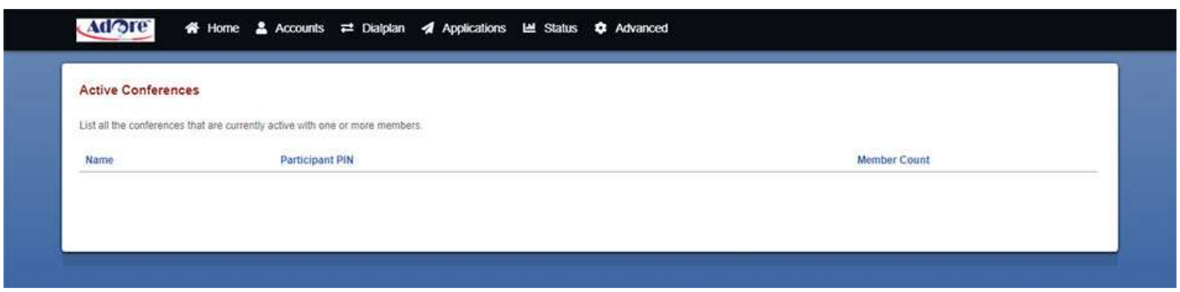
Quiet: How long since the member was talking last.

Has Floor: Who is currently talking.

Mute: Mute a member.

Dead: Make it so the member can't hear what is being said in the conference.

Kick: Kick the member from the conference.



Active Queues

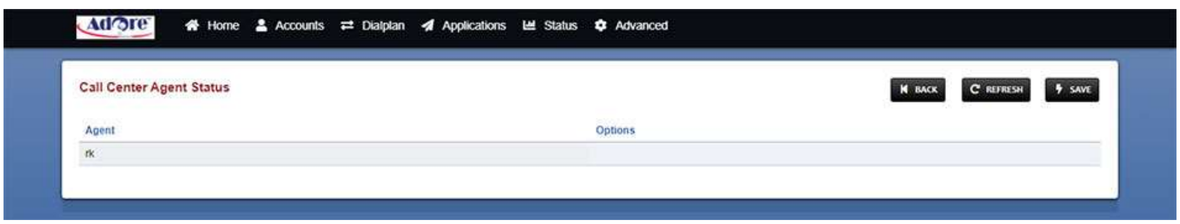
Go to **Status** - **Active Queues**

Queues feature generates a dialplan that uses mod_fifo. FIFO stands for 'first in first out' in other words a queue.

Agent Status

Go to **Status** - **Agent Status**

Here you can get the list all call center agents with the option to change the status of one or more agents.



CDR Statistics

Go to **Status** - **CDR Statistics**

Call Detail Records Statics summarize the call information.

Definitions

- Hours: Specific hour in that day.

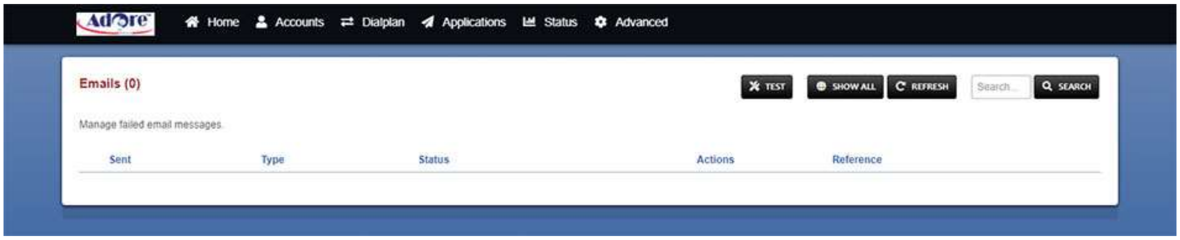
- Date: Specific date in that month.
- Time: Specific time in that day.
- Volume: Number of calls.
- Minutes: Specific number of minutes.
- Calls Per Minute: Specific number of calls per minute.
- Missed: Specific number of missed calls.
- ASR: The answer to seizure ratio. Which is how many calls where answered versus not answered.
- Aloc: ALOC is the average length of call.
- Days: Specific day in that month.

Email Logs

Go to **Status** - **Email Logs**

Manage failed email messages. If for some reason the message doesn't get sent they will sit in a queue. You can view, download or resend each message.

- **Sent**- Date and time last attempt to email was made
- **Type**- If the email was a missed call or voicemail
- **Status**- Status of the email
- **Message**- View, Download or Resend the email
- **Reference**- CDR information
- **Eye icon**- More details about the email
- **X icon**- Deletes the email



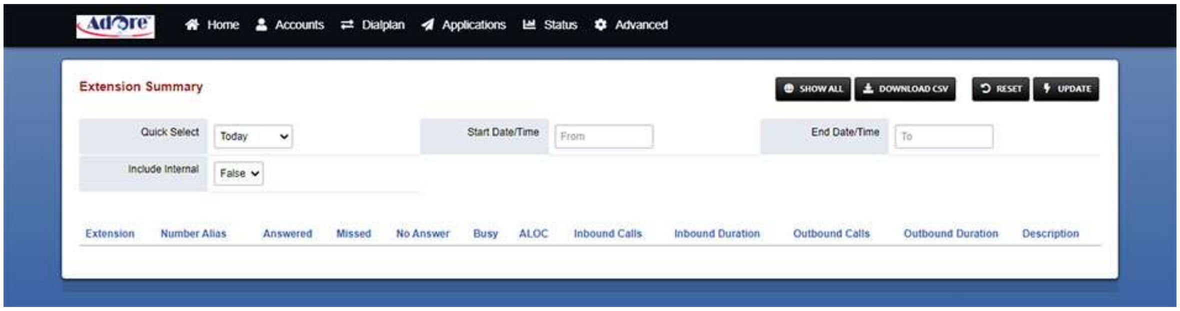
Extension Summary

Go to **Status** - **Extension Summary**

Summary of extension activity per domain such as missed calls, answered calls, no answer, inbound duration, outbound duration, number of outboud calls, number of inbound calls and Average length of Call (ALOC). The summarized information can be downloaded as a CSV file.

