COMXCHANGE 14 SETUP GUIDE

COMMON SETUP TASKS

360 NETWORKS

Setting the New Standard

360 NETWORKS
6116 Darlin Drive Dane, WI 53529
V.1.17.2020

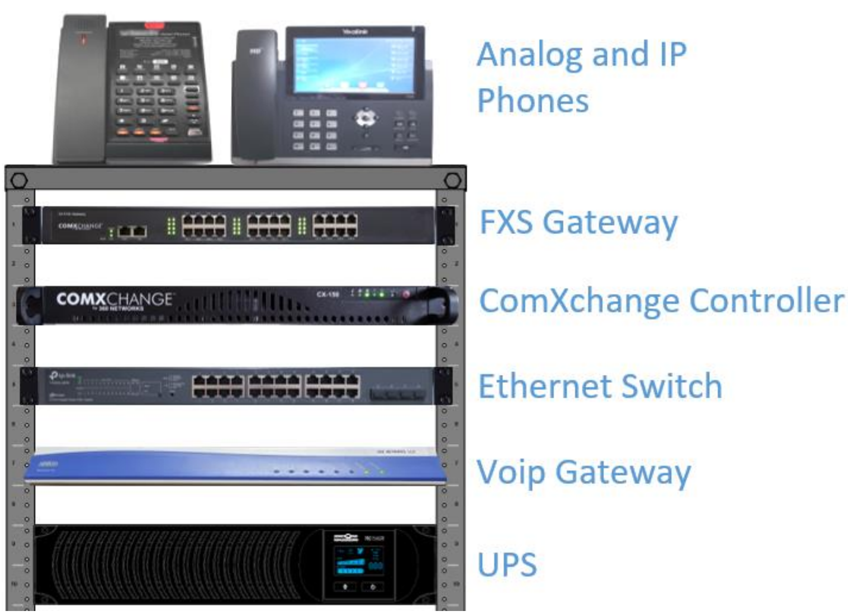
Contents

ComXchange 14 Setup Guide

Introduction

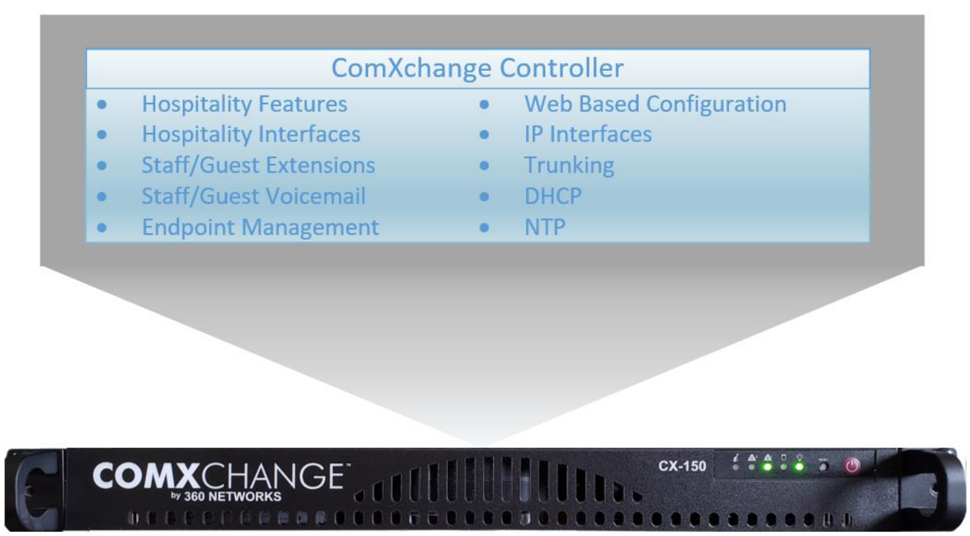
ComXchange is an all-inclusive communications server that includes a PBX, Call Accounting, and Voicemail developed specifically for the hospitality industry. ComXchange leverages the power of hardware and software open standards. This lowers costs and improves quality, providing the flexibility to provide customers with systems suited for large or small properties. ComXchange is also carrier agnostic and allows for multiple carrier options such as T-1/PRI, SIP and POTS providing great flexibility.

Hardware Architecture Overview:



A ComXchange Setup will typically consist of these items:

ComXchange Controller:



The ComXchange controller is a custom designed server platform. It runs headless as a server and therefore will not require a keyboard, mouse, or monitor. While there is a monitor port, typically nothing is being transmitted on it. The Controller hosts a Voice over IP (VoIP) PBX system that uses the internet standard for real-time voice communication SIP (Session Initiation Protocol) to deliver voice communications to other SIP devices.

The ComXchange Controller provides:

- Phone Extensions for Staff and Guests
- Voicemail for Staff and Guests
- Endpoint management for IP phones
- DHCP Services that dynamically provides IP addresses for IP Phones
- NTP Services that provides time synchronization for Network Devices
- Hospitality Features and Interfaces.

ComXchange Ports

The cabling of the ComXchange controller and where the USB drives are plugged in is very important for services to function properly. Below is an overview of the ports that are typically used in a ComXchange setup.

- PMS Serial Interface – Communicates between the PMS and the PMS Interface on the Controller for populating guest names on extensions upon check-in, wake up calls, and room status changes.
- USB Storage Drive – A USB device in the lower left USB port will be used to store backups. The other USB ports are available for USB to serial adapters if necessary.
- LAN Ethernet Port – The Local Area Network (LAN) port will be connected to the local phone network.
- WAN Ethernet Port – The Wide Area Network (WAN) port will be connected to a port with access to the internet for VPN connection, NTP, or a direct SIP connection.
- CAS Serial Port – Communicates between the PMS and the Call Accounting Server on the Controller.

Ethernet Switch

Switches connect the components of the ComXchange phone system together via ethernet. These are a vital part of the system and must meet specific requirements which include:

- Management Interface
- VLAN Tagging and Trunking
- Quality of Service (QoS) for Voice Traffic Prioritization
- Port mirroring for Troubleshooting
- Gigabit backplane
- PoE Capabilities

24-FXS Gateway

The 24-FXS analog gateway provides connectivity to all industry standard analog phones. Phones can be connected directly to the front with a cable or more commonly by an Amphenol cable which is tied into a punch down block. The gateway provides dial tone, battery current, and ring voltage for an analog phone and will translate signals between analog and digital allowing communication to a VoIP server.

VoIP Gateway

The Voice Over Internet Protocol (VoIP) gateway provides a connection between the ComXchange Controller and standard Plain Old Telephone Lines (POTS) or the T1/ Primary Rate Interface (PRI) trunks provided by the telephone carrier. The VoIP gateway provides a sip trunk connection to the ComXchange server over the ethernet port and a connection to the carrier trunk over the T1 0/2 port for a PRI - T1 connection or the voice port via an Amphenol cable for POTS Lines.

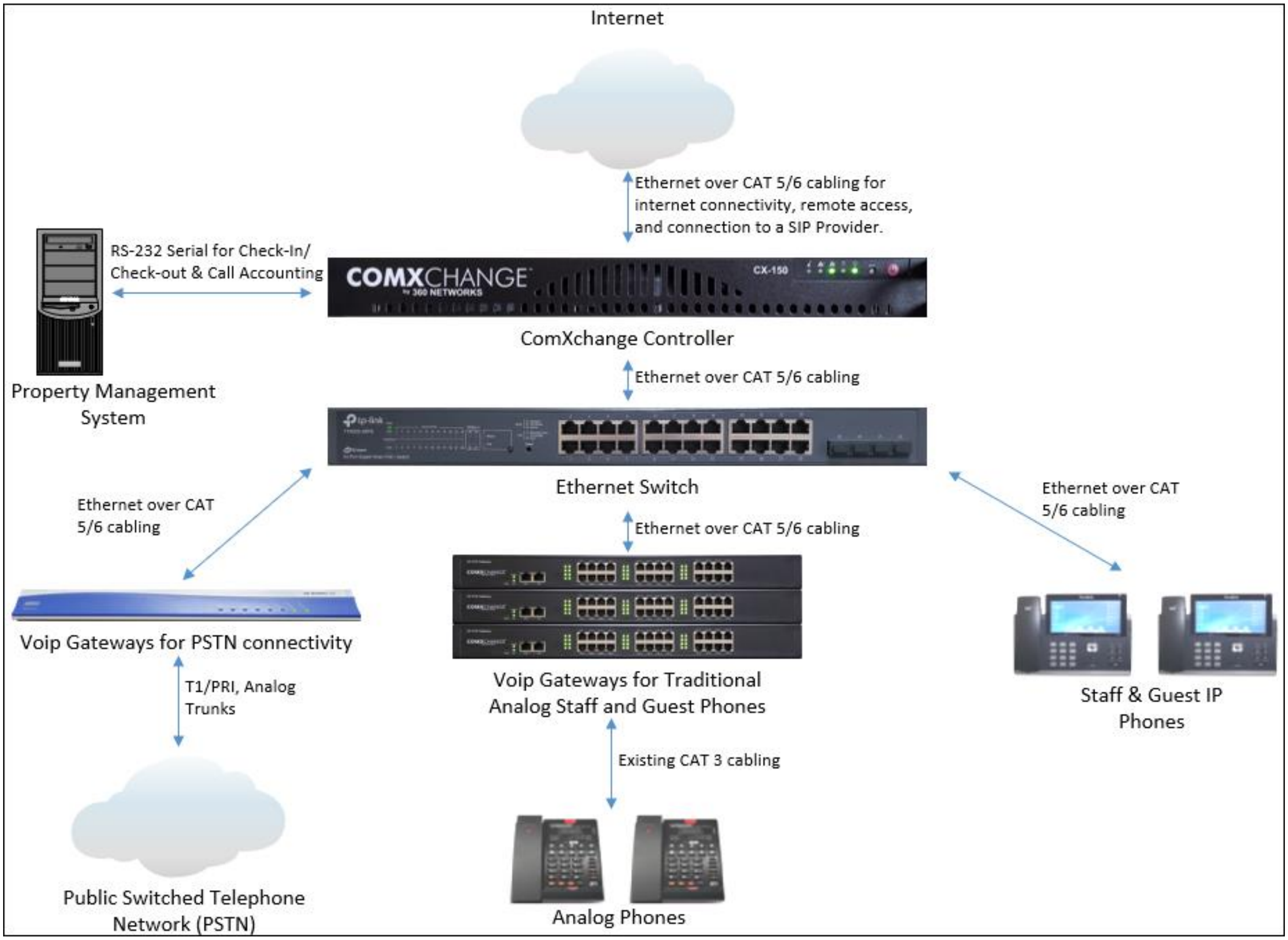
Phones

- IP Phones – Internet Protocol (IP) Phones make and receive calls over an IP Network, providing excellent voice quality. IP phones also offer advanced functionality for programming buttons and xml applications
- Analog Phones – provide basic calling functions mainly used in guest rooms and can have preprogrammed buttons that may need to match feature codes in the ComXchange Controller.

UPS

An Uninterruptable Power Supply (UPS) provides battery backup to the system in a power outage. All system components should be connected to a UPS as noted in the warranty information of the ComXchange System

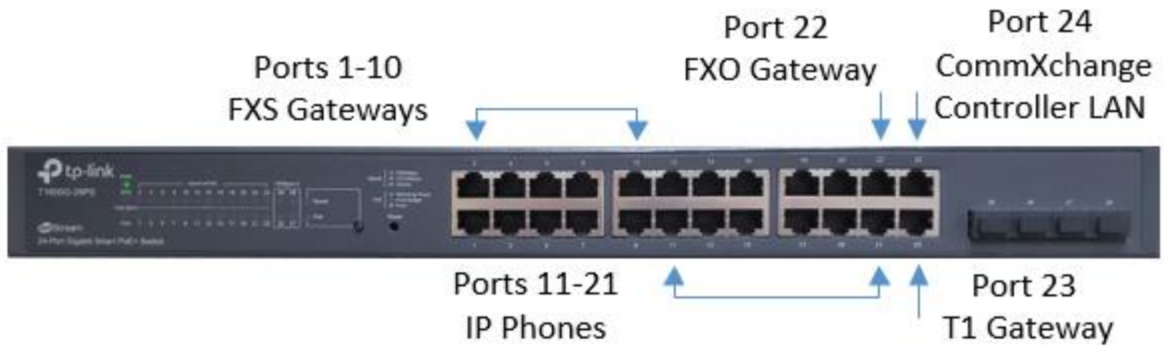
System Architecture



Network Guidelines

Because ComXchange is an IP based system, the network it resides on is very important. All voice equipment must be on its own network. This means using a separate physical network or at a minimum a VLAN containing only the voice equipment.

Example Switch Setup



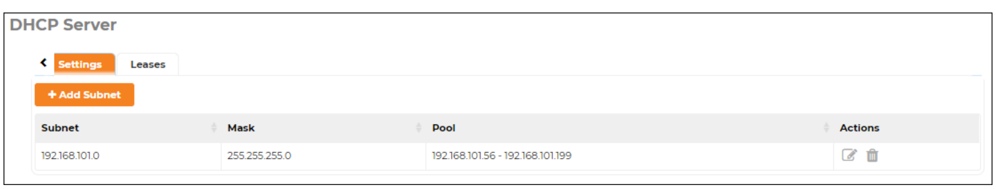
Default IP Address Scheme

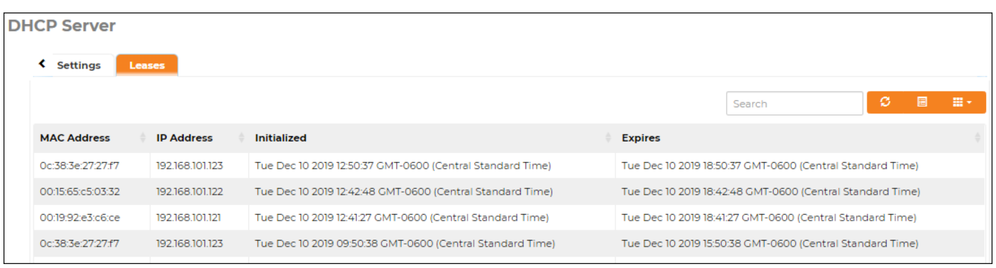
IP Address	Device(s) (if installed)
192.168.101.1	Default Router Gateway.
192.168.101.2	ComXchange Staff SIP Server/Management
192.168.101.5 – 192.168.101.10	Ethernet switches. Start at 192.168.101.5 and increment if necessary.
192.168.101.11	Reserved for VMWare Management.
192.168.101.20	T1/PRI Gateway
192.168.101.21 – 192.168.101.50	Analog station gateways (FXS). Start at 192.168.101.21 and increment if necessary.
192.168.101.51 – 192.168.101.55	Analog trunk gateways (FXO) Start at 192.168.101.51 and increment if necessary.
192.168.101.56 – 192.168.101.200	IP Phones via DHCP
192.168.101.201 – 192.168.101.210	VPN IP Addresses for remote connectivity.

DHCP Server

The ComXchange Controller has a built in DHCP server that will hand out addresses to devices with DHCP enabled. The DHCP server has Option 160 enabled pointing to the controller which allows a phone to automatically find the provisioning server where the configuration is stored.

Admin > System Settings - DHCP





ComXchange Configuration

Basic System Setup

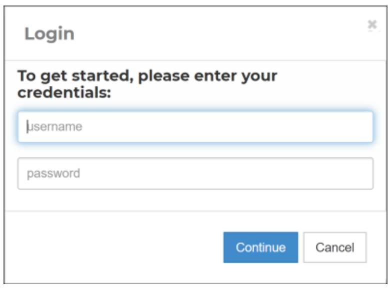
Once the equipment is cabled and turned on, connect a PC or a laptop to the voice network with an ethernet cable. ComXchange is now ready to be configured in the ComXchange PBX system web GUI (Graphical User Interface). Begin by logging into the Controllers web GUI, change the dealer password, verify/change the network settings of the controller, and make any changes to the SIP settings for your setup.

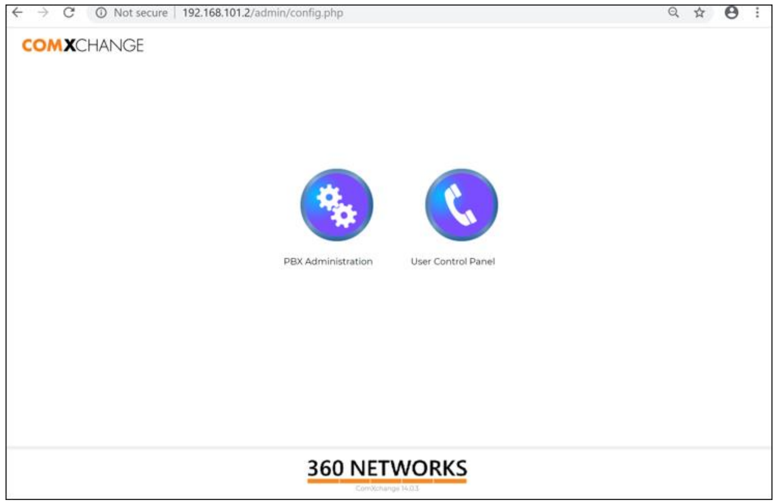
Note: The ComXchange Controller must connect to the license server over the internet for activation. If the Controller is not able to reach the license server, the ability to make configuration changes may fail.

Logging In

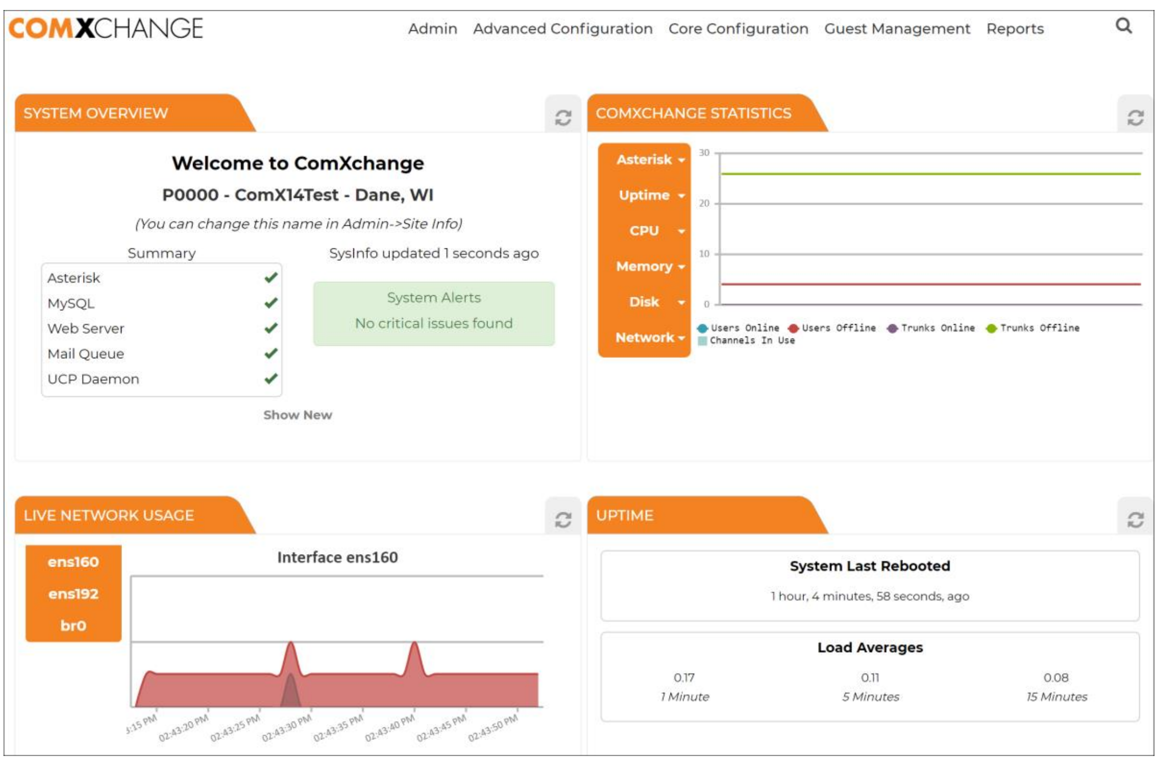
The web interface is accessible at http://192.168.101.2 by default.

Click on PBX Administration and login with the default username/password of dealer/dealer.





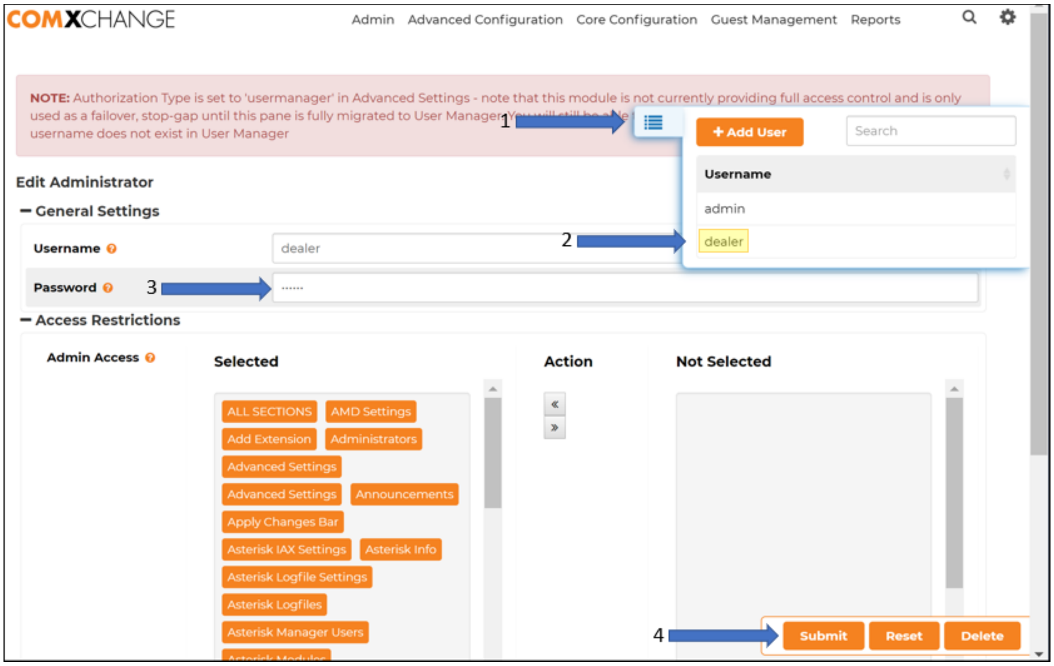
The controller will open the ComXchange status page and from here, you will have access to the configuration menu buttons across the top.