Definitions

- **Extension**: The extension number.
- **Number Alias**: Alias name for the extension number.
- **Missed**: Number of missed calls.
- **No Answer**: Number of calls not answered.
- **Busy**: Number of calss not answered wile busy.
- **ALOC**: The average length of call.
- **Inbound Calls**: Number of calls in.
- **Inbound Duration**: Number of call minutes in.
- **Outbound Calls**: Number of calls out.
- **Outbound Duration**: Number of call minutes out.



Registration

Go to **Status** - **Registration**

View the devices that are registered. This will show User, Agent, IP, Port Number, Hostname and Status. You can also UNREGISTER, PROVISION and REBOOT supported devices from here.

9. Advanced

Advanced

In the Advanced menu you will find following options :

- **Access Controls**
- **Databases**
- **Default Settings**
- **Domains**
- **Email Templates**
- **Group Manager**
- **Menu Manager**
- **Number Translations**
- **Settings**
- **SIP Profiles**
- **Variables**

Access Controls

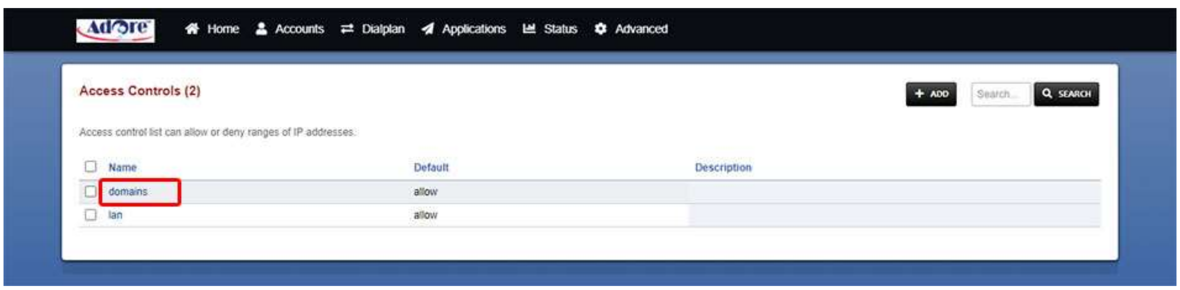
Go to **Advanced** - **Access Controls**

Access control list can allow or deny ranges of IP addresses. There are several purposes for using the ACL.

- The main purpose is for your carriers ip addresses. Add the carrier IP addresses to the CIDR.
- Be careful with what and how you use ACL.

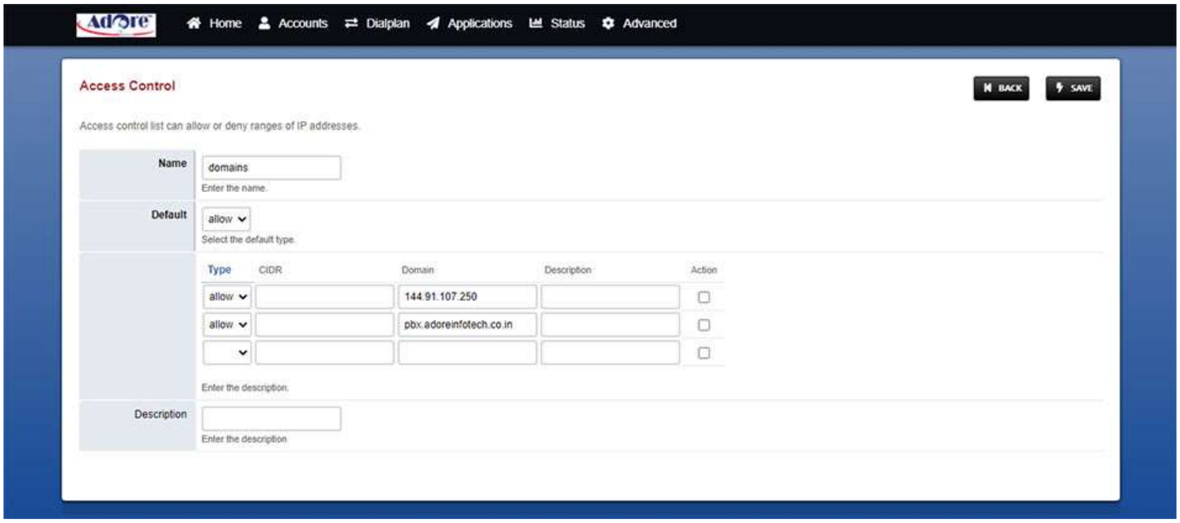
- Most common mistakes result in calls not working between extensions and other undesirable results.
- Be sure to keep Domains access control to default deny.
- Do not put your public ip or phone IP addresses in the domains access control list.
- Don't supply both the domain and the cidr on the same node.
- If adding a single IP address to the CIDR field make sure to add /32 on the end of the IP address.

Click the **domains**.



- Type choose allow
- CIDR enter the Domain IP here
- Domain (Leave Blank, used for advanced scenarios)
- Description (Carrier Name)

and click save



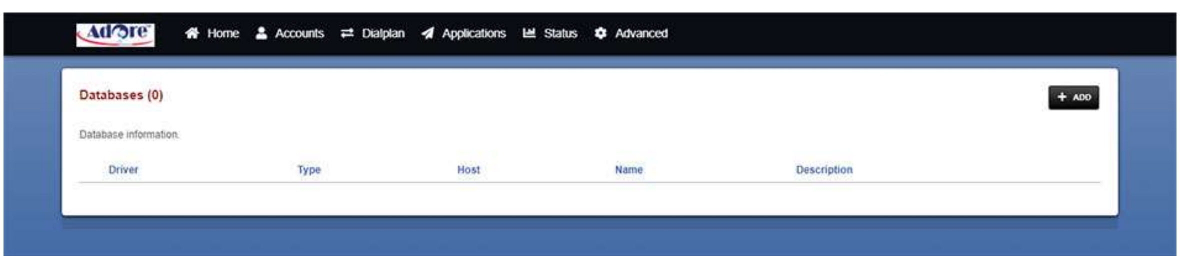
Go to > **Status** > **Sip Status** and click **reloadacl**.

Under **Status** > **log viewer** you should notice the ip added. This can be seen also from command line **fs_cli** by using **reloadacl**

Databases

Go to **Advanced** - **Databases**

Database information. Most Multi Tenant PBX installs use Postgresql for Multi Tenant PBX and SQLite for the switch. This section is for edge case installs.



Default Settings

Go to **Advanced** - **Default Settings**

Default Settings used for all domains. You can set your brand in this section

Default Settings have several different categories.

Cache

Option to use file cache for xml and not memcache.

Subcategory	Type	Value	Enabled	Description
location	text	/var/cache/fusionpbx	True	Location for the file cache.
method	text	file	True	Cache methods file or memcache.

Call Block

Here you can get call block Default setting

Subcategory	Type	Value	Enabled	Description
recent_call_limit	text	50	True	Number of recent calls to show.

Call Center

Defaults for the amount of agent rows for Call Center.

Subcategory	Type	Value	Enabled	Description
agent_add_rows	numeric	5	True	
agent_contact_method	text	user	True	Options: user (default)
agent_edit_rows	numeric	1	True	

CDR

CDR Stat hour limit, call leg, format, limit, http_enabled, archive database, and storage type settings can be set here.

CDR

Subcategory	Type	Value	Enabled	Description
archive_database	boolean	false	False	Enable Dedicated CDR Database Access
archive_database_driver	text	pgsql	False	Archive Database Driver
archive_database_host	text		False	IP/Hostname of Archive Database
archive_database_name	text	fusionpbx	False	Archive Database Name
archive_database_password	text		False	Archive Database Password
archive_database_port	text	5432	False	Archive Database Port
archive_database_username	text		False	Archive Database Username
b_leg	array	outbound	False	
b_leg	array	local	False	
b_leg	array	inbound	False	
cidr	array	127.0.0.1/32	True	Limit allowed range of addresses for CDR ov...
format	text	json	True	
http_enabled	boolean	true	True	
limit	numeric	800	True	
stat_hours_limit	numeric	24	False	
storage	text	db	True	

Conference Center

Here you can get default setting of Conference Center

Conference Center

Subcategory	Type	Value	Enabled	Description
account_code_enabled	boolean	false	True	Request the account ID.
session_enabled	boolean	false	True	Enable Conference Sessions

Contact

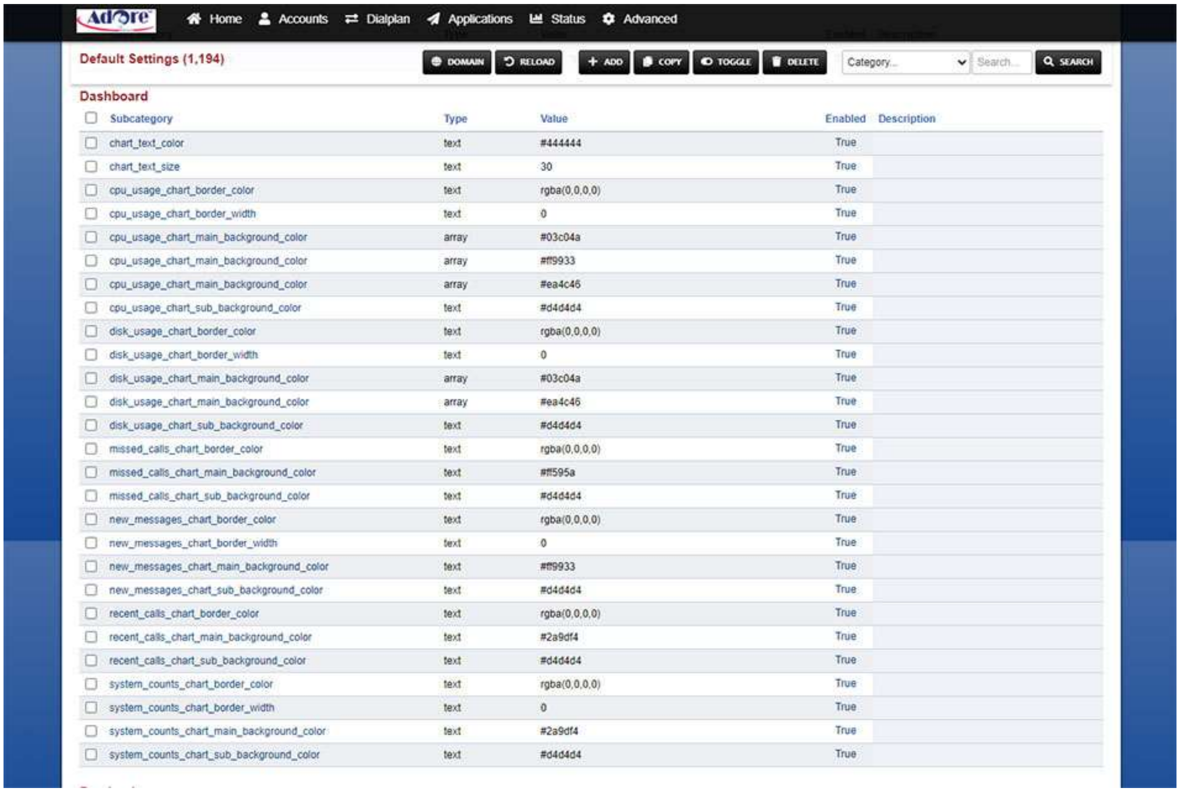
Here you can get default setting of Contact

Contact

Subcategory	Type	Value	Enabled	Description
allowed_attachment_types	text	{"jpg":"image\/jpg","jpeg":"image\/jpg","gif":"image\/gif","p...	True	Define the allowed file attachment extension...

Dashboard

Different user level settings that control what is seen and not seen on the dashboard for each user access level.