Changing the Dealer Password

Navigate to Admin > Administrators. This opens the Add administrator page. On the right-hand side is a flyout menu with a list of current Users.

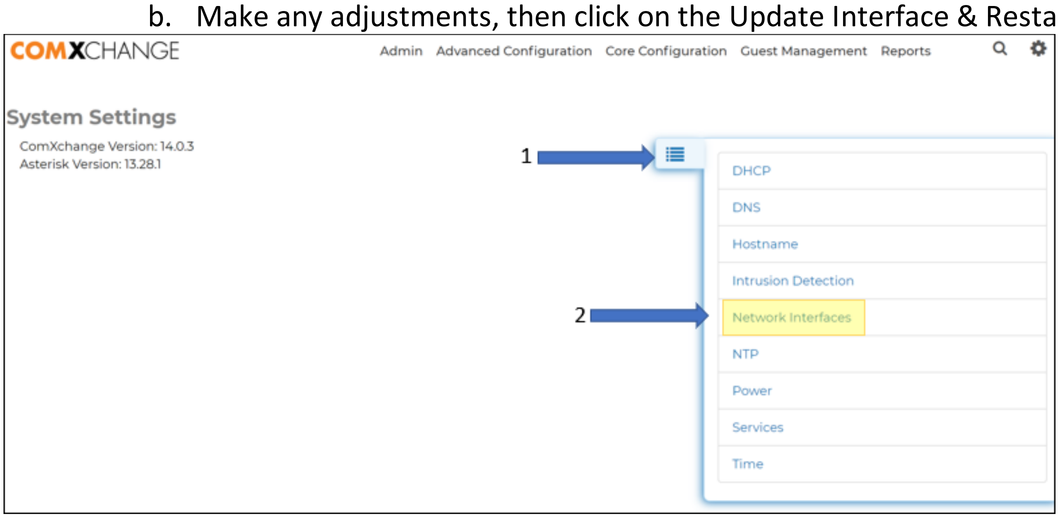
1. Click on the flyout menu
2. Click on dealer
3. Enter the new password
4. Click Submit Changes

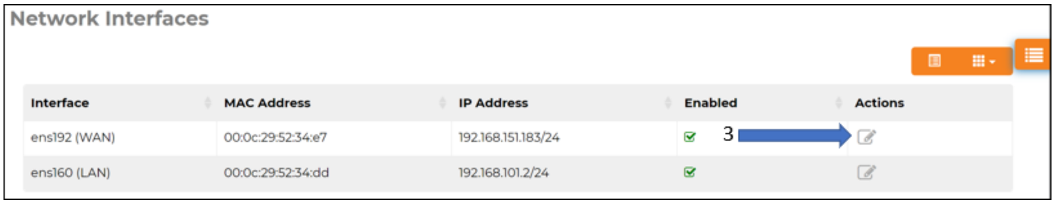


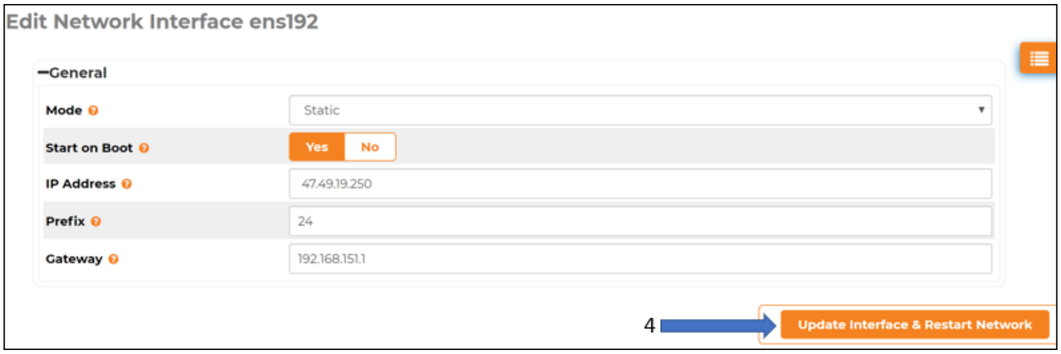
Network Settings

Navigate to Admin > System Settings – Network Interfaces

1. Click on Network Settings in the fly away menu on the right side of the page
2. Ens160 (LAN) will be your local network with the default IP address
3. Ens192 (WAN) is where you will set your static ISP IP address
    a. Click on the edit Actions icon for ens192.
    b. Make any adjustments, then click on the Update Interface & Restart Network button



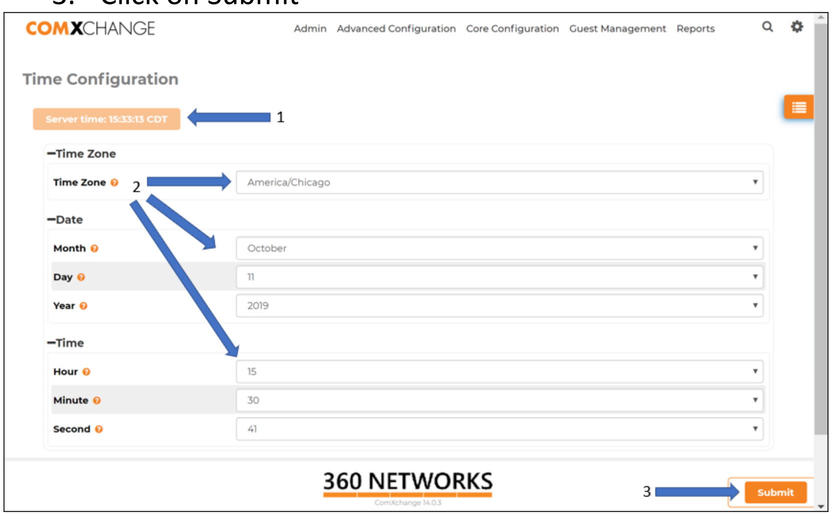




Time Configuration

Navigate to Admin > System Settings - Time

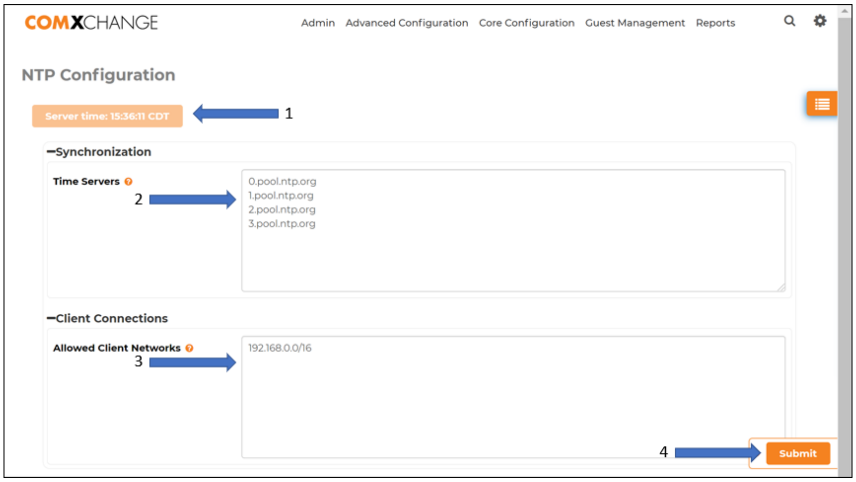
1. Verify the Server time
2. Adjust Time Zone, Date, and Time if necessary
3. Click on Submit



NTP Configuration

Navigate to Admin > System Settings - NTP

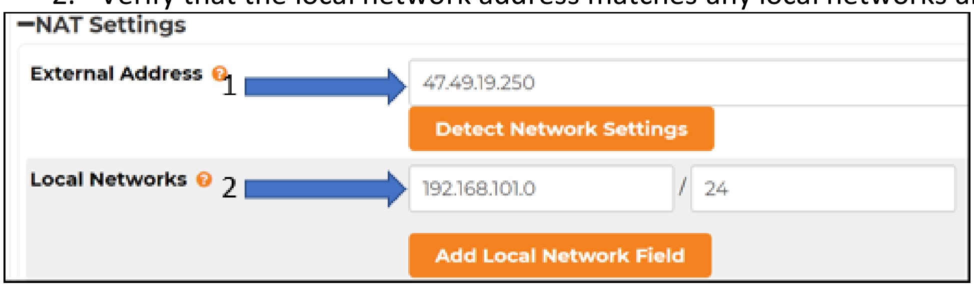
1. Verify that the Server time
2. Verify Time Servers
3. Make any adjustments to Client networks
4. Click on Submit



SIP Settings

Navigate to Admin > SIP Settings - Nat Settings

1. Verify that the External IP reflects the static IP or Network Address Translation (NAT) IP address that can reach the public provider.
2. Verify that the local network address matches any local networks and adjust if necessary.

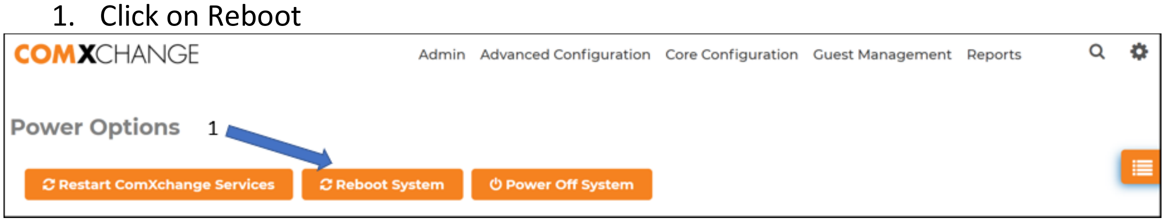


Power Options

After making any changes to the networking be sure to submit and apply the changes to the configuration and as a best practice reboot the server.

Navigate to Admin > System Settings - Power

1. Click on Reboot



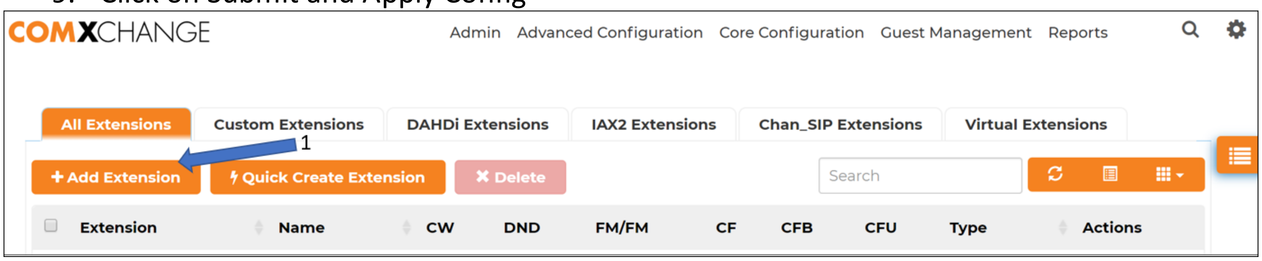
Core Configuration

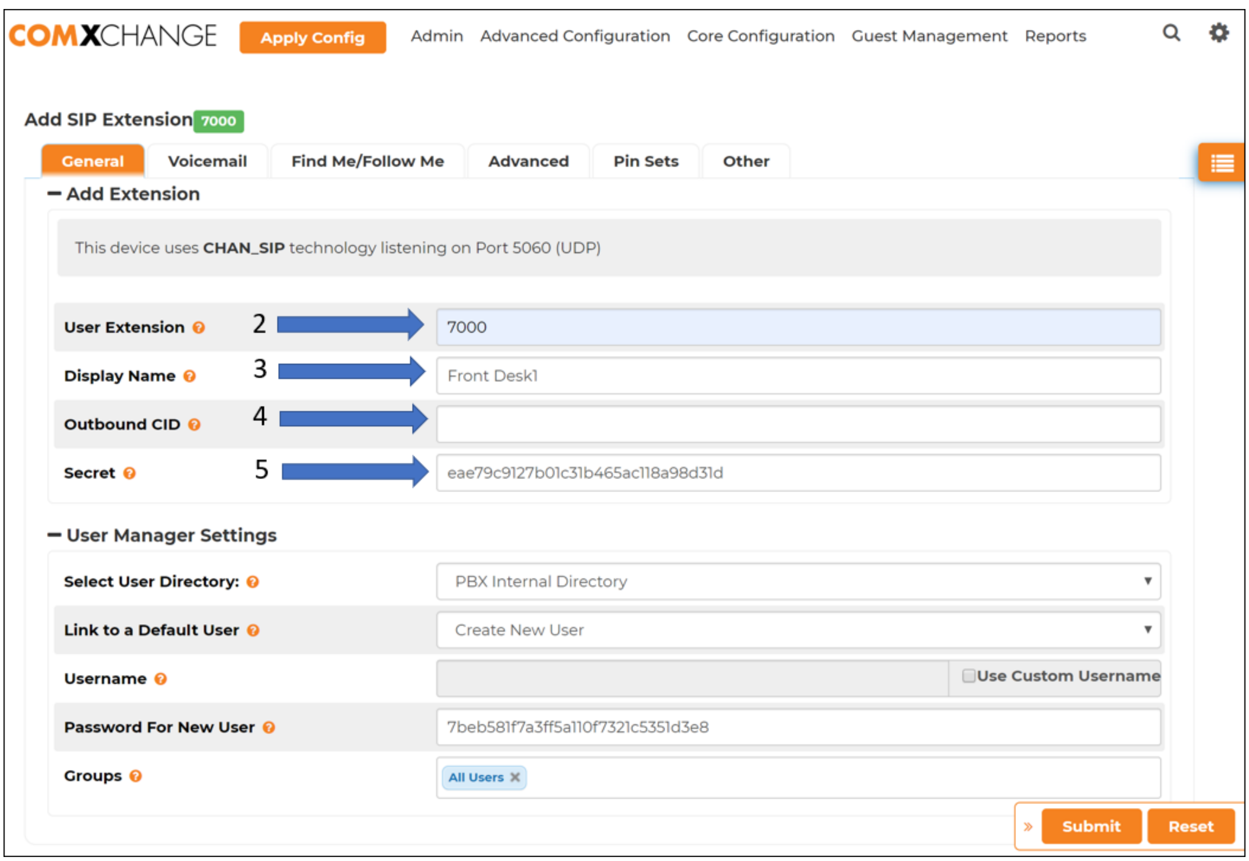
Setup internal calling. To set up and test internal calling we will create Extensions with Voicemail, add a Ring Group that includes the new extensions, add a miscellaneous application for the operator then set up phones that will use these extensions.

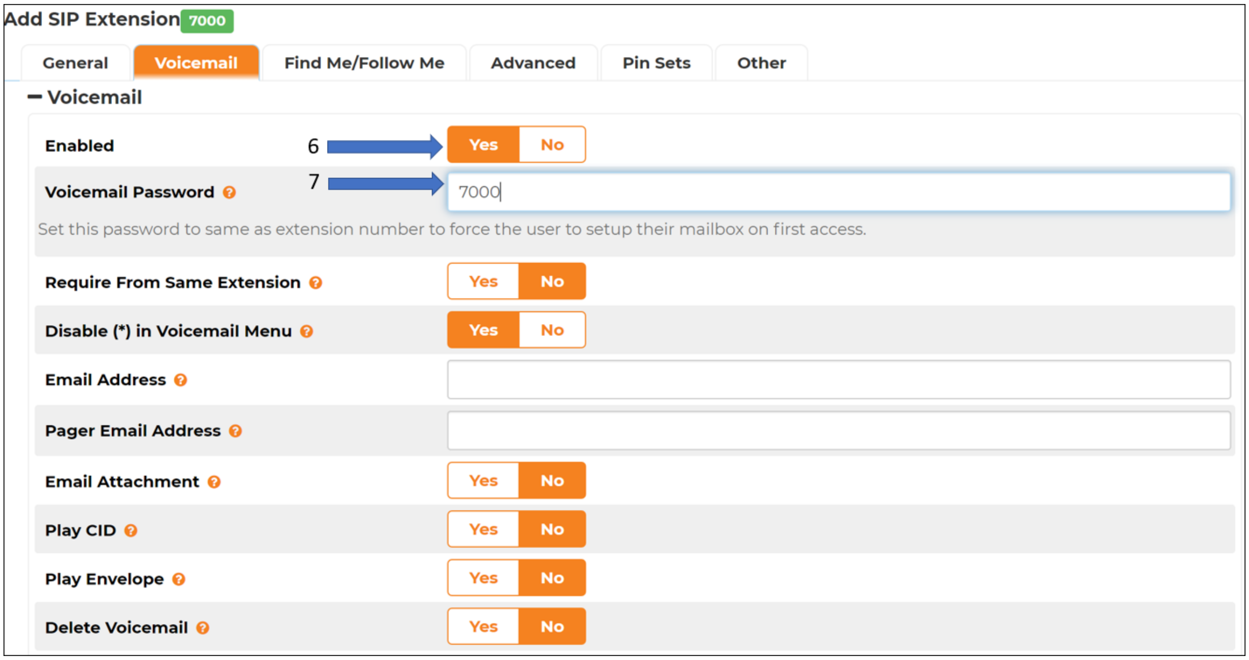
Building Extensions for Staff

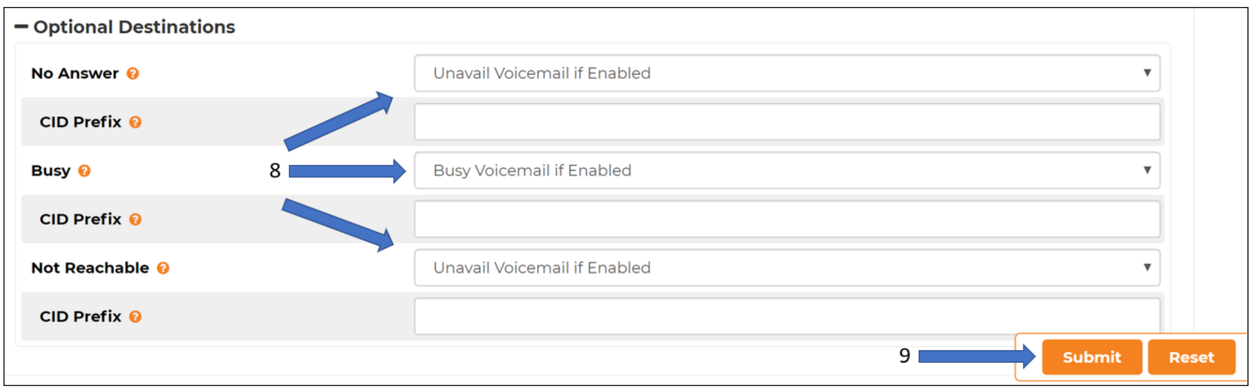
Navigate to Core Configuration > Extensions. Here you can create Extensions 7000 and 7001

1. Click on the Add Extension button
2. Fill in the User Extension
3. Fill in Display name
4. You can fill in an optional outbound Caller ID (number)
5. Make note of the secret field for use setting up a phone later
6. Under the Voicemail Tab click on yes to enable Voicemail
7. Enter a voicemail password
8. Under the Advanced Tab - Optional Destinations other voicemail options can be configured
9. Click on Submit and Apply Config

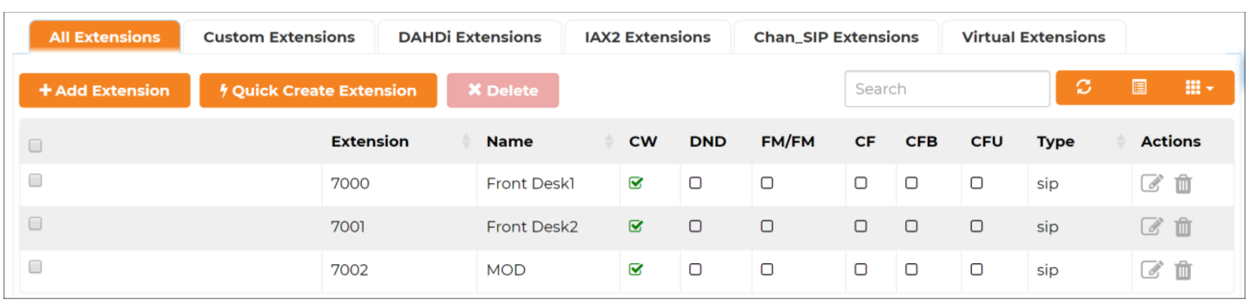








Add Extensions 7001 (Front Desk2) which will be added to a ring group in the next section and 7002 (MOD) to be used as a back of house phone for later testing.