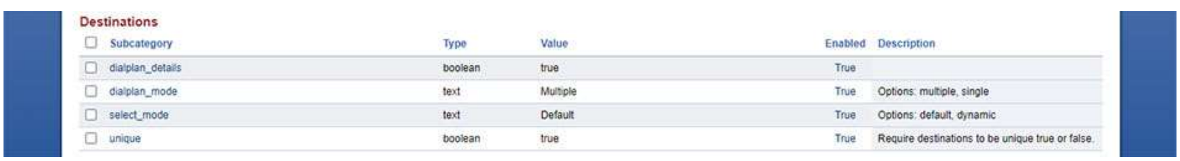


Dashboard

Subcategory	Type	Value	Enabled	Description
chart_text_color	text	#444444	True	
chart_text_size	text	30	True	
cpu_usage_chart_border_color	text	rgba(0,0,0,0)	True	
cpu_usage_chart_border_width	text	0	True	
cpu_usage_chart_main_background_color	array	#03c04a	True	
cpu_usage_chart_main_background_color	array	#ff9933	True	
cpu_usage_chart_main_background_color	array	#ea4c46	True	
cpu_usage_chart_sub_background_color	text	#d4d4d4	True	
disk_usage_chart_border_color	text	rgba(0,0,0,0)	True	
disk_usage_chart_border_width	text	0	True	
disk_usage_chart_main_background_color	array	#03c04a	True	
disk_usage_chart_main_background_color	array	#ea4c46	True	
disk_usage_chart_sub_background_color	text	#d4d4d4	True	
missed_calls_chart_border_color	text	rgba(0,0,0,0)	True	
missed_calls_chart_main_background_color	text	#ff595a	True	
missed_calls_chart_sub_background_color	text	#d4d4d4	True	
new_messages_chart_border_color	text	rgba(0,0,0,0)	True	
new_messages_chart_border_width	text	0	True	
new_messages_chart_main_background_color	text	#ff9933	True	
new_messages_chart_sub_background_color	text	#d4d4d4	True	
recent_calls_chart_border_color	text	rgba(0,0,0,0)	True	
recent_calls_chart_main_background_color	text	#2a9df4	True	
recent_calls_chart_sub_background_color	text	#d4d4d4	True	
system_counts_chart_border_color	text	rgba(0,0,0,0)	True	
system_counts_chart_border_width	text	0	True	
system_counts_chart_main_background_color	text	#2a9df4	True	
system_counts_chart_sub_background_color	text	#d4d4d4	True	

Destinations

Here you get Destinations specific default setting



Destinations

Subcategory	Type	Value	Enabled	Description
dialplan_details	boolean	true	True	
dialplan_mode	text	Multiple	True	Options: multiple, single
select_mode	text	Default	True	Options: default, dynamic
unique	boolean	true	True	Require destinations to be unique true or false.

Domains

Here you get Domain specific default setting

Domain

Subcategory	Type	Value	Enabled	Description
bridge	text	loopback	True	outbound,loopback,lcr
cidr	array		False	
country	code	us	True	
dial_string	text	{sip_invite_domain=${domain_name},leg_timeout=${call...	True	
language	code	en-us	True	
menu	uuid	en-us - default	True	
paging	numeric	100	True	Set the maximum number of records display...
template	name	Default	True	
time_format	text	24-Hour	False	Toggle between 24 hour and 12 hour time fo...
time_zone	name	UTC	True	

Email

Here you configure email settings to receive email notifications of voicemail, missed calls and fax.

Email

Subcategory	Type	Value	Enabled	Description
address_type	text	add_address	True	Options: add_address, add_bcc
method	text	smtp	True	Options: smtp, sendmail, mail, qmail
smtp_auth	text	true	True	
smtp_from	text		True	
smtp_from_name	text		True	
smtp_host	text		True	
smtp_hostname	text		False	Define the local hostname to be used when ...
smtp_password	text		True	
smtp_port	numeric	0	False	use non-default port if enabled and non-zero
smtp_secure	text	tls	True	
smtp_username	text		True	
smtp_validate_certificate	boolean	true	True	set to false to ignore SSL certificate warning...

Extension

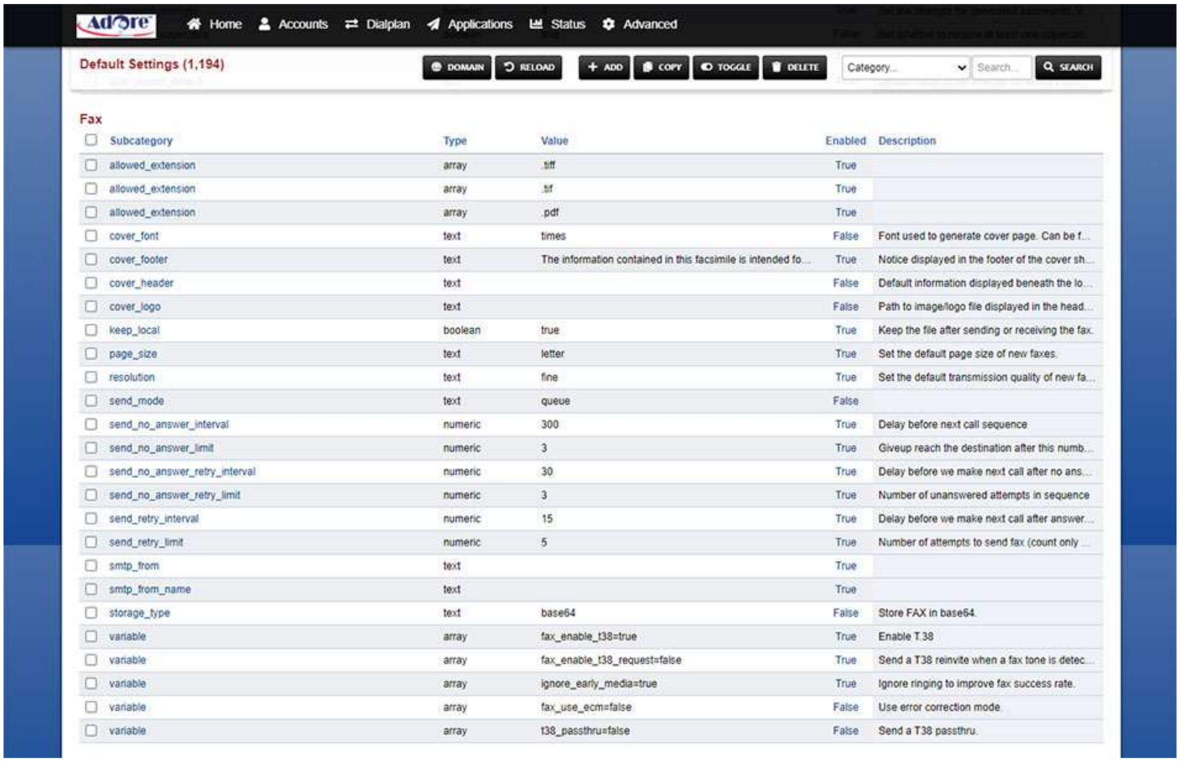
Here you configure Extension default setting

Extension

Subcategory	Type	Value	Enabled	Description
password_length	numeric	20	True	Set the length for generated passwords for e...
password_lowercase	boolean	true	True	Set whether to require at least one lowecase...
password_number	boolean	true	False	Set whether to require at least one number i...
password_special	boolean	true	False	Set whether to require at least one special c...
password_strength	numeric	4	True	Set the strength for generated passwords. V...
password_uppercase	boolean	true	False	Set whether to require at least one uppercas...
session_rotate	boolean	true	True	Whether to regenerate the session ID.
user_record_default	text		True	Default value to set whether to record inbou...

Fax

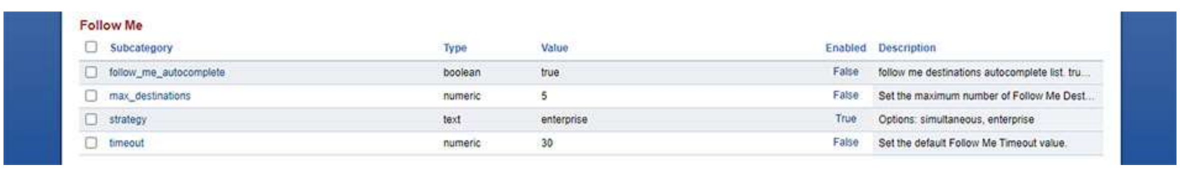
Specific default settings for fax server.



Subcategory	Type	Value	Enabled	Description
allowed_extension	array	.tiff	True	
allowed_extension	array	.tif	True	
allowed_extension	array	.pdf	True	
cover_font	text	times	False	Font used to generate cover page. Can be f...
cover_footer	text	The information contained in this facsimile is intended fo...	True	Notice displayed in the footer of the cover sh...
cover_header	text		False	Default information displayed beneath the lo...
cover_logo	text		False	Path to image/logo file displayed in the head...
keep_local	boolean	true	True	Keep the file after sending or receiving the fax.
page_size	text	letter	True	Set the default page size of new faxes.
resolution	text	fine	True	Set the default transmission quality of new fa...
send_mode	text	queue	False	
send_no_answer_interval	numeric	300	True	Delay before next call sequence
send_no_answer_limit	numeric	3	True	Giveup reach the destination after this numb...
send_no_answer_retry_interval	numeric	30	True	Delay before we make next call after no ans...
send_no_answer_retry_limit	numeric	3	True	Number of unanswered attempts in sequence
send_retry_interval	numeric	15	True	Delay before we make next call after answer...
send_retry_limit	numeric	5	True	Number of attempts to send fax (count only ...
smtp_from	text		True	
smtp_from_name	text		True	
storage_type	text	base64	False	Store FAX in base64.
variable	array	fax_enable_t38=true	True	Enable T.38
variable	array	fax_enable_t38_request=false	True	Send a T38 reinvite when a fax tone is detec...
variable	array	ignore_early_media=true	True	Ignore ringing to improve fax success rate.
variable	array	fax_use_ecm=false	False	Use error correction mode.
variable	array	t38_passthru=false	False	Send a T38 passthru.

Follow Me

Specific default settings for Follow Me



Subcategory	Type	Value	Enabled	Description
follow_me_autocomplete	boolean	true	False	follow me destinations autocomplete list. tru...
max_destinations	numeric	5	False	Set the maximum number of Follow Me Dest...
strategy	text	enterprise	True	Options: simultaneous, enterprise
timeout	numeric	30	False	Set the default Follow Me Timeout value.

IVR Menu

Specific default settings for IVR Menu

Limit

Specific default settings for Limit

Log

Specific default settings for Log

Login

Specific default settings for Login

Operator Panel

Specific default settings for Operator Panel

Provision

In the Provisioning section, there are a few key options that have to be set in order to turn auto provisioning on.

- enabled: Must be enabled and set to value true and enabled True. It is disabled by default.
- http_auth_username: Must be enabled and set to value true and enabled True. It is disabled by default. Be sure to use a strong username.
- http_auth_password: Must be enabled and set to value true and enabled True. It is disabled by default. Be sure to use a strong password.

Recordings

Specific default settings for Recordings

Registration

Specific default settings for Registration

Ring Group

Specific default settings for Ring Group

Server

Specific default settings for Server

Switch

Specific default settings for Switch

Theme

Specific default settings for Theme

Time Conditions

Specific default settings for Time Conditions

Users

Specific default settings for Users

VoiceMail

Specific default settings for Voice Mail

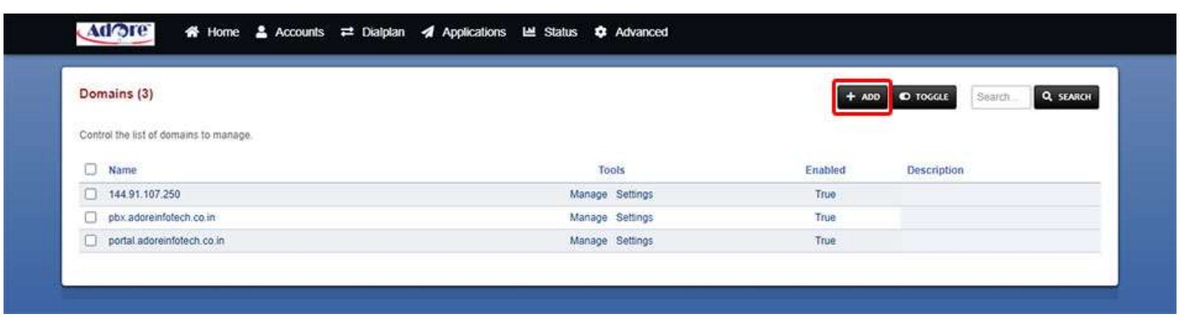
Domains

Go to **Advanced** - **Domains**

Here you can add a domain so that you can reach the specific tenant from the multi-tenant domain side menu to configure and allow secure administration from the world wide web.

Click **Add** button to add new domain (new tenant) in system.