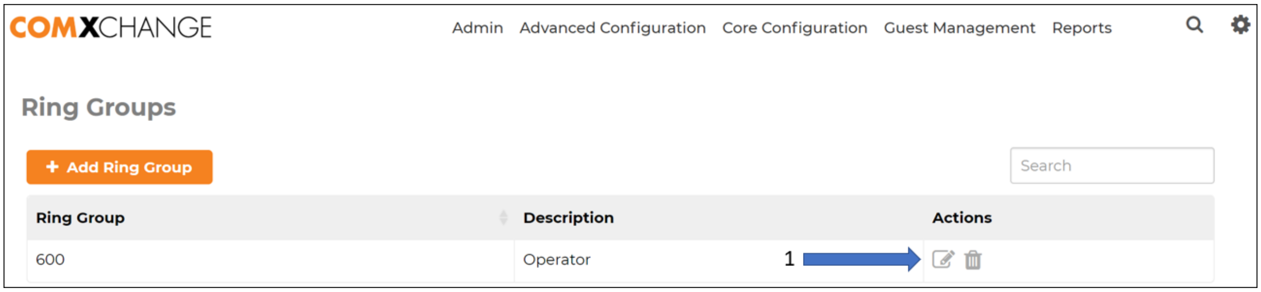


Ring Groups/Operator Routing

The Operator Ring Group will be available by default. Any extensions that are in this group will ring when 600 is dialed. Note: It is possible to add an external number to a ring group by typing the number followed by #.

Navigate to Core Configuration > Ring Groups.

1. Click on the Ring Group 600 edit Actions button on the right side
2. Add Extensions 7000 and 7001 in the User Quick Select Dropdown
3. Click on Submit at the bottom of the page and Apply

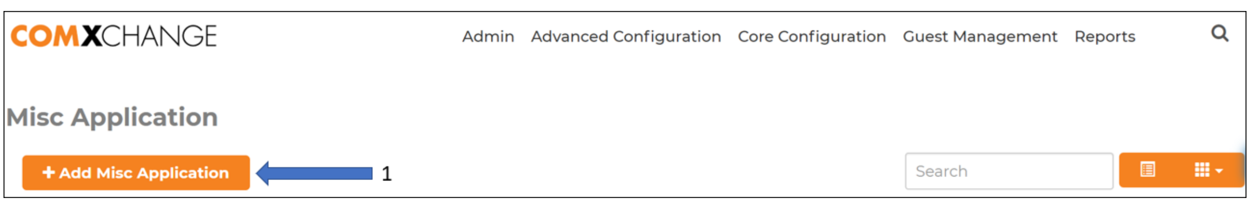


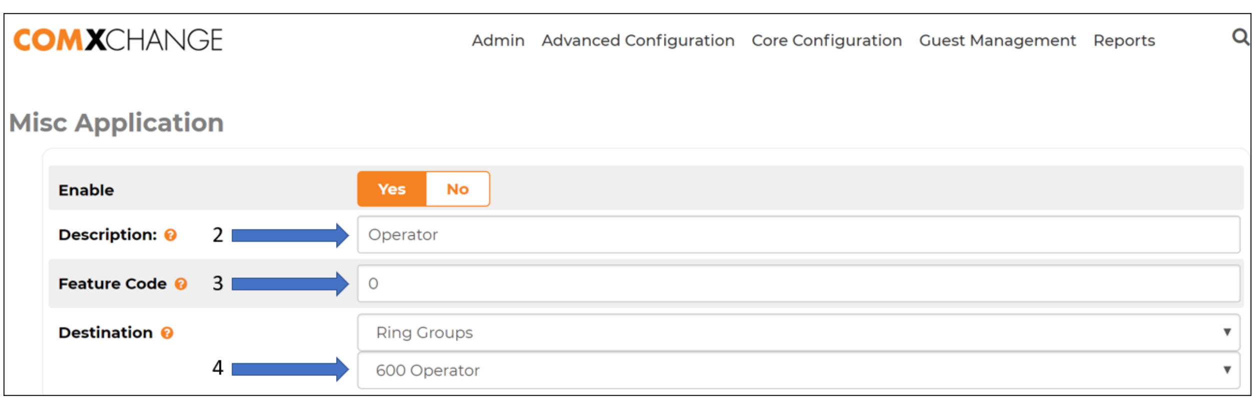
Miscellaneous Application

Creating a miscellaneous application will create a feature code that when dialed can point to destinations available in ComXchange, such as a Ring Group. Below shows the process of adding a Misc Application called Operator that allows 0 to dial the Front desk Ring Group

Navigate to Advanced Configuration > Misc Applications.

1. Click Add Misc Application
2. Fill in a Description “Operator”
3. Fill in Feature Code Number “0”
4. Choose a Destination
5. Submit Changes and Apply Configuration





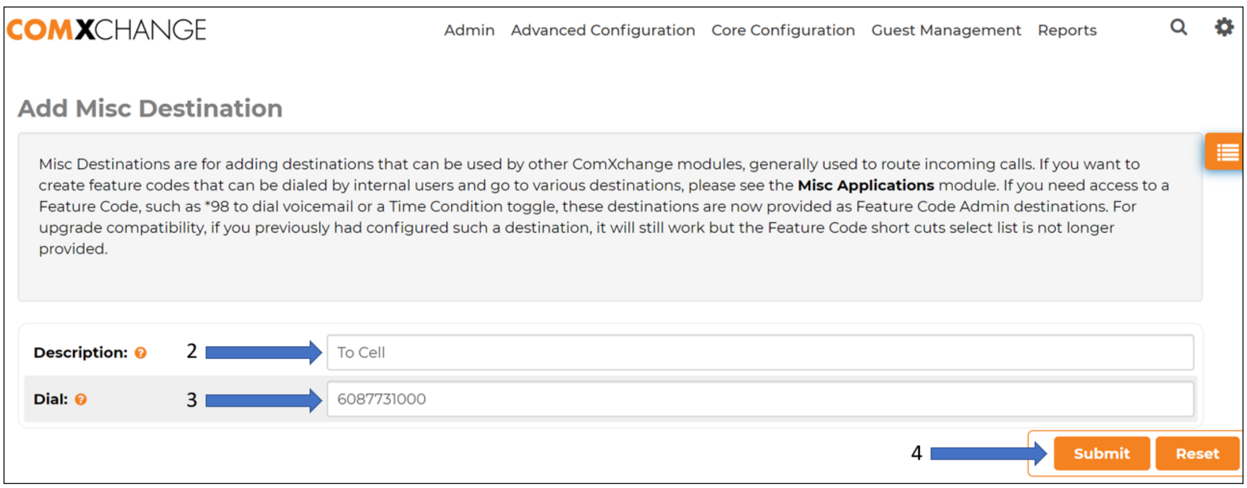
Miscellaneous Destination

A Miscellaneous Destination will allow calls to be routed from another module by simulating a number being dialed, such as a feature code or an external number to reach an outside reservations office or a mobile phone. These are often used in combination with a Misc Application, IVR's, or Speed Dials.

Create a Destination to an External Phone Number.

Navigate to Advanced Configuration > Misc Destinations.

1. Click on the Add Misc Destination button
2. Add a Description
3. In the Dial Field insert numbers exactly as you would dial them from a phone
4. Submit and Apply



This can now be used as a destination for an IVR, an Inbound Route, or a Speed Dial.

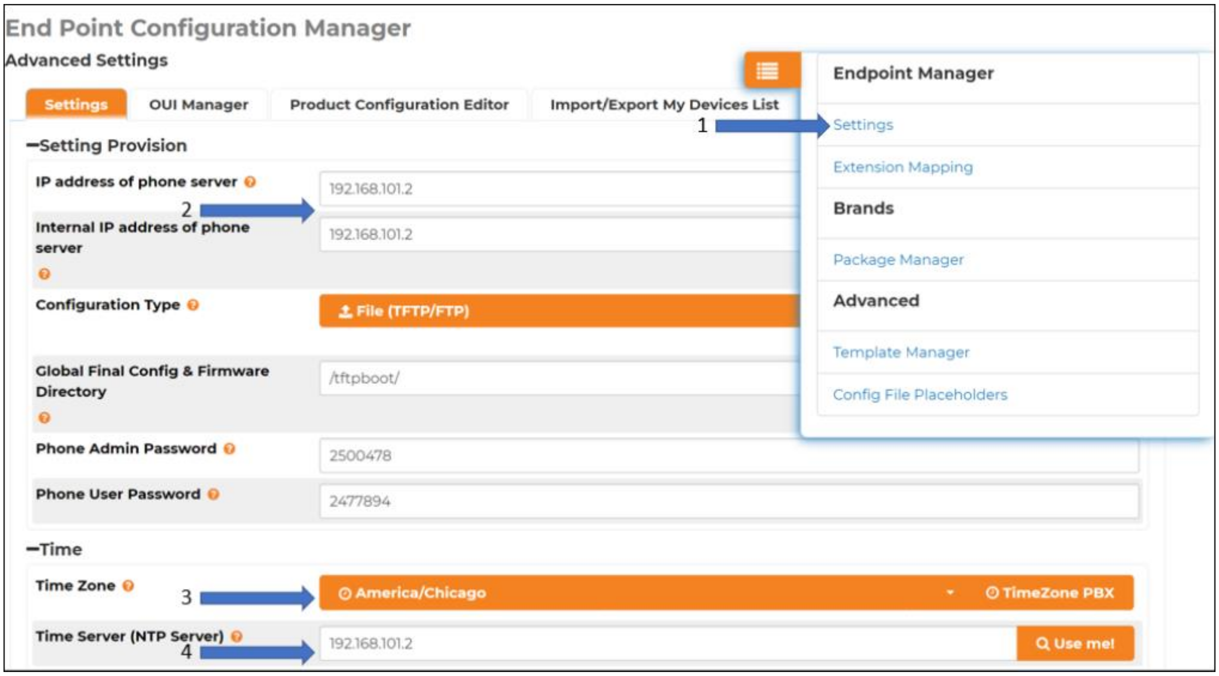
Basic IP Phone Configuration

IP phones can be setup in two ways. Phones can be provisioned by the server where a configuration is created from a template for a phone's MAC Address which will automatically download from the network or they can be (hardcoded) which involves configuring the individual phone via the individual phone's web GUI.

Endpoint Provisioning

Navigate to Core Configuration > End Point Manager

1. Click on Settings in the Endpoint Manager fly away menu on the right
2. Verify the IP address of the phone server
3. Verify the Time Zone
4. Verify the NTP server information (can point to ComXchange)

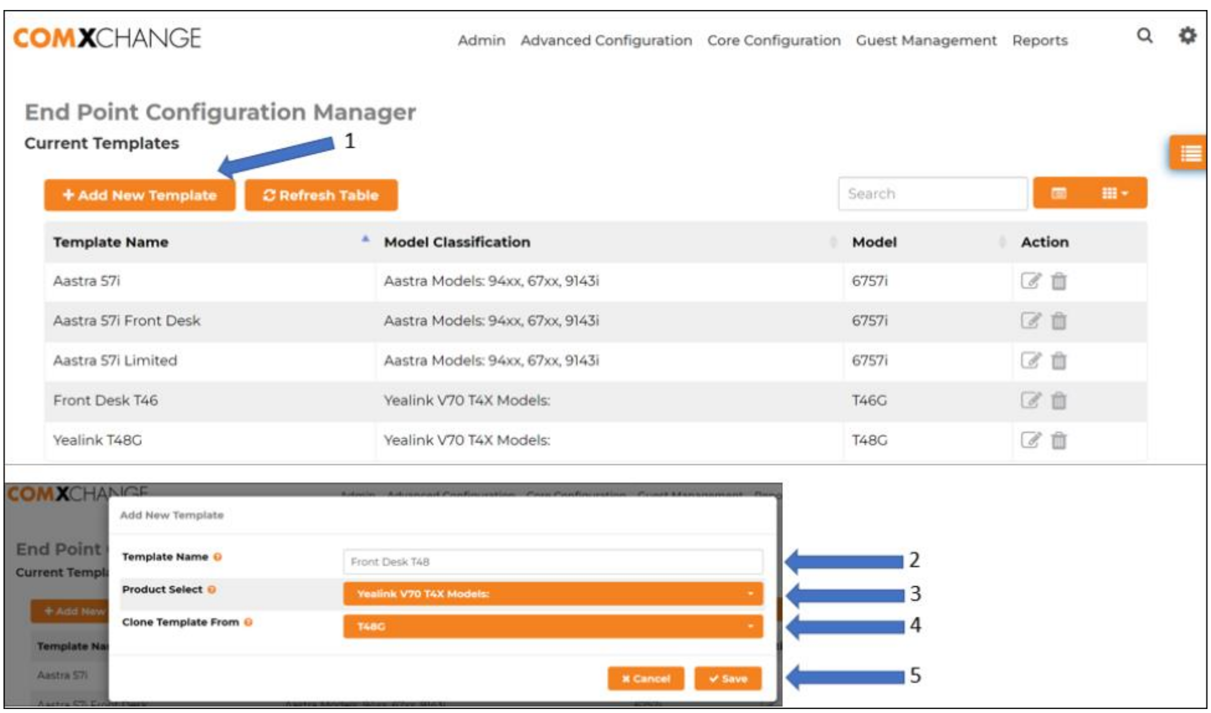


End Point Templates

Templates are an easy way to deploy phones that will have similar functionality such as a front desk phone. Phones based on these templates will have the same buttons on the display, dial plan information, and any other specific information added to the template. Phones will typically check for config changes on bootup.

Navigate to Core Configuration > End Point Manager – Template Manager (fly away menu)

1. Click on Add New Template
2. Fill in the Template Name
3. Choose from the type of phone from the drop-down menus
4. Choose a template that you would like to clone from
5. Click on Save



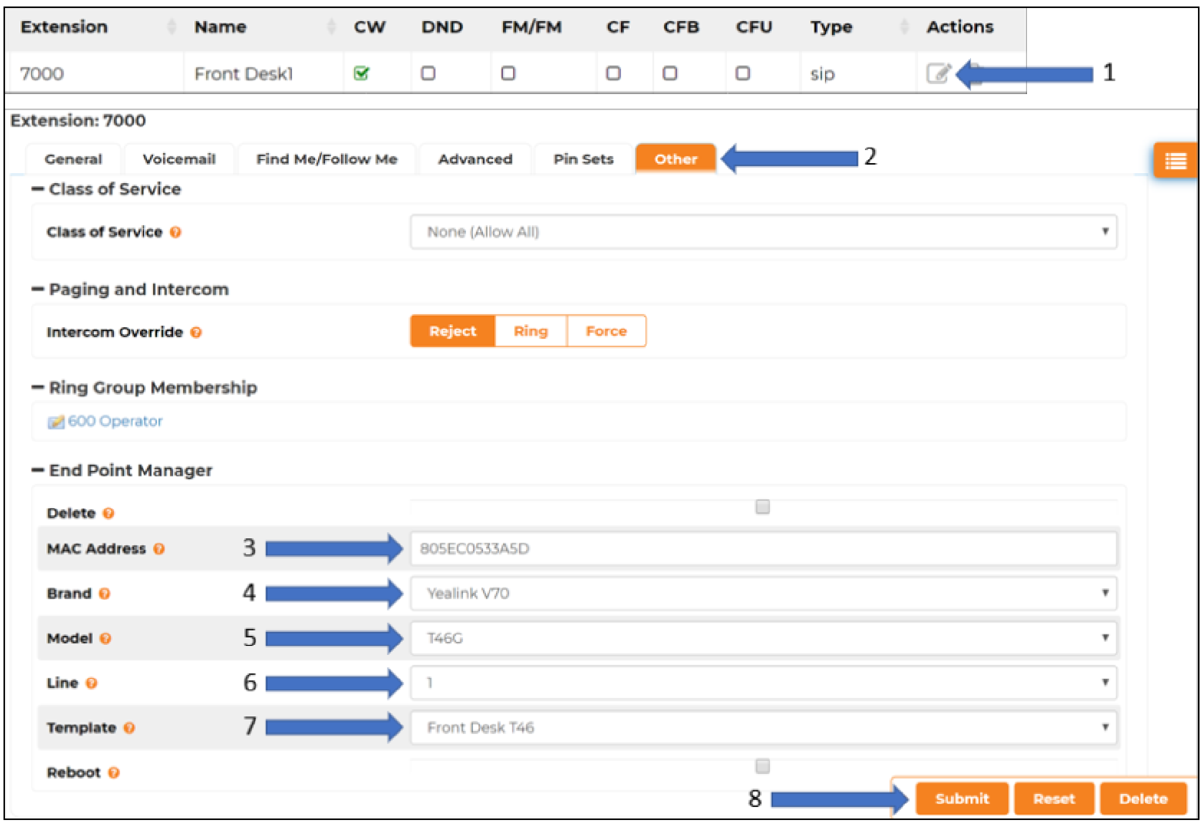
Mapping an Extension to a Phone Device

You can map an extension to a phone device from ComXchange in one of two places. In the Extension settings or in the End Point Dashboard. Either way you will need the MAC Address from the phone which is usually located on the back of the phone.

Extension Settings

Navigate to Core Configuration > Extensions

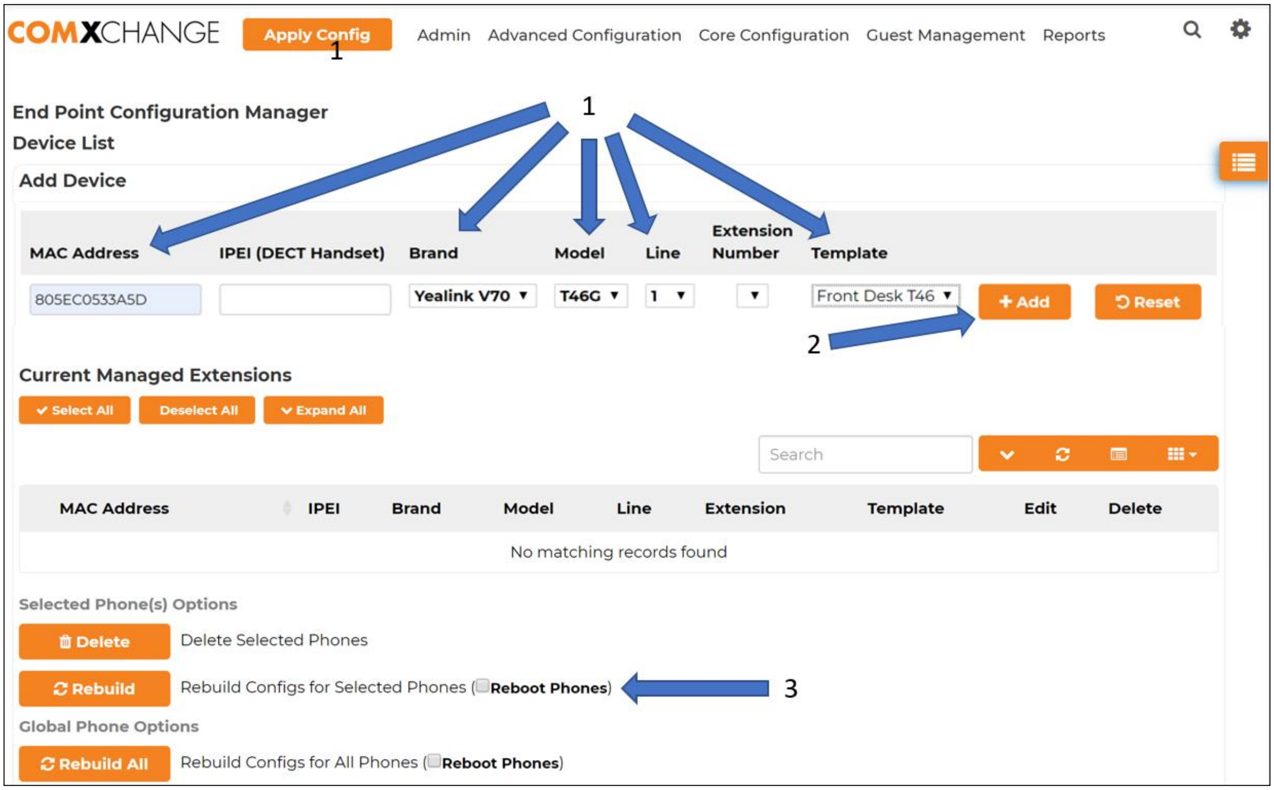
1. Click on the edit Actions Button of the extension you are adding a device to
2. Click on the Other tab
3. Fill in the MAC Address
4. Choose the Brand of phone
5. Choose the Model of phone
6. Choose the line for the extension number (default is 1)
7. Choose the Template to apply to the phones configuration
8. Click on Submit and Apply



End Point Dashboard

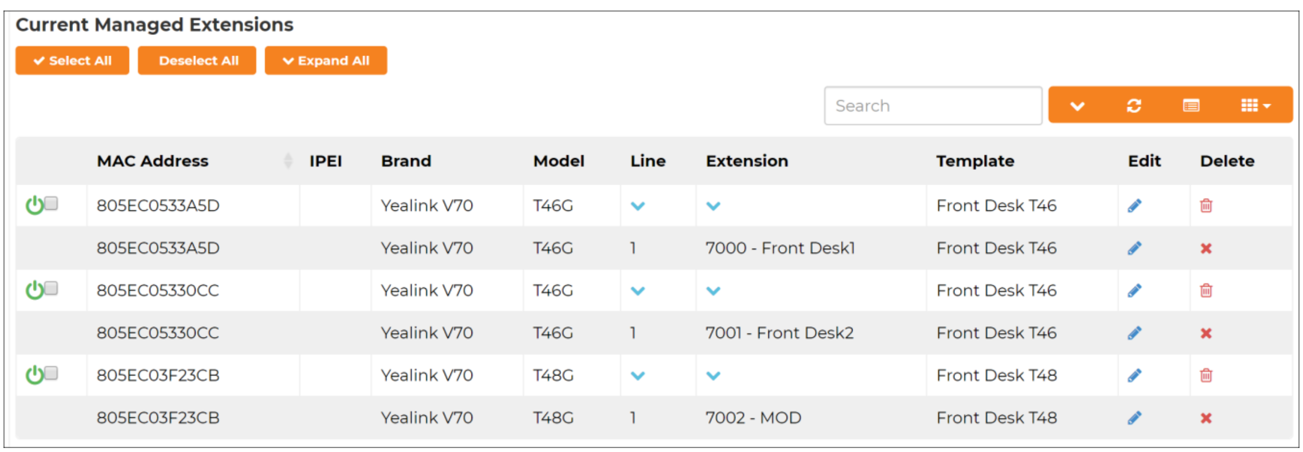
Navigate to Core Configuration > End Point Manager – Extension Mapping (fly away menu)

1. Fill in the MAC Address, Brand, Model of phone, the extension number and the template to use
2. Click on Add
3. Click on Rebuild
4. Connect the phone to a Power over Ethernet (POE) switch port or reboot the phone



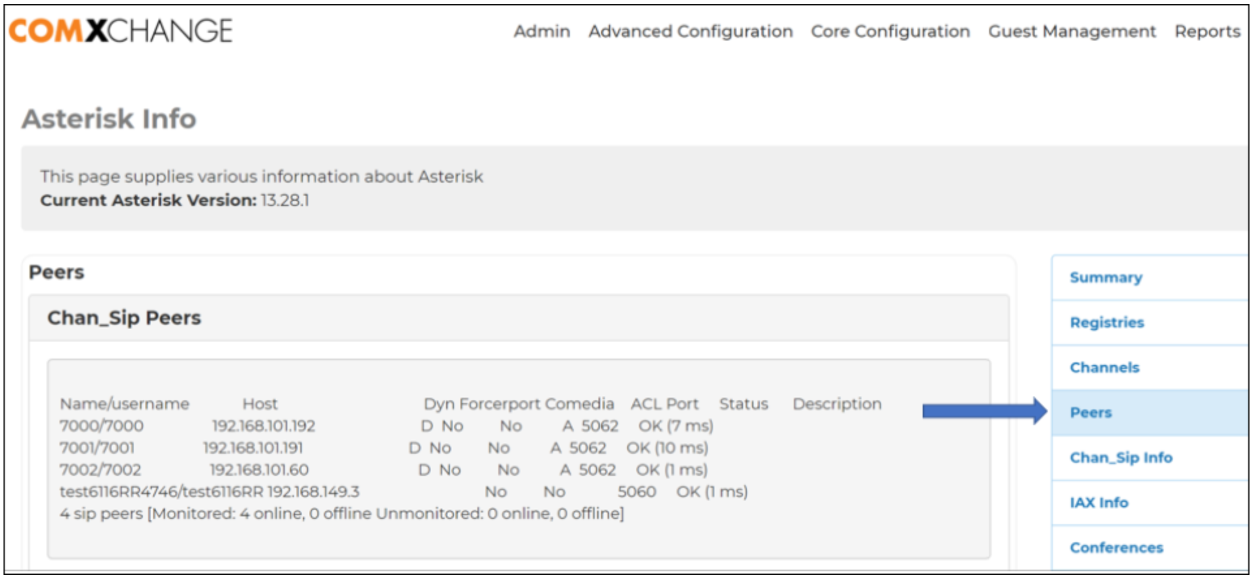
As changes are made to the templates you can click on Rebuild and check the Reboot Phone check box to force the phones to download the new configuration.