There are several reasons to create a domain (tenant). One reason would be to organize customers and so customers have a unique login ie superadmin@pbx.adoreinfotech.co.in or superadmin@subdomain.domain.co.in as the username.



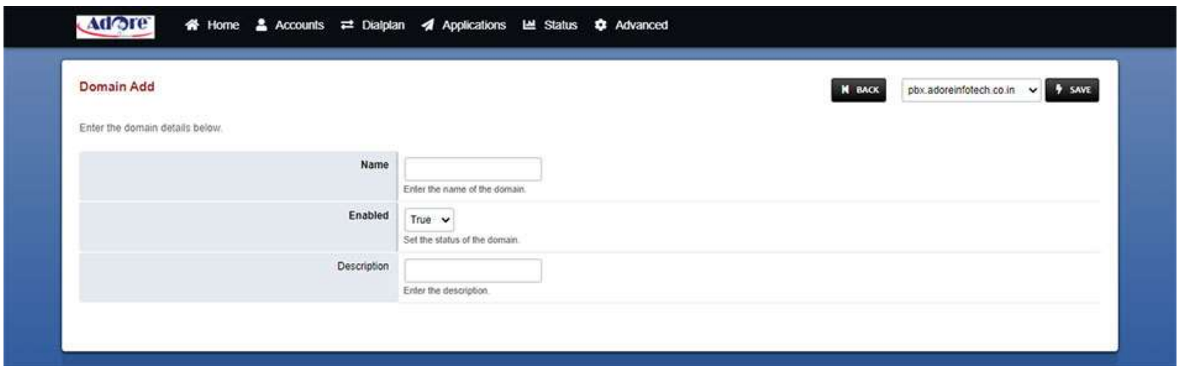
Here you to enter domain info. (Please make sure you have add “**A record**” on your domain hosting account with your domain/subdomain)

Name : example.com or pbx.example.com

Enabled : TRUE or False

Description : you can add as per your reference.

and click SAVE button to add New Domain

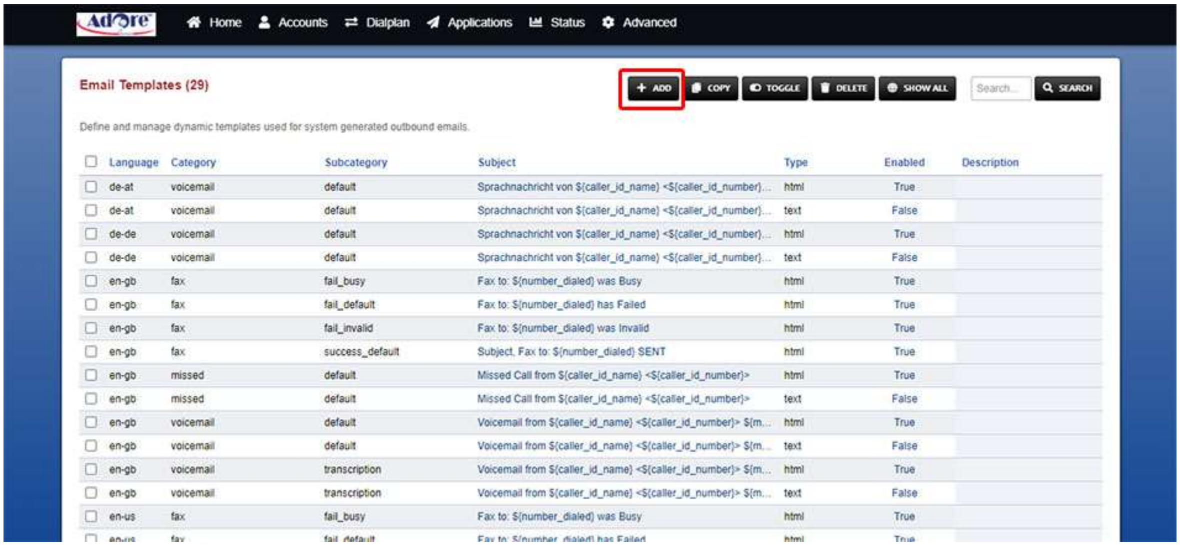


Email Templates

Go to **Advanced** - **Email Templates**

Define and manage dynamic templates used for system generated outbound emails.

Here you can **Add** , **Edit**, **Delete** Email Templates



Group Manager

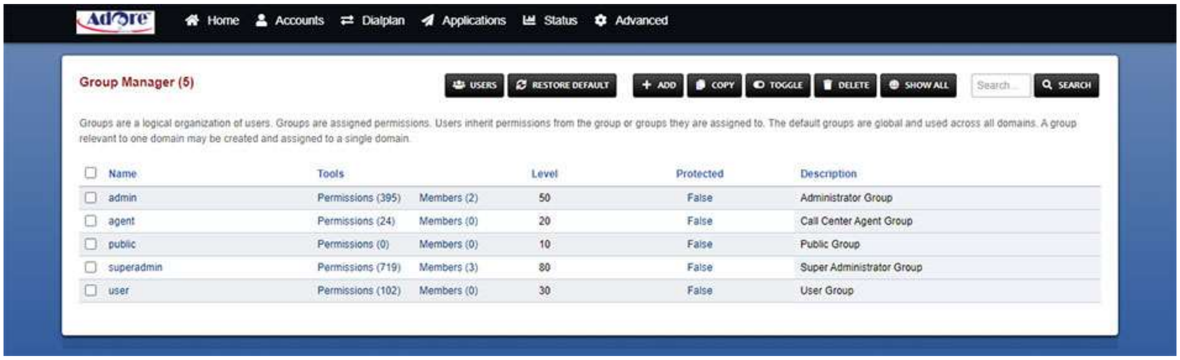
Go to **Advanced** - **Group Manager**

Permit access levels to different group of users. The group permissions allow customizing permissions for existing groups or custom groups.

Here you can **Create**, **edit**, **remove users**.

superadmin- the global administrator
admin- the domain administrator
users- the group for regular users

NOTE : Group- assign the user to a group. Be wise as to who has access to what.

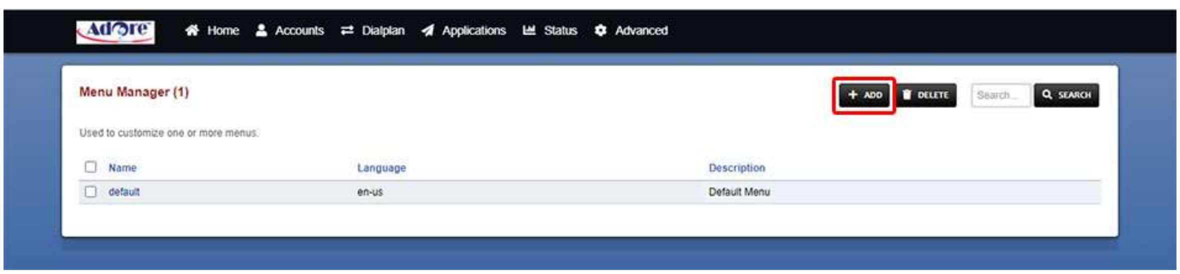


Menu Manager

Go to **Advanced** - **Menu Manager**

Here you can used to customize one or more menus.

Click **Add** button to new menu if you required, if not don't add any new Menu in system.



Number Translations

Go to **Advanced** - **Number Translations**

Here you can translate numbers from the original number to a new number using regular expressions.

Activating mod-translate:

Install the package “freeswitch-mod-translate”. If using Debian Package then use the following command “apt install freeswitch-mod-translate”

Configure the module to your likes via the GUI: Advanced -> Number Translations.

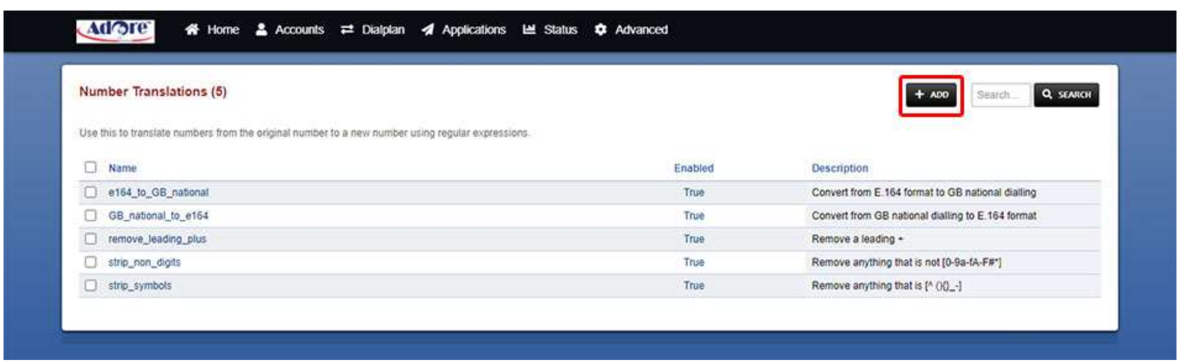
Activate the module in PBX Advanced -> Modules in the Applications section

To use mod-translate to modify inbound calls before they hit the dialplan the following setting for the SIP-profile must be modified:

dialplan “XML” -> dialplan “Translate,XML”

With 1.8.x it is now possible to specify the translation profile to be used: dialplan “XML” -> dialplan “Translate:my_profile1,XML”

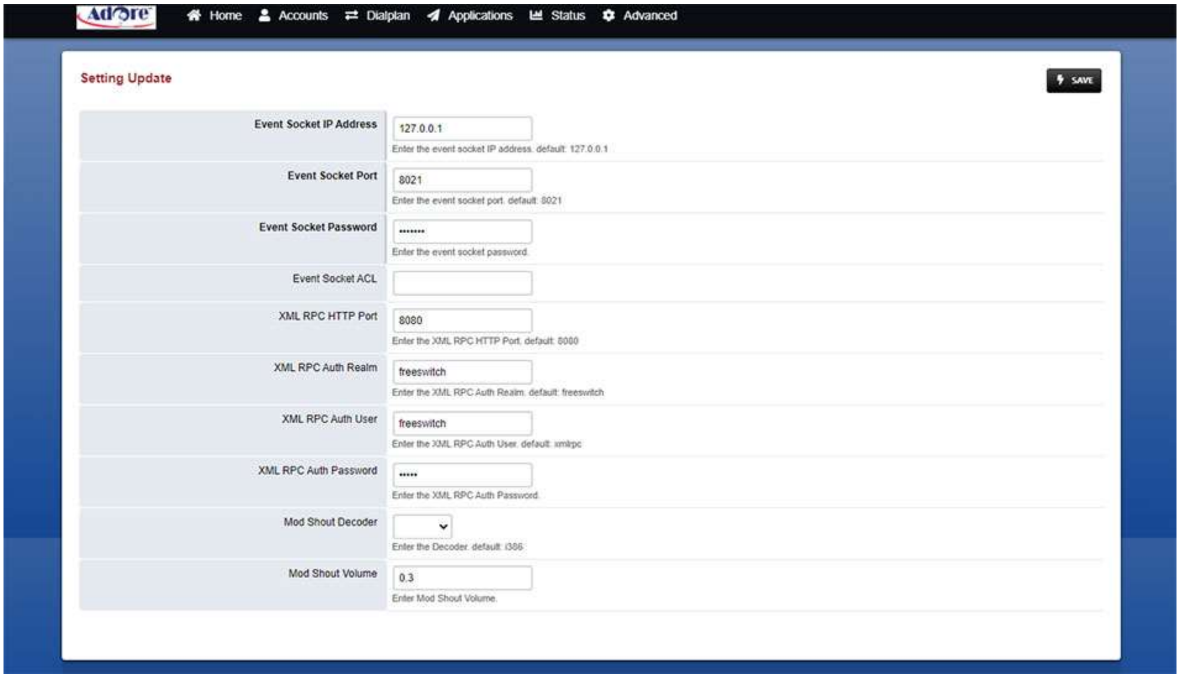
To activate this setting, you must flush cache once and then restart or rescan each SIP-profile



Settings

Go to **Advanced** - **Settings**

Switch settings for event socket ip address, event socket port, event socket password, xml rpc http port, xml rpc auth realm, xml rpc auth user, xml rpc auth password, mod shout decoder, and mod shout volume.



SIP Profiles

Go to **Advanced** - **Settings**

Internal

Internal sip profiles (port 5060/5061) require registration or access controls cidr range to allow the IP address in without SIP authentication. Once the access controls are setup correctly, the carrier will be allowed to send calls to the internal profile.