You should now see the devices appearing as up by in the Current Managed Extensions in the End Point Dashboard which means they have registered to the ComXchange Controller.


Current Managed Extensions

MAC Address	IPEI	Brand	Model	Line	Extension	Template	Edit	Delete
805EC0533A5D		Yealink V70	T46G			Front Desk T46		
805EC0533A5D		Yealink V70	T46G	1	7000 - Front Desk1	Front Desk T46		
805EC05330CC		Yealink V70	T46G			Front Desk T46		
805EC05330CC		Yealink V70	T46G	1	7001 - Front Desk2	Front Desk T46		
805EC03F23CB		Yealink V70	T48G			Front Desk T48		
805EC03F23CB		Yealink V70	T48G	1	7002 - MOD	Front Desk T48		

You can also verify the phones are registered by navigating to Reports > Asterisk Info – Peers, where you will get a list of the devices registered to the phone server.



Test Internal Calls and Voicemail

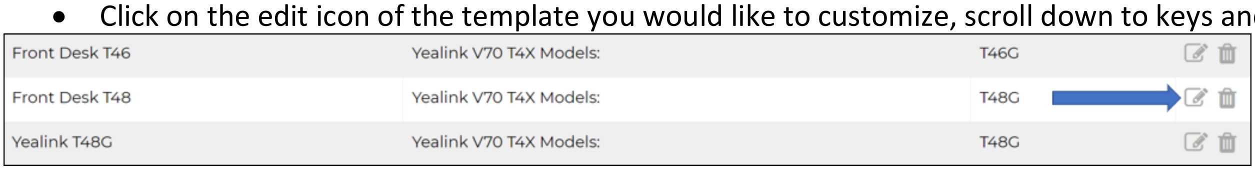
1. You can now make calls from extension to extension
2. Test the "Operator" Misc Application by dialing 0 from extension 7002 and verifying that it rings both 7000 and 7001 via the Ring group 600.
3. Dial the Voicemail Feature Code *55 to change the default greeting and password

IP Phone Customization

Common Features that are customized in IP Phones are Busy Lamp Field (BLF) buttons, Voicemail Speed Dial buttons, Parking and setting up Dial Plan Rules. These can be pushed out to phones by using a template.

Navigate to Core Configuration > End Point Manager – Template Manager (fly away menu)

- Click on the edit icon of the template you would like to customize, scroll down to keys and dial plan



Front Desk T46	Yealink V70 T4X Models:	T46G
Front Desk T48	Yealink V70 T4X Models:	T48G
Yealink T48G	Yealink V70 T4X Models:	T48G

Keys

Keys are related to the buttons whether physical or soft buttons on the phone screen. They can be configured to perform many different actions. Below is a list of some of the common keys.

- Busy Lamp Field (BLF) buttons monitor the state of another extension/feature and can be used to dial the extension when the extension is idle. Typically, a BLF will do direct pickup when ringing if monitoring an extension.
- Speed Dials can be set to dial a number
  - Note: if a ringroup has call pickup enabled a speed dial of **<EXT> can pickup that ring group
- You can set a key to direct dial voicemail
- Park is a built-in feature code that allows you to place calls on hold in up to 8 slots that can be picked by other extensions
- You can also create speed dials to connect to an application to do basic wakeup call and guest management

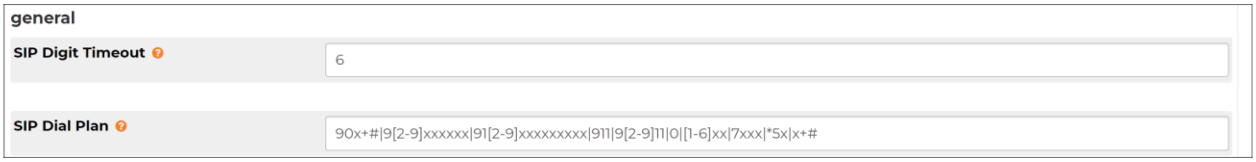
keys

Line Key 1 Type	BLF
Line Key 1 Line	Line 1
Line Key 1 Label	7000
Line Key 1 Value	7000
Line Key 1 Pickup Number	{$pickup_value}
Line Key 2 Type	BLF
Line Key 2 Line	Line 1
Line Key 2 Label	7001
Line Key 2 Value	7001
Line Key 2 Pickup Number	{$pickup_value}
Line Key 3 Type	XML Browser
Line Key 3 Line	Line 1
Line Key 3 Label	Wakeup
Line Key 3 Value	http://192.168.101.2/yl/wakeup.php
Line Key 3 Pickup Number	{$pickup_value}
Line Key 4 Type	XML Browser
Line Key 4 Line	Line 1
Line Key 4 Label	Guest Management
Line Key 4 Value	http://192.168.101.2/yl/guestmgmt.php
Line Key 4 Pickup Number	{$pickup_value}
Line Key 5 Type	BLF
Line Key 5 Line	Line 1
Line Key 5 Label	Mod
Line Key 5 Value	7002
Line Key 5 Pickup Number	{$pickup_value}
Line Key 6 Type	BLF
Line Key 6 Line	Line 1
Line Key 6 Label	Park
Line Key 6 Value	70
Line Key 6 Pickup Number	{$pickup_value}
Line Key 7 Type	BLF
Line Key 7 Line	Line 1
Line Key 7 Label	Park1
Line Key 7 Value	71
Line Key 7 Pickup Number	{$pickup_value}
Line Key 8 Type	BLF
Line Key 8 Line	Line 1
Line Key 8 Label	Park2
Line Key 8 Value	72
Line Key 8 Pickup Number	{$pickup_value}

Dial Plan

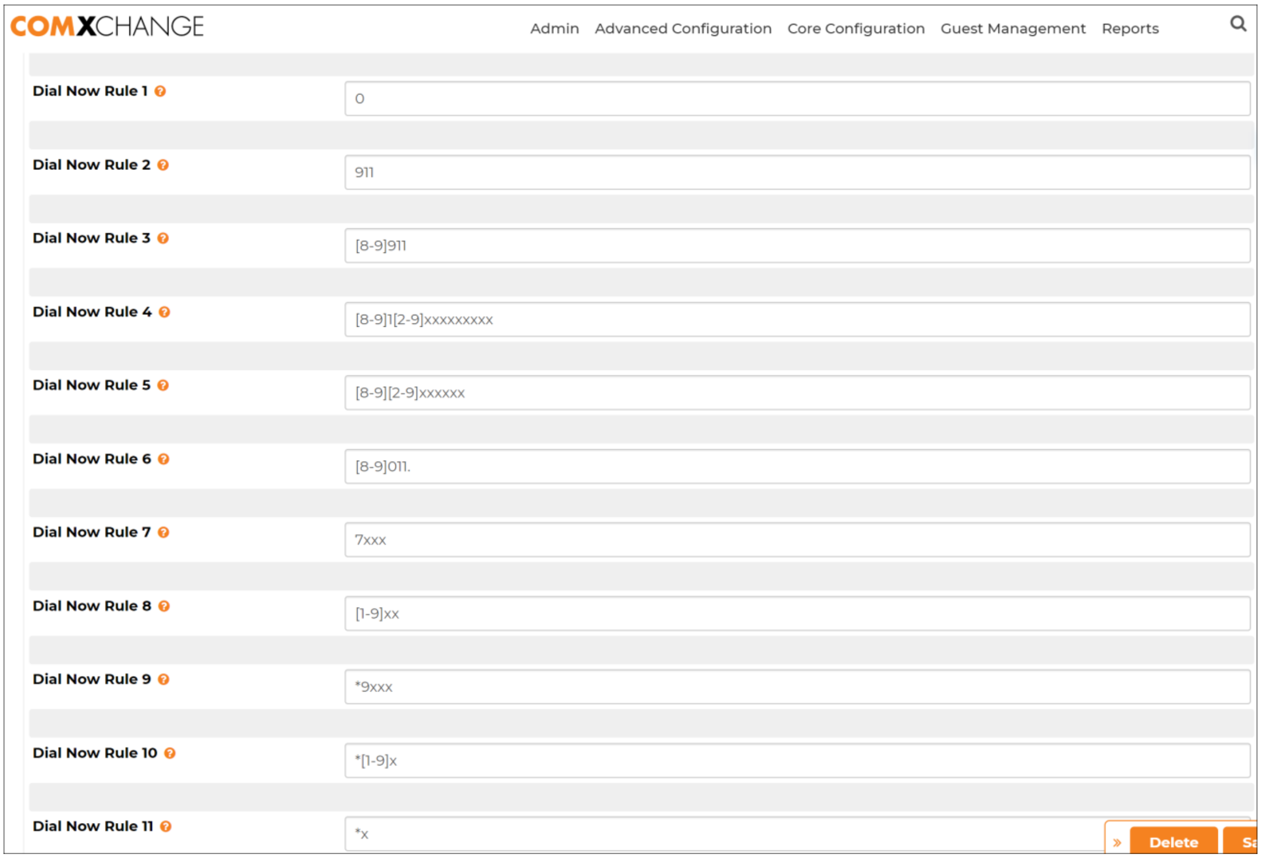
Dial plans are needed to be able to dial numbers without a timeout. Full documentation on dialing rules can be found in the phone's administration Guide. Below are examples from an Aastra Dial Plan and the Yealink Dial Now Rules.

Aastra



90x+#|9[2-9]xxxxxx|91[2-9]xxxxxxxxx|911|9[2-9]11|0|[1-6]xx|7xxx|*5x|x+#

Yealink



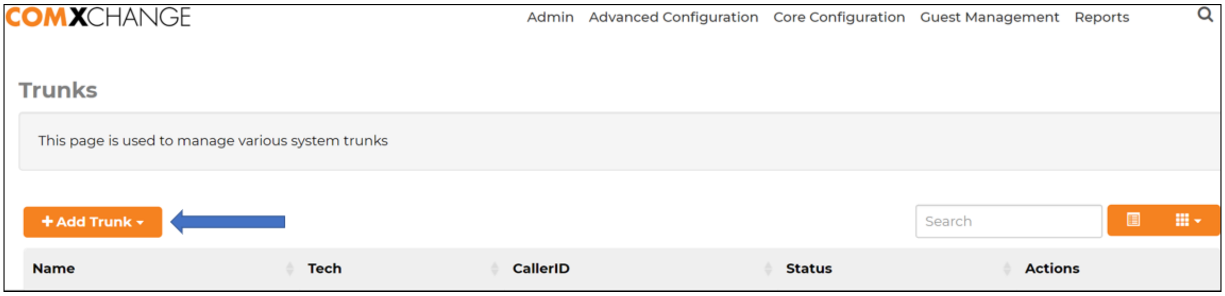
Trunking Overview and Setup

- Trunks are delivered to the ComXchange using Session Initiated Protocol (SIP)
- T1,PRI, POTS are "converted to SIP using a VoIP gateway.
- Direct SIP Trunks are usually routed to the ComXchange controller on the eth1/WAN interface.

Configuring a Trunk

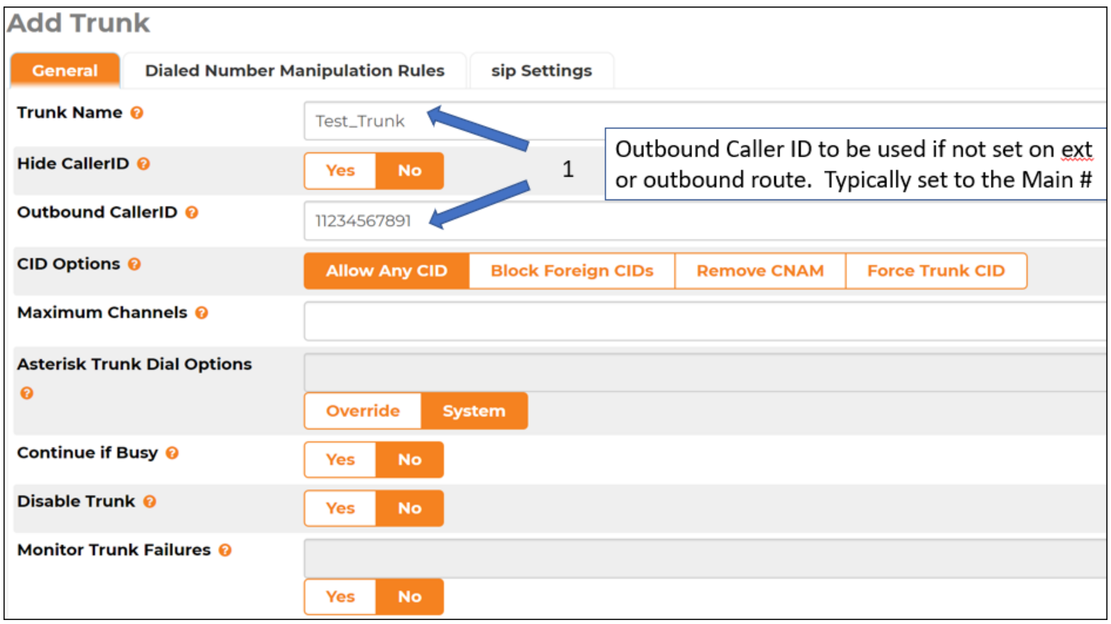
Navigate to Core Configuration > Trunks - Click on Add Trunk

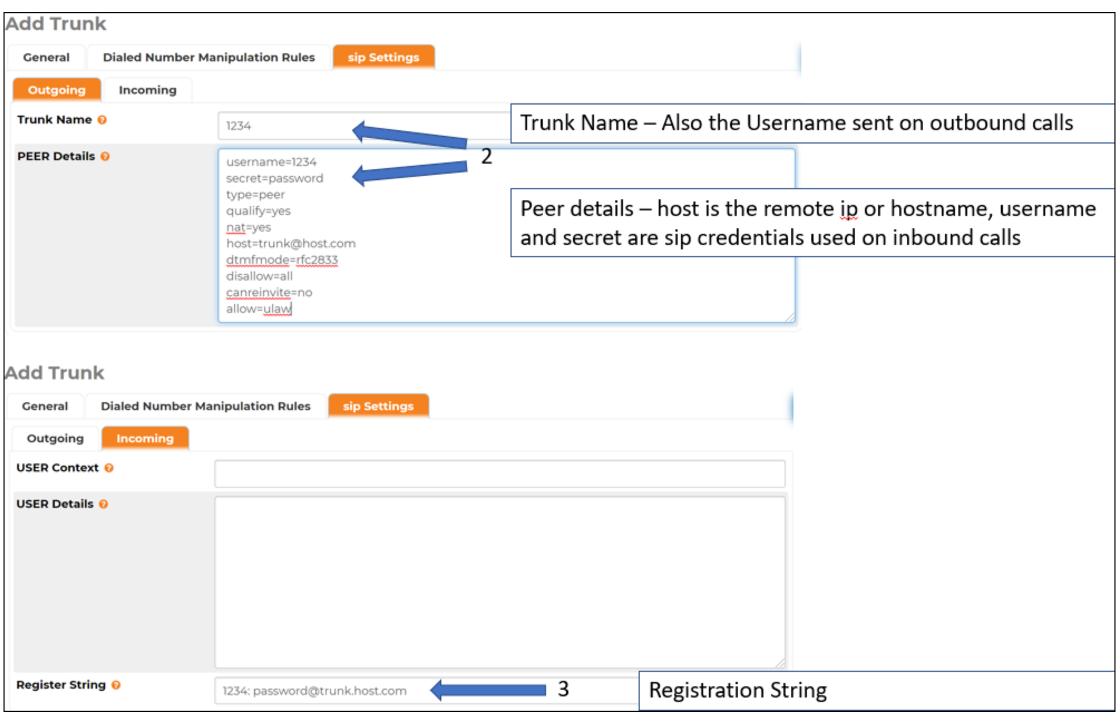
- Choose add SIP (chan_sip) Trunk



Direct SIP Trunk

1. Under the General Tab, fill in the Trunk name (descriptive name), and Outbound CallerID
2. Under the sip Settings – Outgoing Tab, fill in the Trunk Name, Peer Details
3. Under the sip Settings – Incoming Tab, add the Register String



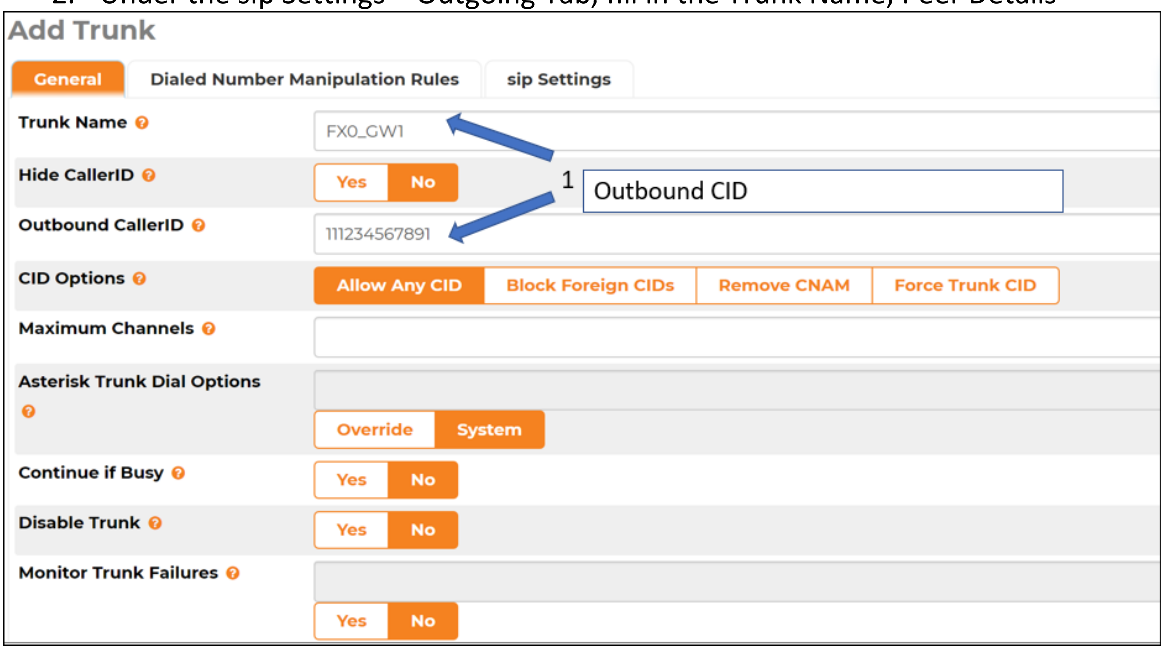


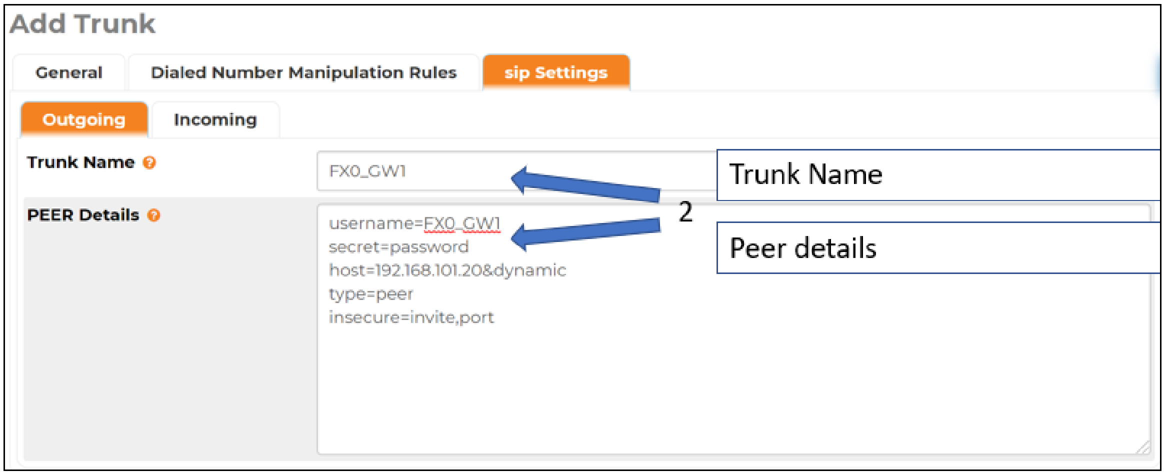
If there is connectivity with the SIP Provider or Gateway and the Trunk settings match on both sides of the trunk, you should now be registered to a host which can be verified in Reports > Asterisk Info – Registries.