External

External sip profiles (port 5080-5081) allow anonymous connection to FusionPBX and is optional. External profile is optional when freewitch has a public ip address. Can be useful when setting behind nat. Being anonymous doesn’t mean totally open due to the inbound routes call conditions.(call filtering)

Internal ipv6

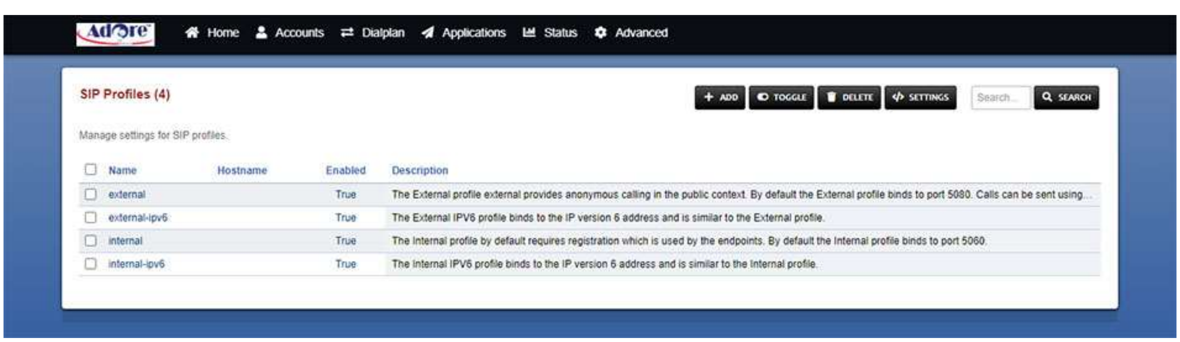
Internal ipv6 sip profiles (port 5060/5061) require registration or access controls cidr range to allow the IP address in without SIP authentication. Once the access controls are setup correctly, the carrier will be allowed to send calls to the internal ipv6 profile.

- If you don’t have ipv6 then the ipv6 profiles should be disabled.
- Be sure to stop the profile before disabling it. To disable goto Advanced > SIP Profiles and click the pencil edit icon to the right of the profile you want to disable. From the dropdown box select enabled to false.

External ipv6

External ipv6 sip profiles (port 5080-5081) allow anonymous connection to PBX and is optional.

- If you don’t have ipv6 then the ipv6 profiles should be disabled.
- Be sure to stop the profile before disabling it. To disable goto Advanced > SIP Profiles and click the pencil edit icon to the right of the profile you want to disable. From the dropdown box select enabled to false.



Variables

Go to **Advanced** - **Variables**

Here you can define preprocessor switch variables. Don't touch any setting without requirements.

Adore Home Accounts Dialplan Applications Status Advanced

Switch Variables (76)

ADD COPY TOGGLE DELETE Search... SEARCH

Define preprocessor variables here. A switch restart is required for changes to take effect.

Codecs

Name	Value	Hostname	Enabled	Description
global_codec_prefs	G7221@32000h,G7221@16000h,G722,PCMU,PCMA		True	
media_mix_inbound_outbound_codecs	true		True	
outbound_codec_prefs	PCMU,PCMA		True	

Defaults

Name	Value	Hostname	Enabled	Description
domain_uuid	086e44bc-1723-4512-97c9-ad27f0bbab33		True	
ajax_refresh_rate	3000		True	
call_debug	false		True	
console_loglevel	info		True	
default_areacode	208		True	
default_country	US		True	
default_countrycode	1		True	
default_dialect	us		True	
default_exitcode	011		True	
default_language	en		True	
default_voice	callie		True	
record_ext	wav		True	
ringback	$${us-ring}		True	
sit	%(274,0,913.8);%(274,0,1370.6);%(380,0,1776.7)		True	
sleep	0		True	
transfer_ringback	$${us-ring}		True	
use_profile	internal		True	

IP Address

Name	Value	Hostname	Enabled	Description
external_rtp_ip	$${local_ip_v4}		True	
external_sip_ip	$${local_ip_v4}		True	

Music on Hold

Name	Value	Hostname	Enabled	Description
hold_music	local_stream://default		True	

Ringtones

Name	Value	Hostname	Enabled	Description
au-ring	%(400,200,383,417);%(400,2000,383,417)		True	
be-ring	%(1000,3000,425)		True	
bong-ring	v		True	
ca-ring	%(2000,4000,440,480)		True	
cn-ring	%(1000,4000,450)		True	
cy-ring	%(1500,3000,425)		True	
cz-ring	%(1000,4000,425)		True	
de-ring	%(1000,4000,425)		True	
dk-ring	%(1000,4000,425)		True	
dz-ring	%(1500,3500,425)		True	
eg-ring	%(2000,1000,475,375)		True	
fi-ring	%(1000,4000,425)		True	
fr-ring	%(1500,3500,440)		True	
hk-ring	%(400,200,440,480);%(400,3000,440,480)		True	
hu-ring	%(1250,3750,425)		True	
il-ring	%(1000,3000,400)		True	
in-ring	%(400,200,425,375);%(400,2000,425,375)		True	
it-ring	%(1000,4000,425)		True	
jp-ring	%(1000,2000,420,380)		True	
ko-ring	%(1000,2000,440,480)		True	
pk-ring	%(1000,2000,400)		True	
pl-ring	%(1000,4000,425)		True	
pt-ring	%(1000,5000,400)		True	
ro-ring	%(1850,4150,475,425)		True	
rs-ring	%(1000,4000,425)		True	
ru-ring	%(800,3200,425)		True	
sa-ring	%(1200,4600,425)		True	
tr-ring	%(2000,4000,450)		True	
uk-ring	%(400,200,400,450);%(400,2000,400,450)		True	
us-ring	%(2000,4000,440,480)		True	

Security

Name	Value	Hostname	Enabled	Description
disable_system_api_commands	true		True	
disable_system_app_commands	true		True	

SIP

Name	Value	Hostname	Enabled	Description
hangup_on_subscriber_absent	true		False	
sip_tls_version	tlsv1,tlsv1.1,tlsv1.2		True	
unroll_loops	true		True	

SIP Profile: External