Trunk to Gateway

When you are using a VoIP Gateway the ComXchange and the gateway will connect over the LAN connection. The Settings will be very similar to a direct SIP connection however you will not need a registration string.

1. Under the General Tab, fill in the Trunk name (descriptive name), and Outbound CallerID
2. Under the sip Settings – Outgoing Tab, fill in the Trunk Name, Peer Details





Inbound / Outbound Routing Overview

Inbound phone calls are routed to extensions when incoming dialed digits match a rule telling the ComXchange which extension should receive the call. Outbound calls must also match rules that tell the ComXchange where to send the call, whether it is an internal call or one that should go out a Trunk. These rules are setup as Inbound and Outbound routes.

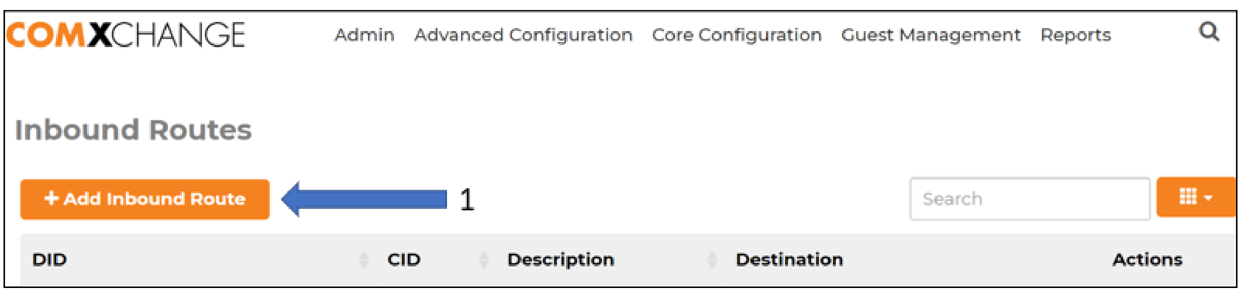
Inbound Routes

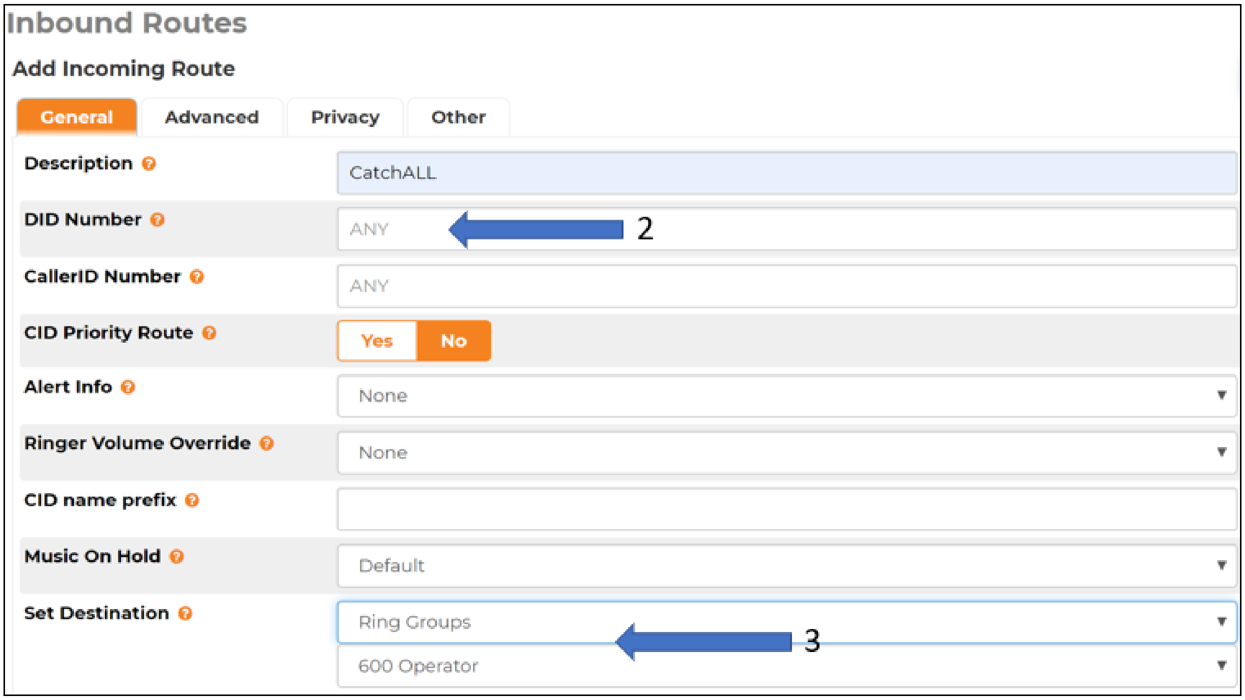
A telephone carrier will direct calls to a property based on a telephone number. There can be multiple Direct Inward Dialing (DID) numbers that point to the main phone number. To receive calls ComXchange must know where to route them. Below we will set up a catchall route that will send all calls to the Operator Ring Group.

Catchall Route

Navigate to Core Configuration > Inbound Routes.

1. Click on Add Inbound Route
2. Leave the DID number field blank. Which means any calls with or without a DID will match.
3. Set the Destination to Ring Group to Operator 600



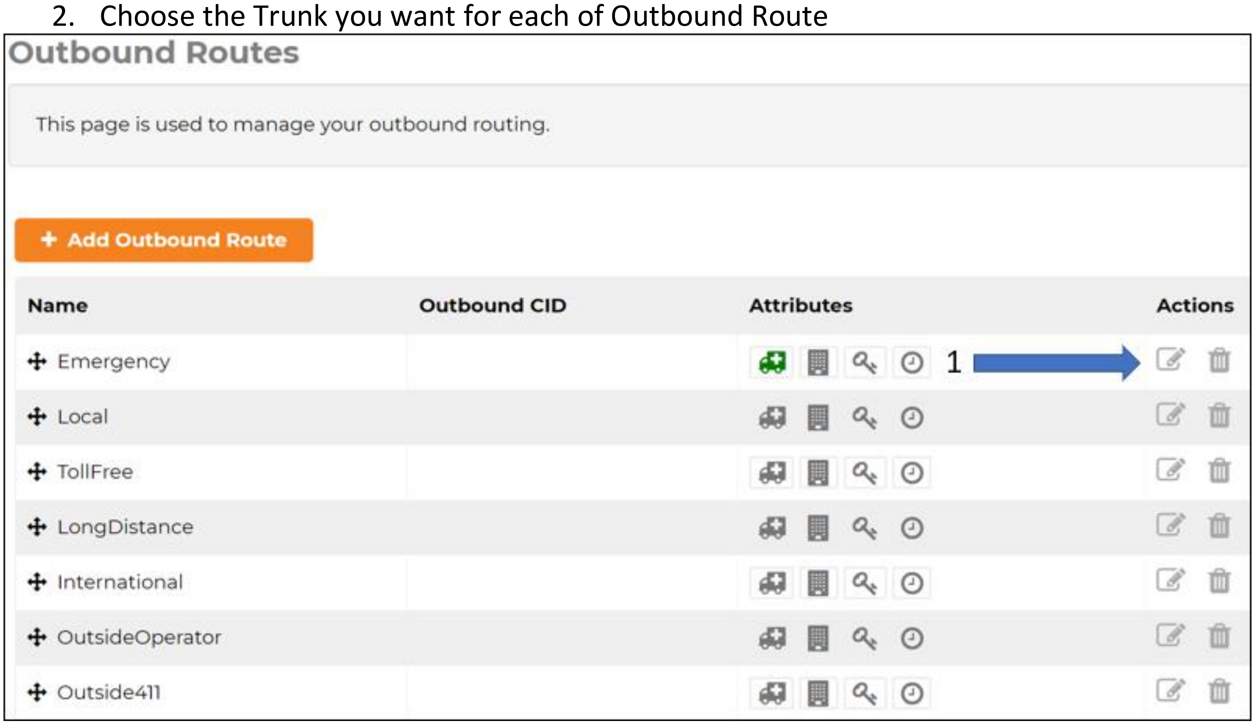


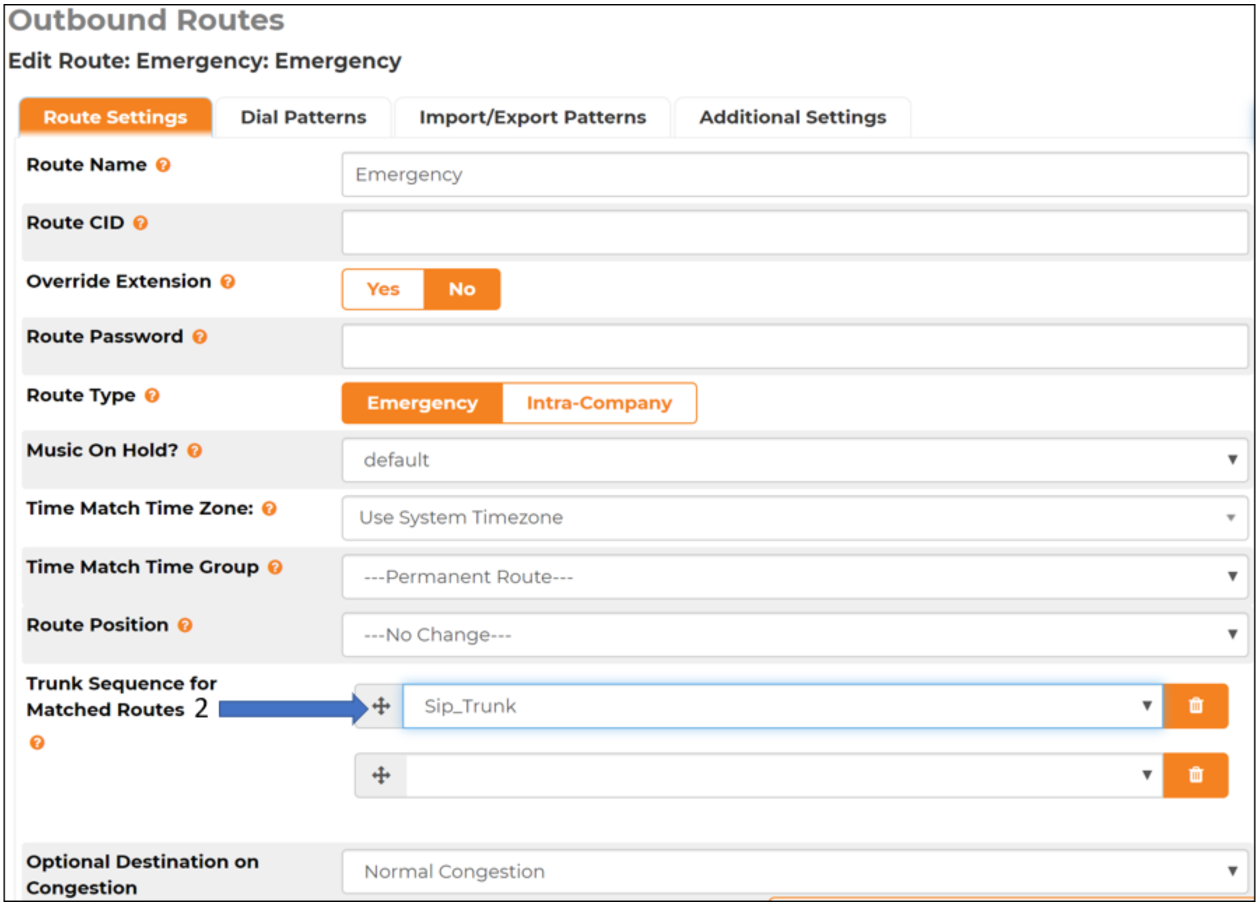
Outbound Routes

ComXchange has default outbound routes prepopulated with dial patterns that need to be associated with a Trunk. Each of these routes will need to be associated with a Trunk for calls to be directed to a telephone provider. Order is important. For example, toll free should be before long distance.

Navigate to Core Configuration > Outbound Routes

1. Open each route by clicking on the edit Actions icon
2. Choose the Trunk you want for each of Outbound Route





Dial Patterns

ComXchange uses dial patterns to match dialed digits. You can use the below dial pattern rules to set up unique routes, however, ComXchange has common default routes already set up for you in the Outbound Routes Dial Pattern tab.

A Dial Pattern is a unique set of digits that will select this route and send the call to the designated trunks. If a dialed pattern matches this route, no subsequent routes will be tried. If Time Groups are enabled, subsequent routes will be checked for matches outside of the designated time(s).

Rules:
X matches any digit from 0-9
Z matches any digit from 1-9
N matches any digit from 2-9
[1237-9] matches any digit in the brackets (example: 1,2,3,7,8,9)
. wildcard, matches one or more dialed digits
prepend: Digits to prepend to a successful match. If the dialed number matches the patterns specified by the subsequent columns, then this will be prepended before sending to the trunks.
prefix: Prefix to remove on a successful match. The dialed number is compared to this and the subsequent columns for a match. Upon a match, this prefix is removed from the dialed number before sending it to the trunks.
match pattern: The dialed number will be compared against the prefix + this match pattern. Upon a match, the match pattern portion of the dialed number will be sent to the trunks
CallerID: If CallerID is supplied, the dialed number will only match the prefix + match pattern if the CallerID being transmitted matches this. When extensions make outbound calls, the CallerID will be their extension number and NOT their Outbound CID. The above special matching sequences can be used for CallerID matching similar to other number matches.

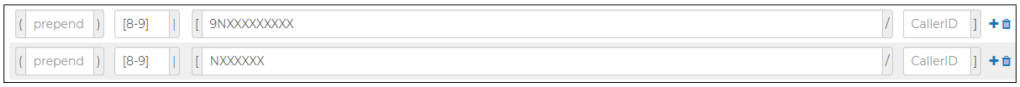
Navigate to Core Configuration > Outbound Routes – Dial Pattern tab

Click on Each of the Routes

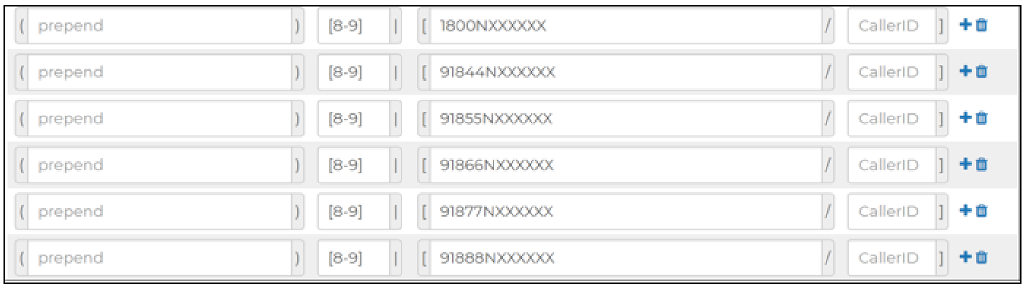
Default Dial Patterns have a prefix that is needed to dial out then stripped before sending to the trunk

Emergency

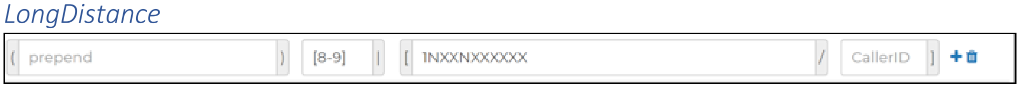
Local



TollFree



LongDistance



International

Outside operator

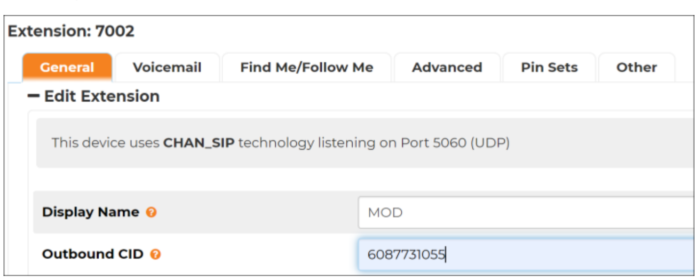
Outbound Caller ID Considerations

- Caller ID (CID) can be set in Extensions, Outbound Routes, and Trunks.
- CID set in Extensions override Outbound Routes (unless override extension is checked) in the Trunk
- CID on Outbound Routes overrides CID set in Trunks.
- Trunk Caller ID is used as a last resort. If only one number is being sent from the system, typically just set the CID at the trunk level.

Test Calls

Calls can now be made to and from the ComXchange. You can now make a test call from one of the phones.

Change the Caller ID on Extension 7002 to 6087731055 and place a call to see the CID change.



Emergency Notification Overview

There is an Emergency Outbound route set to match emergency dialing patterns such as 911. By selecting the Emergency Route Type Box any calls matching the dial patterns will also trigger a notification to be sent to the destinations set in the Emergency Notifications Settings Module.

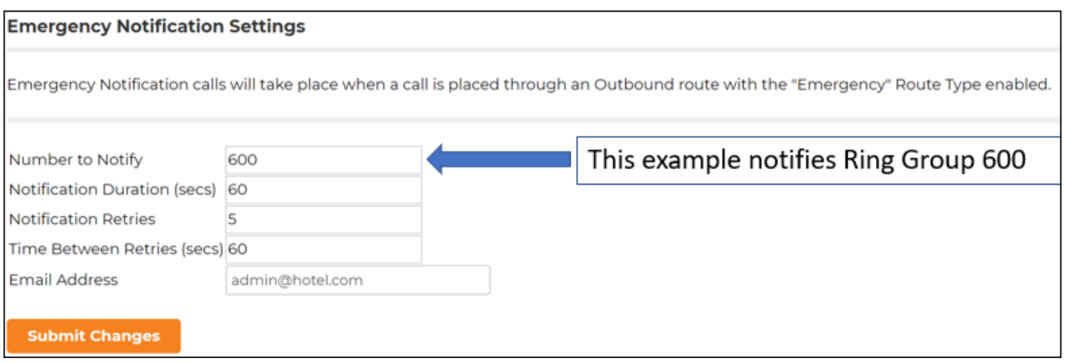
Emergency Outbound Route

Emergency Notifications Settings

A notification phone call and or email can be sent when a call is made on an outbound route designated as an emergency route. The notification will contain the extension that dialed 911. A call notification must be answered and acknowledged by pressing 1.

Navigate to Admin > Emergency Notifications

- Add a Number to Notify - If a 911 call is placed this number will be notified
- Add Email Address or Addresses separated by a comma



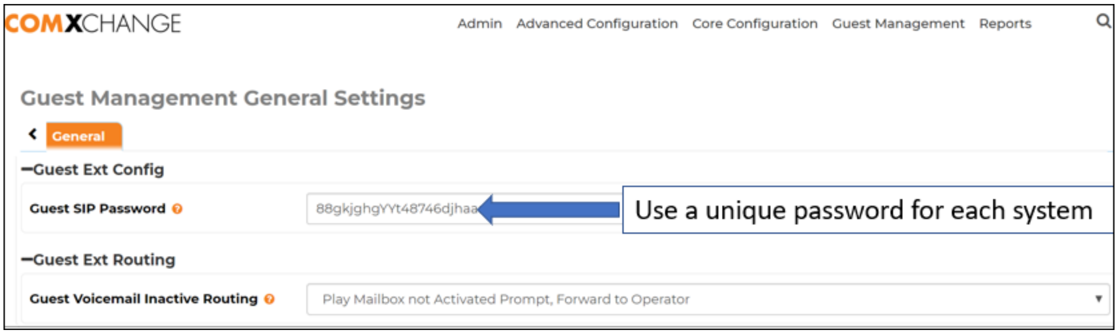
Guest extension Overview

General Settings

This is where the Global SIP password for guest extensions is set. You will also find the Room Status/ Housekeeping feature code and the Guest Voicemail Access Code.

Navigate to Guest Management > General Settings.

- The Guest SIP Password is a universal password that guest phones use to register to the guest system
- The Guest Extension Routing option allows you to choose an action for an inactive mailbox.

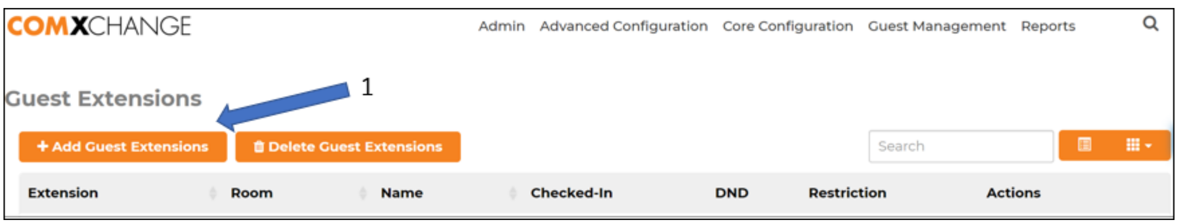


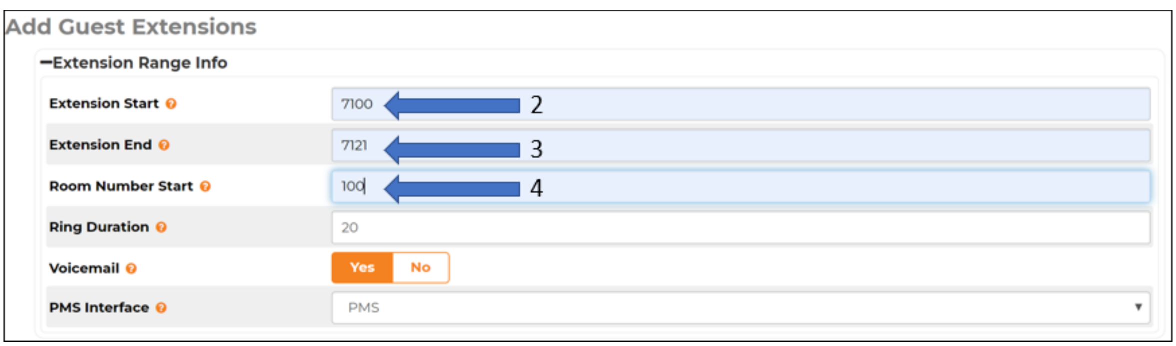
Guest Extensions

- Extension Range Creation creates extension(s) and assigns them to a room.
- The room number is the field used in communications with the PMS system.
- The room number can be left blank if it matches the extension
- Guest extension with the same room number will share a mailbox and ring all phones

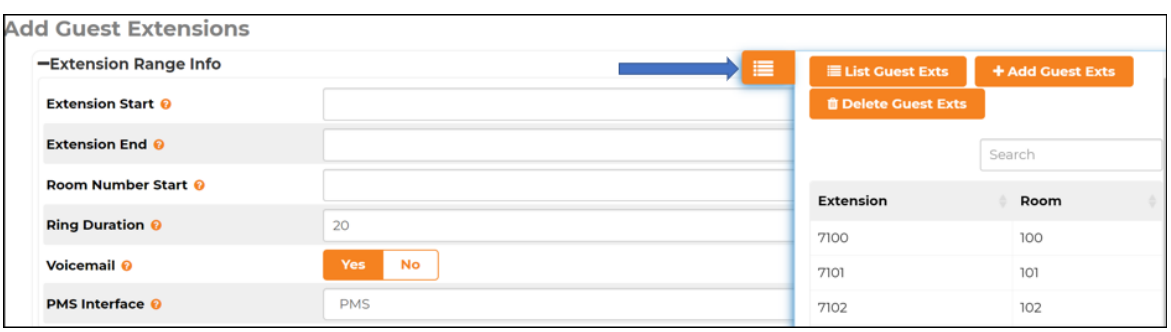
Navigate to Guest Management > Guest Extensions

1. Click on Add Guest Extensions
2. Enter Extension Start: 100
3. Enter Extension End: 121
4. Enter 100 for Room Number Start
5. Submit and Apply Config





The Created Guest Can be viewed in the right side fly away menu.



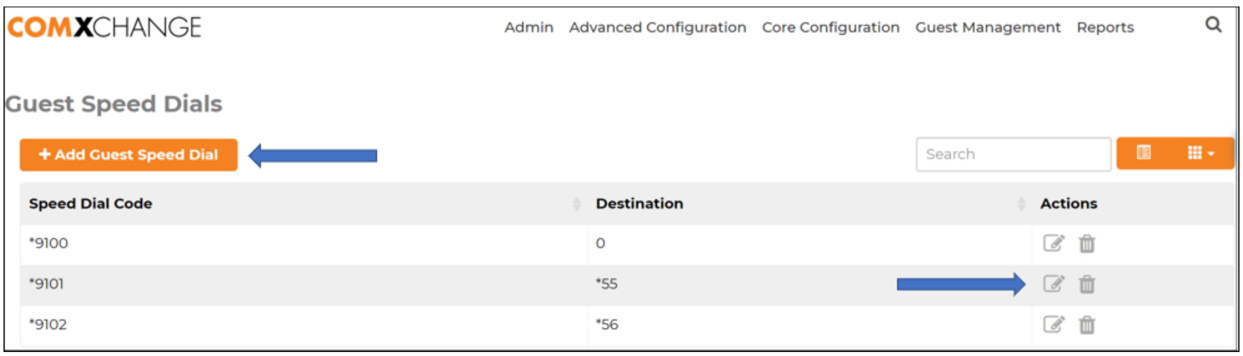
You can click on an extension to verify the room number and make any changes.

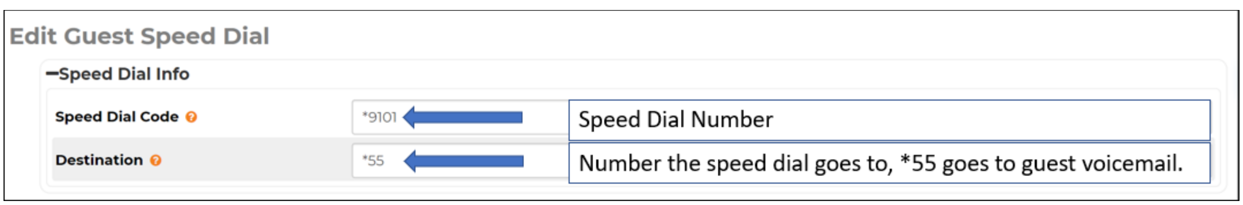
Guest Speed Dials

Guest Speed Dials can be set to match the speed dial buttons programmed on many analog guest phones. The phone buttons will be programmed specifically for a phone by the manufacturer.

Navigate to Guest Management > Guest Speed Dials - [Speed Dial Number]

- Click on the edit Actions icon to edit a speed dial or you can add a Guest Speed Dial
- Speed Dial Code is what the manufacturer programmed into the phone
- Number is the digits a call is forwarded to when it receives a Speed Dial Code





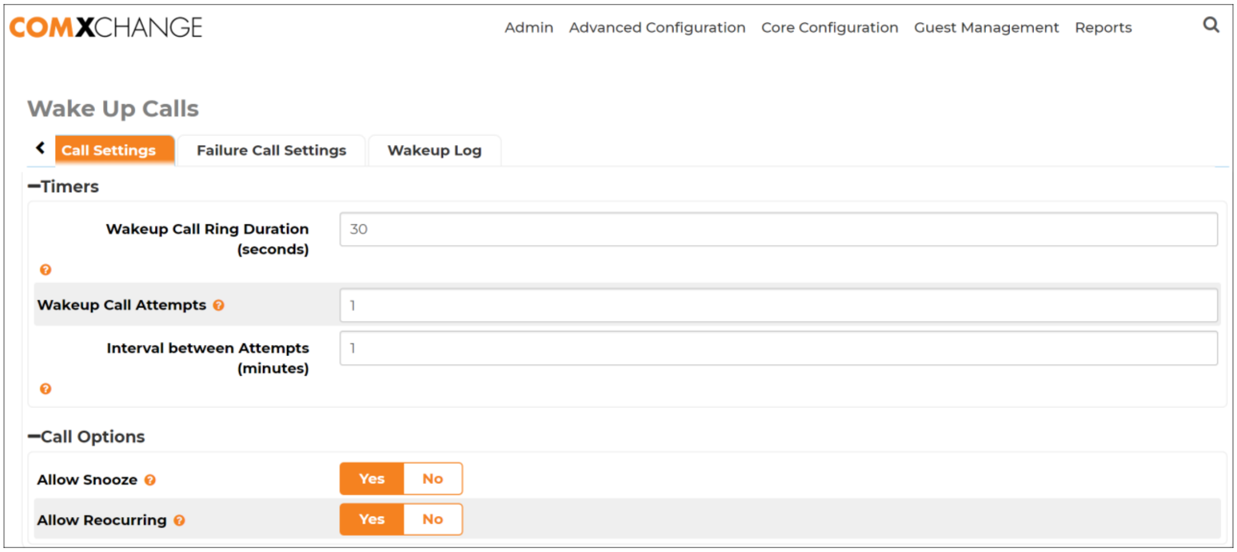
Wake Up Calls

Wake up calls can be created through the Hotel Dashboard, using the Guest Console phone App and directly dialing a feature code. Wakeup call settings and logs can be found in Guest Management > Wakeup Calls

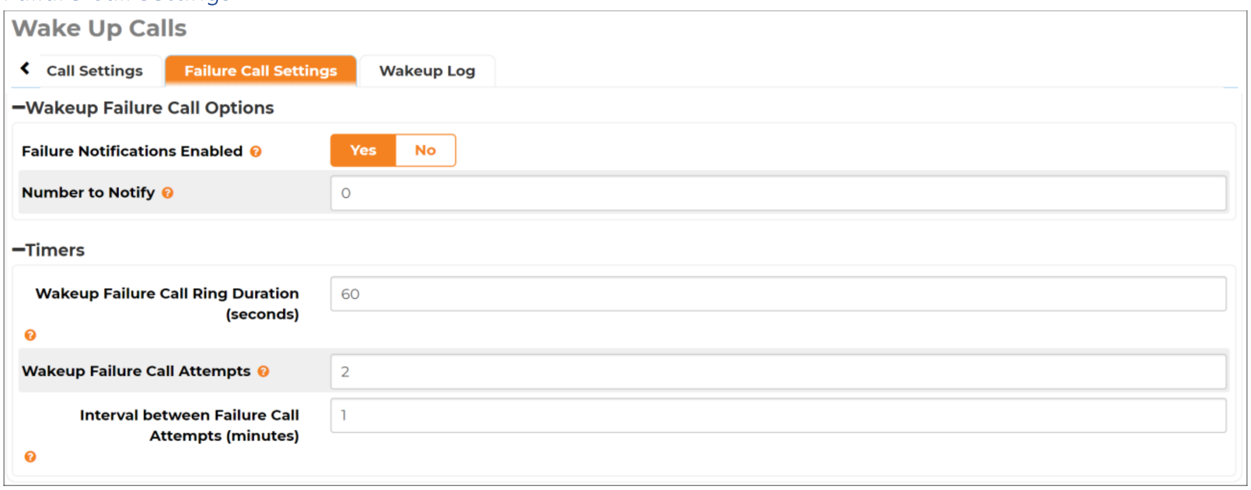
Wakeup Call Settings

- Navigate to Guest Management > Wakeup Calls
- You can change the default Wakeup Call Settings for ring duration, call attempts, allowing snooze, etc
- The Failure Calls Settings tab allows you to set an extension to notify if a wakeup call is not answered

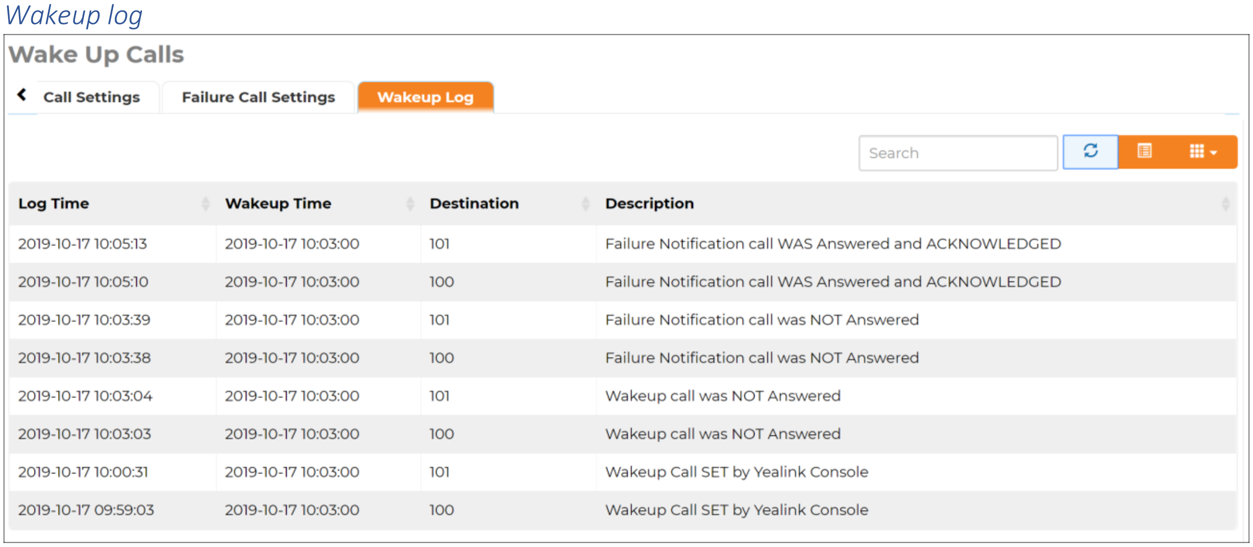
Call Settings



Failure Call Settings



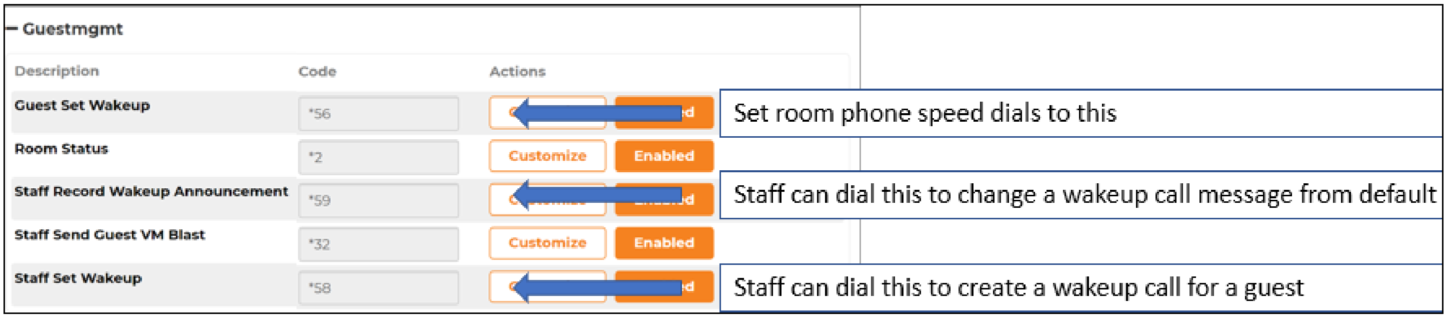
Wakeup log



Log Time	Wakeup Time	Destination	Description
2019-10-17 10:05:13	2019-10-17 10:03:00	101	Failure Notification call WAS Answered and ACKNOWLEDGED
2019-10-17 10:05:10	2019-10-17 10:03:00	100	Failure Notification call WAS Answered and ACKNOWLEDGED
2019-10-17 10:03:39	2019-10-17 10:03:00	101	Failure Notification call was NOT Answered
2019-10-17 10:03:38	2019-10-17 10:03:00	100	Failure Notification call was NOT Answered
2019-10-17 10:03:04	2019-10-17 10:03:00	101	Wakeup call was NOT Answered
2019-10-17 10:03:03	2019-10-17 10:03:00	100	Wakeup call was NOT Answered
2019-10-17 10:00:31	2019-10-17 10:03:00	101	Wakeup Call SET by Yealink Console
2019-10-17 09:59:03	2019-10-17 10:03:00	100	Wakeup Call SET by Yealink Console

Guest management Wakeup Feature Codes

There are Guest management feature codes that can be directly dialed to set Wakeup Calls and Announcements.



Analog Guest Phones

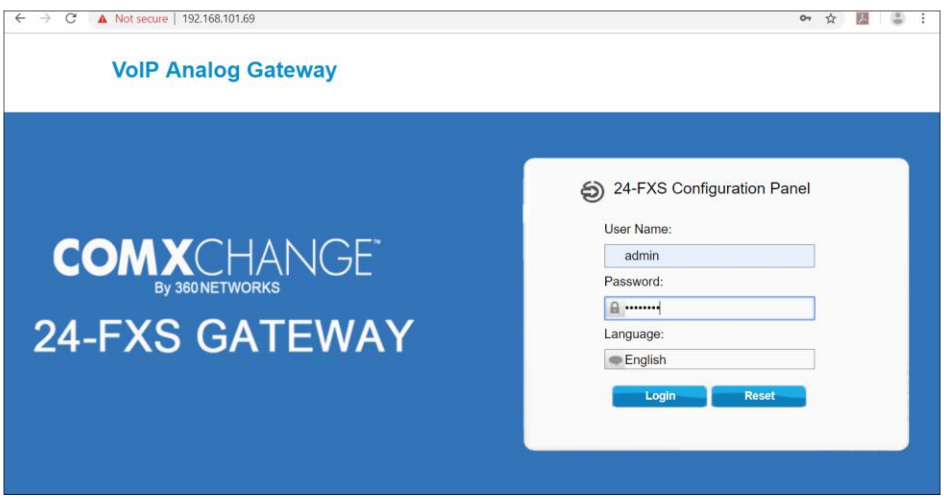
Guest phones are typically analog phones, these phones translate an audio signal into electronic pulses that travel across a phone line to another analog device. Because ComXchange communicates using VoIP the analog signal must be translated by an FXS Gateway into a digital VoIP call to be processed by the ComXchange Phone System. The gateway will have ports/connections that provides dial tone, battery current, and ring voltage for an analog phone. The gateway will also have a port that connects to an IP network for VoIP communication. Any calls made to or from an analog phone connected to a gateway will be translated from analog to digital or digital to analog making communication possible.

ComXchange Gateway Configuration

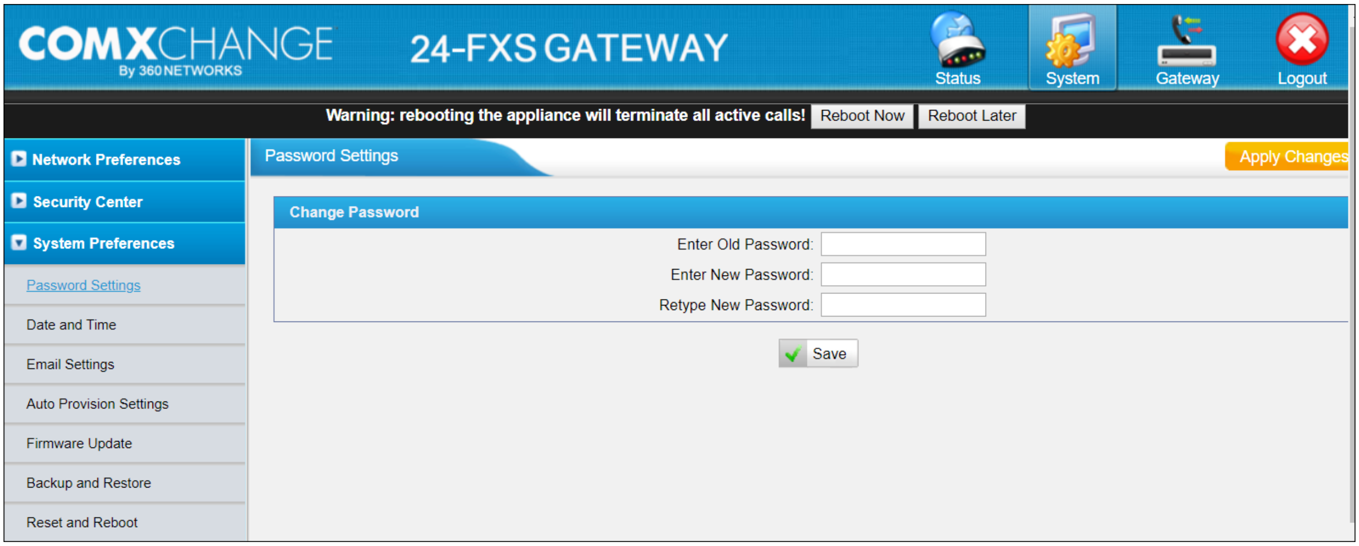
To configure a ComXchange gateway you will first need to find the IP address. Gateways will be shipped with DHCP enabled on them. They will need to be plugged into the switch to get an IP address. You can find the IP address by plugging an analog phone into the gateway and dialing *** or look in the dhcp leases in Admin> System Settings > DHCP Server -leases. Type the IP address into the address bar of a browser to bring up the web GUI.

Password

Default username and password is User Name: admin and Password: password.



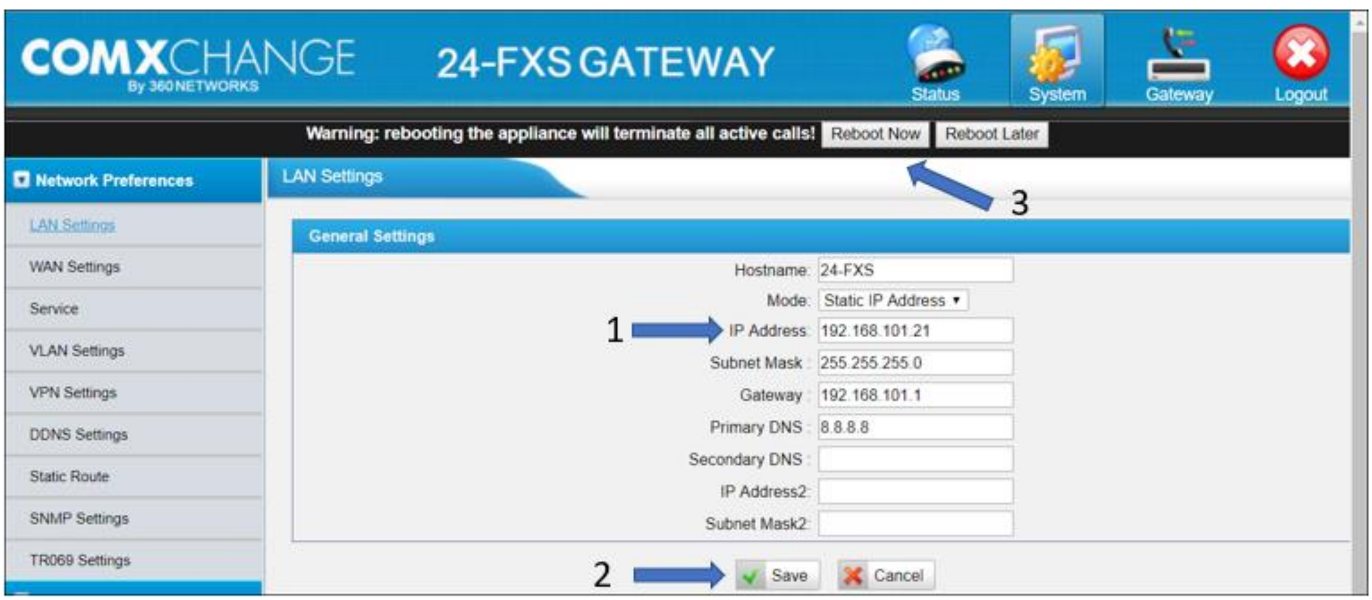
Navigate to System > System Preferences > Password Settings to change the password.



Set IP Address

Navigate to System > LAN Settings

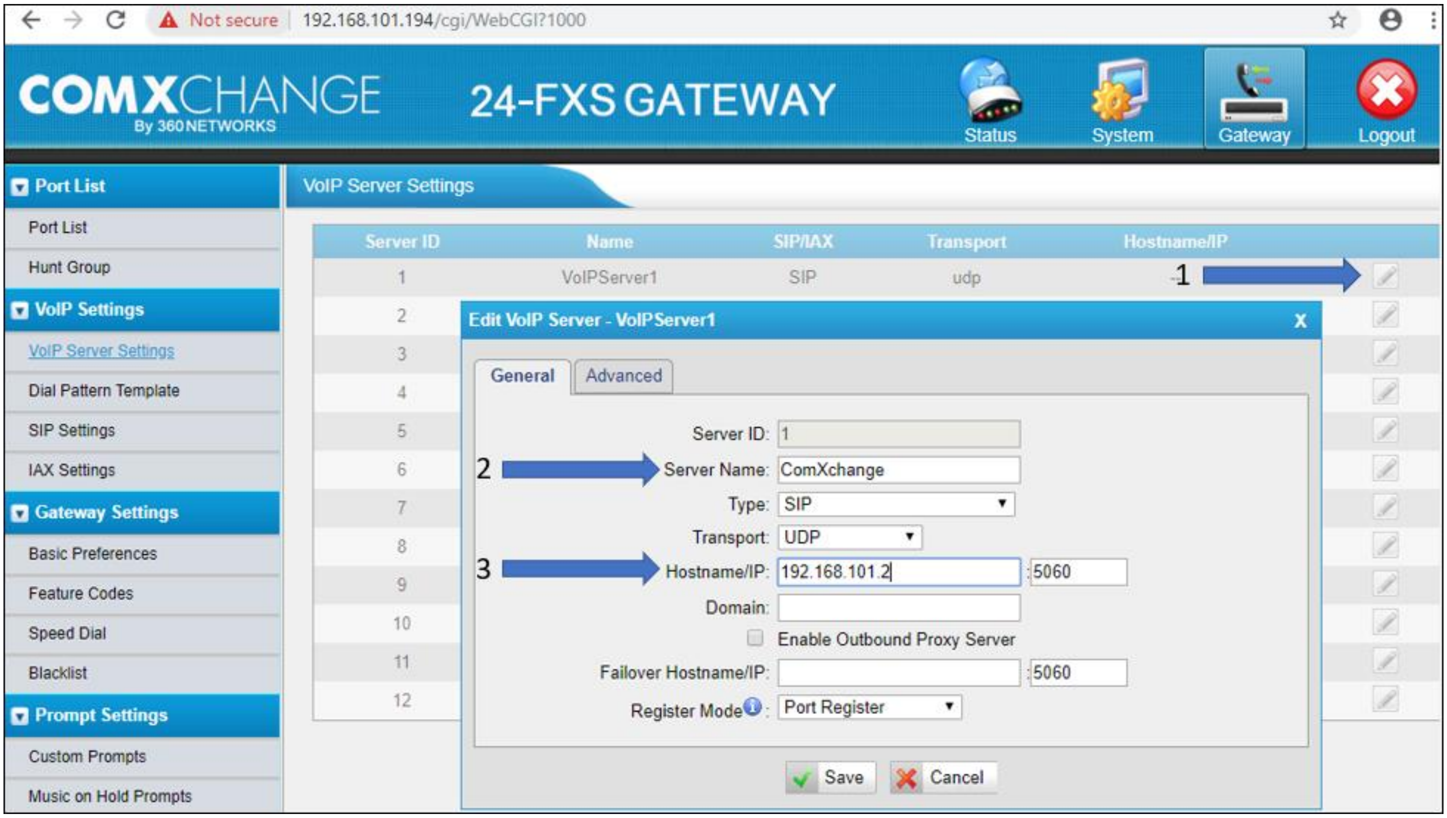
1. Change the IP address information. In the addressing scheme 192.168.101.21 can be used as the first gateway IP address.
2. Save the changes.
3. Reboot the gateway
4. Log into the web GUI using the new IP address.



Voip Servers

Navigate to Gateway > Voip Settings > VoIP Server Settings

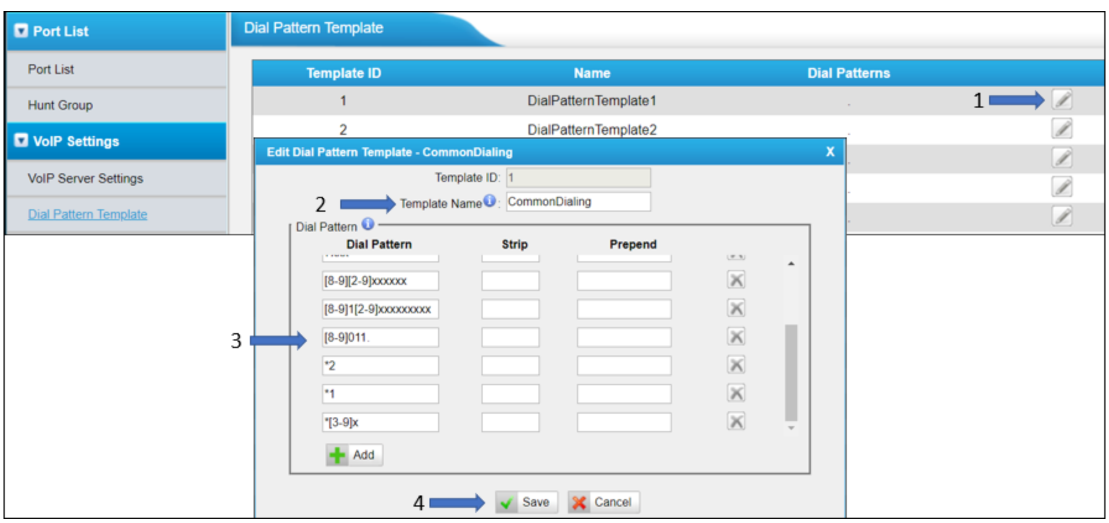
1. Click on Edit VoIPServer1
2. Edit the Server Name to ComXchange (or other)
3. Enter 192.168.101.2 in the Hostname/IP field and save then Apply Change



Dial Plan

Like IP phones; gateways have their own dial plan configuration setup to match patterns that can be dialed without a timeout. Navigate to Gateway > Voip Settings > Dial Pattern Template

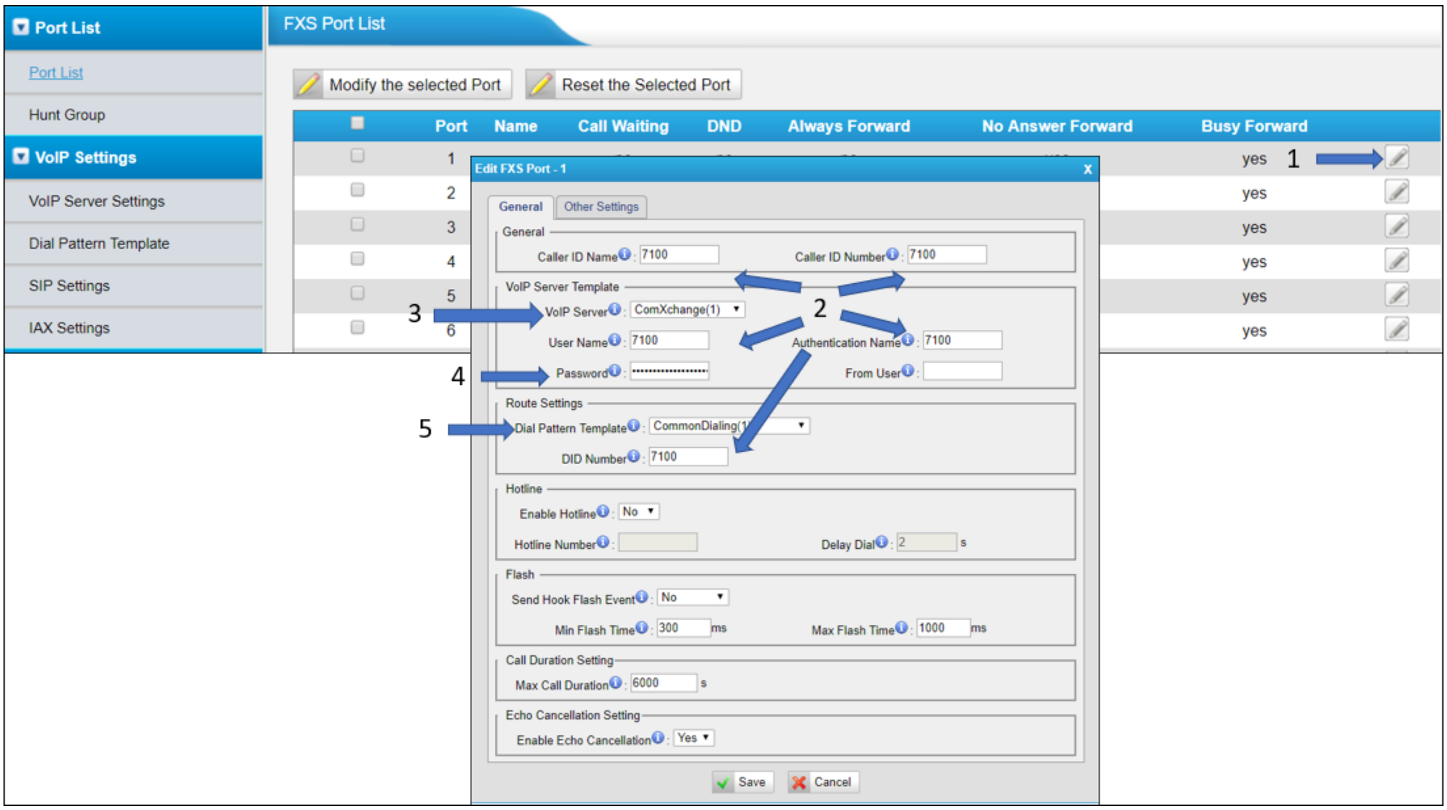
1. Click on Edit for DialPatternTemplate1
2. Edit the name to CommonDialing
3. Set Dial Patterns for 0, 911, [8-9]911, [8-9], [2-9]xxxxxx, [8-9]1[2-9]xxxxxxxxx, [8-9]011., [1-9]xx,7xxx, *2, *1, *[3-9]x
4. Click on Save



Extensions/Ports

A port on a gateway acts as a proxy SIP device for an analog phone. You will need to associate an extension on the phone server with a port on the gateway much like setting up an extension on an IP Phone. Below are the settings needed to register a port with extension 7100 to the ComXchange controller which will allow the gateway to translate calls to an analog phone connected to the port.

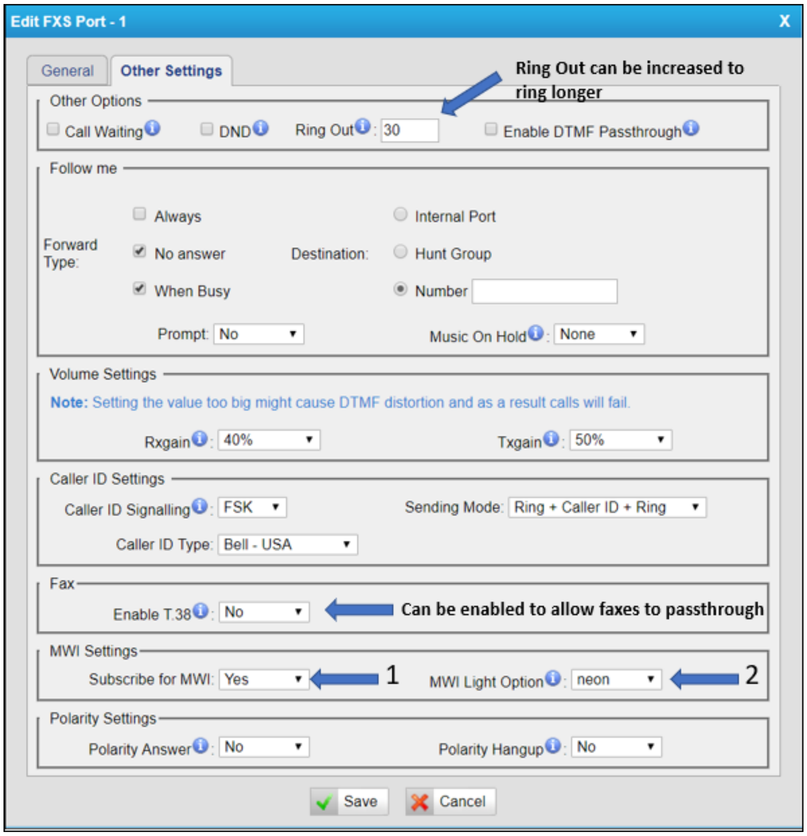
1. Click on Edit Port
2. Set the Caller ID Name, Caller ID Number, Username, and Authentication Name and DID Number field to the extension number (7100).
3. In the VoIP Server dropdown choose the Guest PBX.
4. Fill in the SIP password that was created in the Guest General settings in the ComXchange.
   a. Guest Management > General Settings > Guest SIP Password
5. In the Dial Pattern Template dropdown associate a dial pattern for the device.



The other tab has additional extension settings can be set including the MWI light.

MWI Settings

1. Set Subscribe to MWI to Yes.
2. Set MWI Light Option to neon.

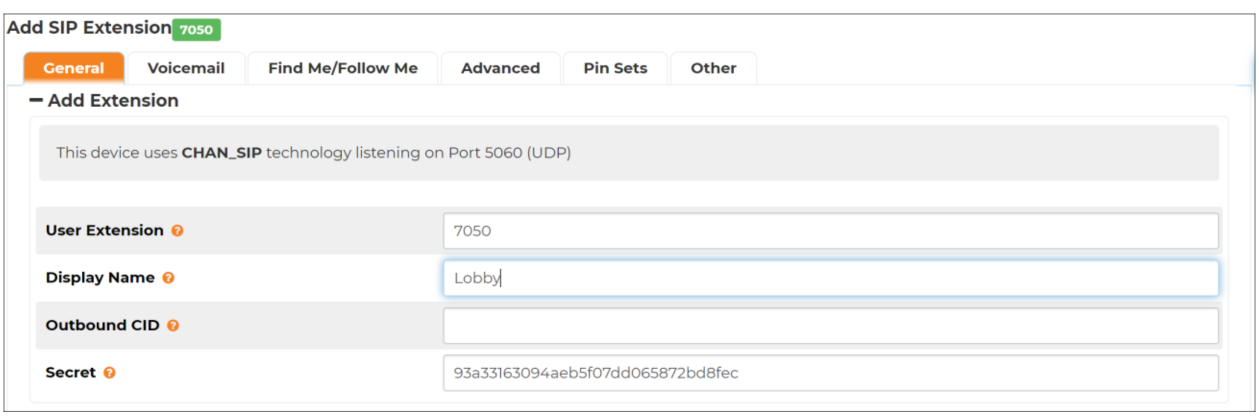


Add extension 7101 on port 2 with the same settings. You can now call from 7101 (room 101) to 7100 (room 100) and they can both dial 0 to get to the operator queue.

HotLine

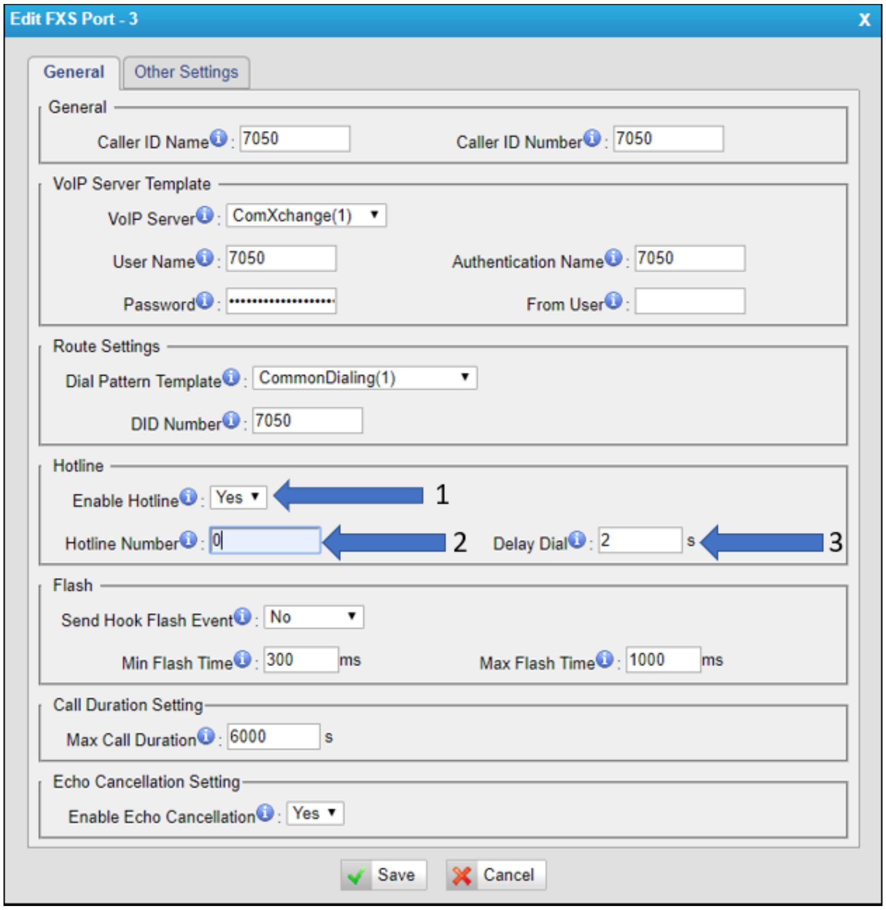
A hotline can be used by an extension such as a lobby phone to automatically dial a number such as 0 (the operator group). To do this you will need to add an extension for the Lobby phone on the Business PBX and set the gateway port to use that Extension and the Staff VoIP Server then enable the hotline.

Add Staff Lobby Phone extension



On the gateway port match the extension properties to register the port, then set the Hotline Settings.

1. Set Enable Hotline to Yes
2. Set the Hotline Number to 0
3. Adjust the Delay Dial if necessary



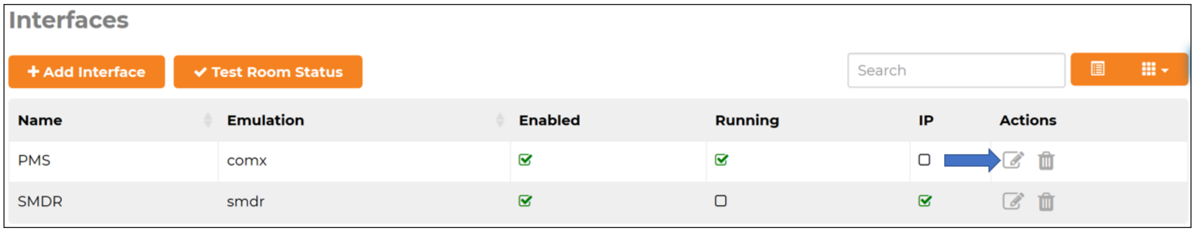
Guest Management Interfaces

ComXchange interfaces with leading Property Management Systems for both room status and call accounting. Communication between the systems takes place over a serial connection. The settings for these can be found in Guest Management > Interfaces. Here you can also view the interface logs allowing you to verify the information sent to the PMS over the PMS interface and SMDR (Station Messaging Detail Records) logs sent from the PBX to the Call Accounting Server where call costs are processed then sent from the Call Accounting server to the PMS.

PMS Interface

The PMS interface can connect with a PMS system to send and receive information. The ComXchange server can send Room Status updates that are keyed in at the guest rooms and the PMS can send Check-in information and call restrictions to the ComXchange server. In ComXchange 14 you can create and manage interfaces to connect to multiple PMS systems this is necessary when dealing with dual branded properties.

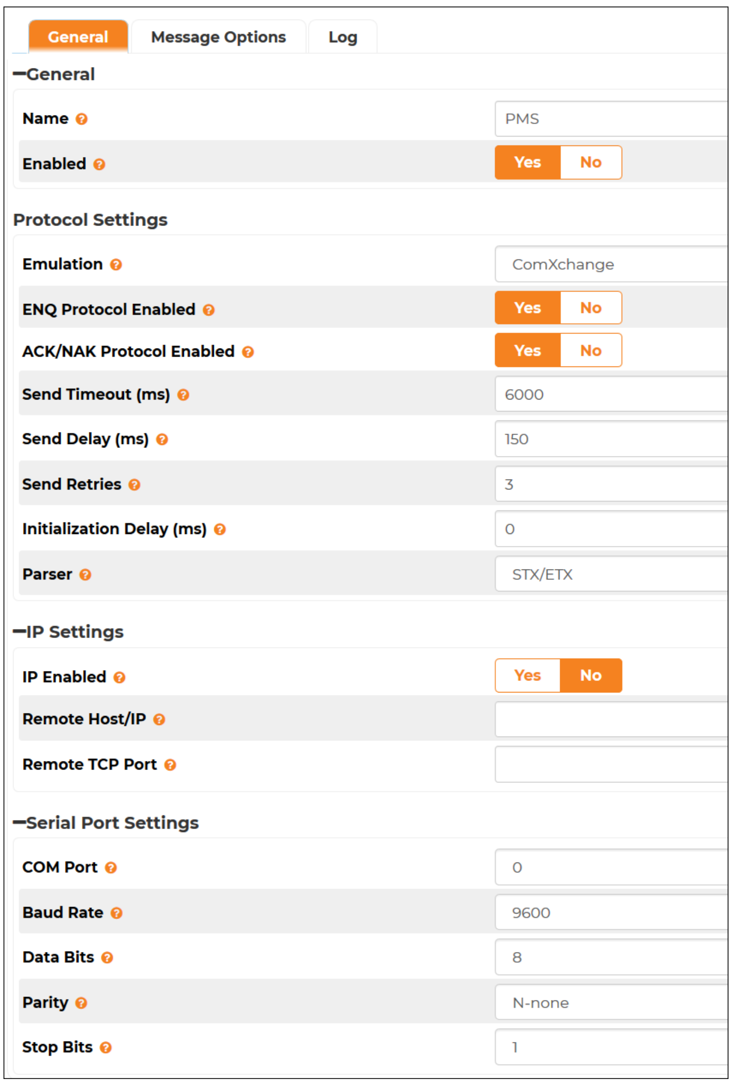
Navigate to Guest Management > Interfaces – Click on the PMS edit Actions icon.



Here you can Start and Stop the interface or Request a Resync.

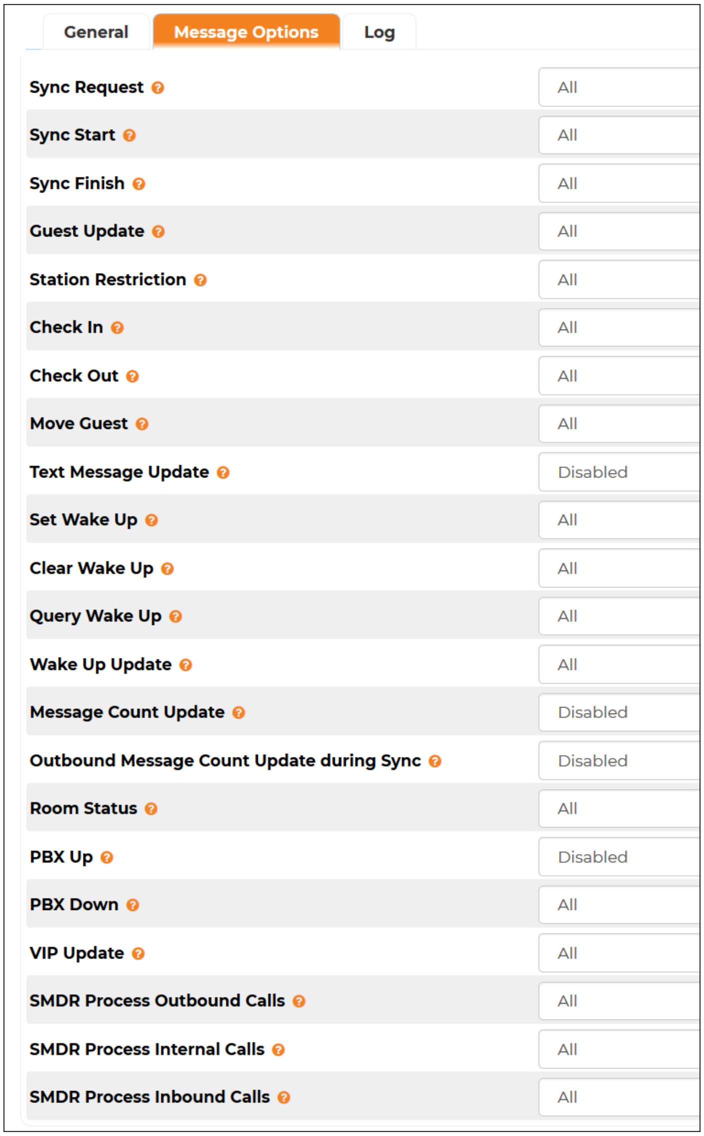
PMS General Tab

In the General tab you can enable the interface and setup the Protocol and Connection settings



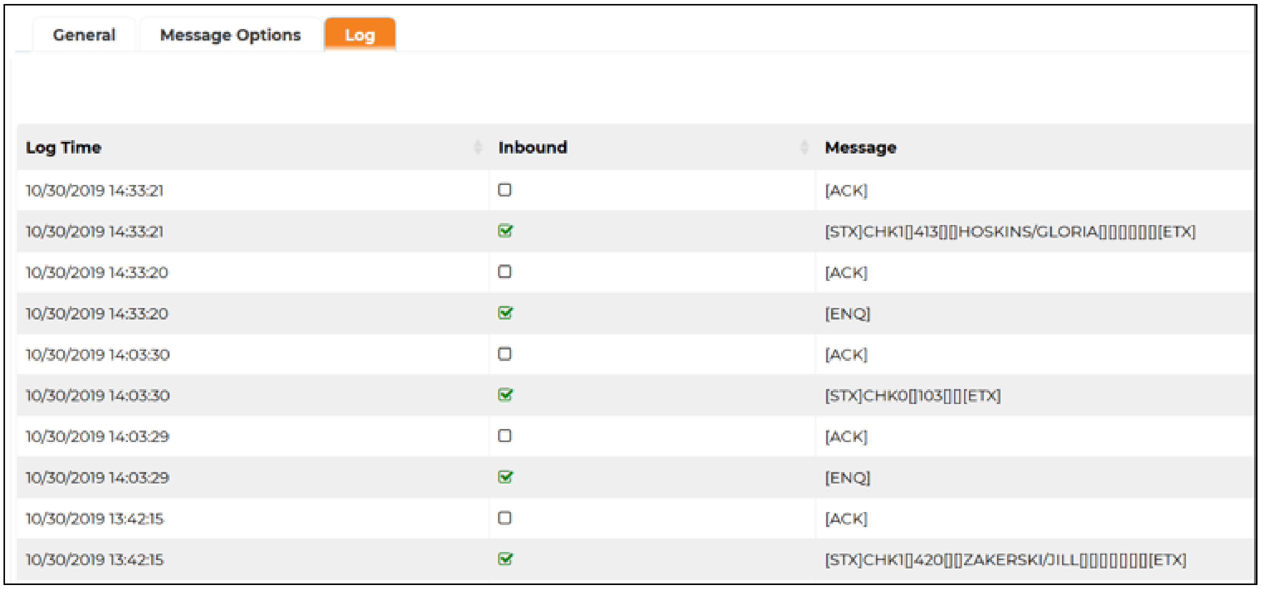
PMS Message Options Tab

Message options are the types of data "messages" that make up the language sent between the PBX and the PMS systems interface so they can communicate. Some PMS systems do not understand all the messages which can cause communication problems. In this tab you can choose which message are enabled, disabled, or if they are expected to be inbound or outbound messages.



PMS Log Tab

The PMS Log will show information that is exchanged between the PMS and the ComXchange Phone System. When the Interface is started the ComXchange PMS Interface will send out an ENQ message and will expect to receive an ACK message from the PMS System. This acknowledgement is the first sign the 2 sides are communicating.



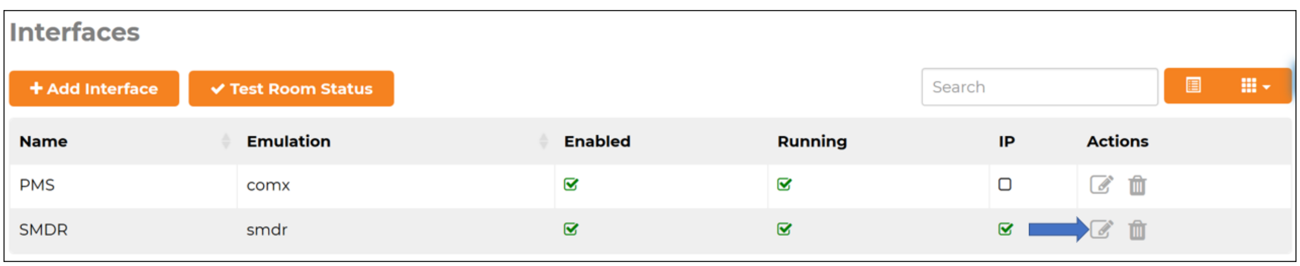
General | Message Options | Log

Log Time	Inbound	Message
10/30/2019 14:33:21	☐	[ACK]
10/30/2019 14:33:21	☑	[STX]CHK1▯413▯▯HOSKINS/GLORIA▯▯▯▯▯▯▯[ETX]
10/30/2019 14:33:20	☐	[ACK]
10/30/2019 14:33:20	☑	[ENQ]
10/30/2019 14:03:30	☐	[ACK]
10/30/2019 14:03:30	☑	[STX]CHK0▯103▯▯[ETX]
10/30/2019 14:03:29	☐	[ACK]
10/30/2019 14:03:29	☑	[ENQ]
10/30/2019 13:42:15	☐	[ACK]
10/30/2019 13:42:15	☑	[STX]CHK1▯420▯▯ZAKERSKI/JILL▯▯▯▯▯▯▯▯[ETX]

SMDR Interface (Call Accounting)

The SMDR interface monitors the Station Message Detail Recording (SMDR) records and captures detailed information about incoming and outgoing voice and data calls. This information is sent to the call accounting server for analysis. If Call Accounting determines there is a charge for a call the charge information is sent out the CAS Serial port to the PMS.

Navigate to Guest Management > Interfaces – Click on the SMDR edit Actions icon.

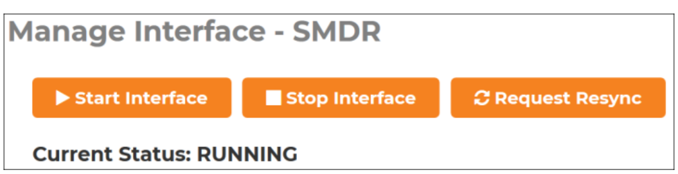


Interfaces

+ Add Interface | ✓ Test Room Status | Search

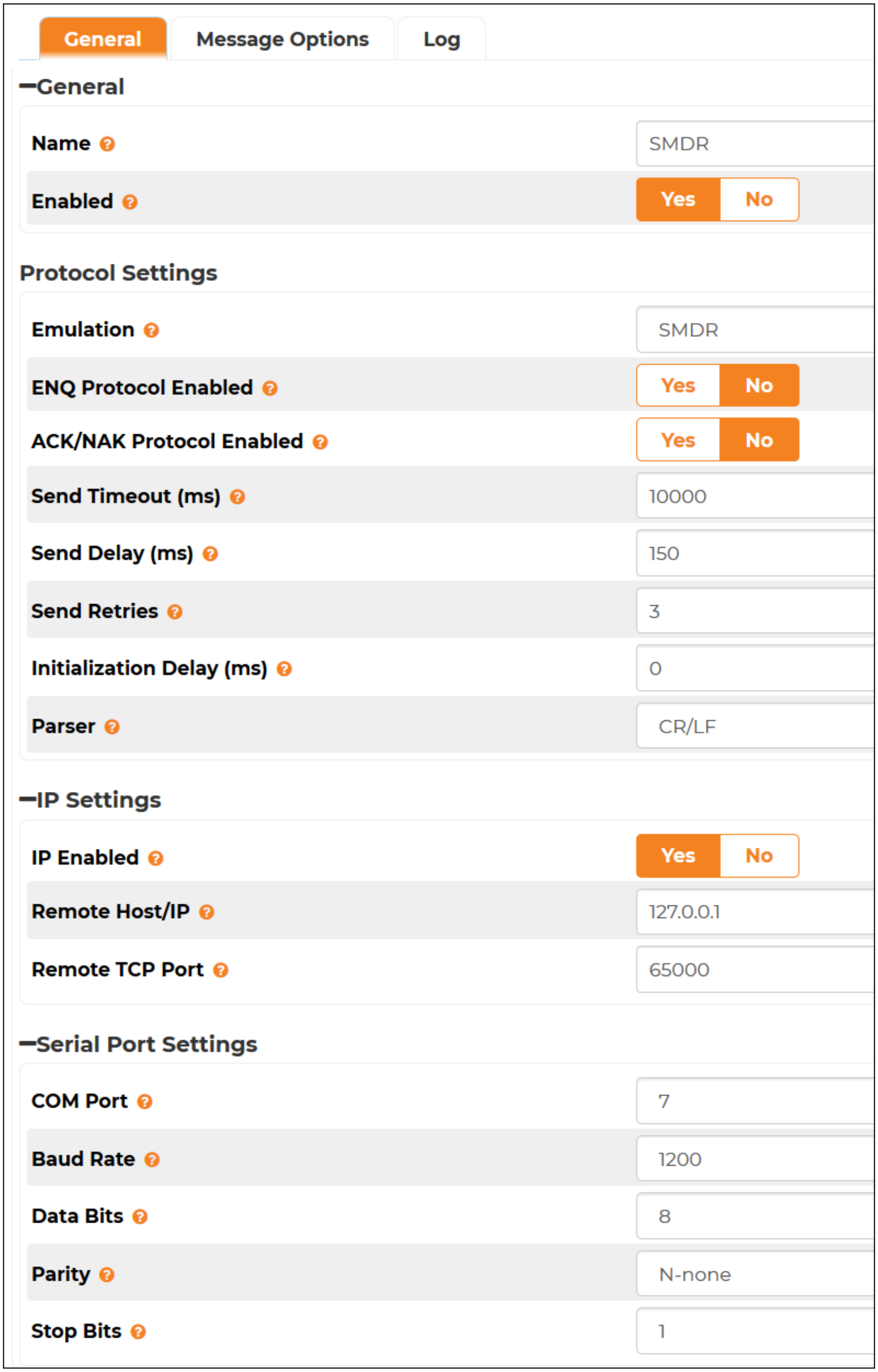
Name	Emulation	Enabled	Running	IP	Actions
PMS	comx	☑	☑	☐	
SMDR	smdr	☑	☑	☑	

Here you can Start and Stop the interface or Request a Resync



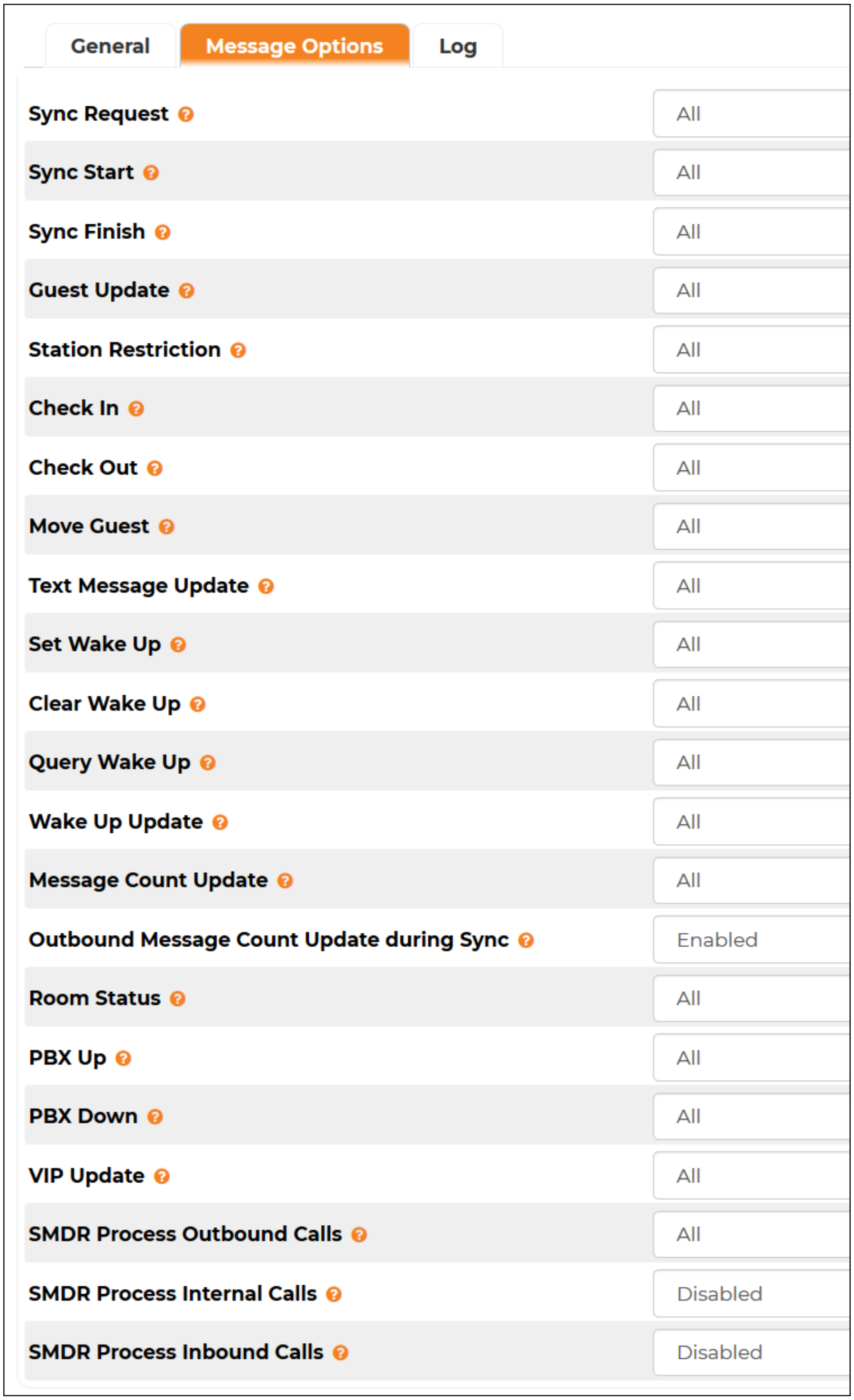
SMDR General Tab

In the General tab you can enable the interface and setup the Protocol and Connection settings



SMDR Message Options Tab

Message options are the types of data "messages" that make up the language sent between the PBX and the PMS systems interface so they can communicate. Some PMS systems do not understand all the messages which can cause communication problems. In this tab you can choose which message are enabled, disabled, or if they are expected to be inbound or outbound messages.



SMDR Log Tab

The SMDR Interface Logs show you the information that the Call Accounting Server received from the PBX. Only when the Call Accounting Server receives a SMDR that has a charge is it sent to the PMS System over the CAS serial port.

Log Time	Inbound	Message
10/30/2019 17:01:23	▯	502▯▯▯▯▯▯▯▯▯▯▯▯▯10/30/2019▯16:57▯00:03:59▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯18136[illegible]▯▯▯▯▯▯▯▯▯▯OB[CR][LF]
10/30/2019 16:10:23	▯	501▯▯▯▯▯▯▯▯▯▯▯▯▯10/30/2019▯16:09▯00:00:33▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯1248[illegible]7▯▯▯▯▯▯▯▯▯▯OB[CR][LF]
10/30/2019 16:09:18	▯	501▯▯▯▯▯▯▯▯▯▯▯▯▯10/30/2019▯16:08▯00:00:08▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯1248[illegible]7▯▯▯▯▯▯▯▯▯▯OB[CR][LF]
10/30/2019 15:09:04	▯	501▯▯▯▯▯▯▯▯▯▯▯▯▯10/30/2019▯15:07▯00:01:50▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯12487[illegible]▯▯▯▯▯▯▯▯▯▯OB[CR][LF]
10/30/2019 12:40:51	▯	501▯▯▯▯▯▯▯▯▯▯▯▯▯10/30/2019▯12:40▯00:00:24▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯1248[illegible]7▯▯▯▯▯▯▯▯▯▯OB[CR][LF]
10/30/2019 12:12:14	▯	501▯▯▯▯▯▯▯▯▯▯▯▯▯10/30/2019▯12:11▯00:00:56▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯▯15865[illegible]▯▯▯▯▯▯▯▯▯▯OB[CR][LF]

Serial -To -IP (IPOCKET) Device Settings

Communication Between the PMS System and both the PBX and Call Accounting Interfaces can also take place over a Serial to IP Device. To Configure an IPOCKET Press and hold reset button on IPOCKET to access settings via the serial port of the unit. Set monitoring software on laptop to 9600-N-8-1.

	PMS	Call Accounting
IP Address	192.168.101.11	192.168.101.12
Subnet Mask	255.255.255.0	255.255.255.0
Gateway	192.168.101.1	192.168.101.1
Network Address	0.0.0.0	0.0.0.0
Network Mask	0.0.0.0	0.0.0.0
Gateway	0.0.0.0	0.0.0.0
Protocol	Transparent (tcp)	Transparent (tcp)
Port Setting	9600 bps 8N1 (no)	1200 bps 8N1 (no)
Connection Control	Automatic	Automatic
Local Port	62000	62001
Remote IP	192.168.101.3	192.168.101.3
Remote Port	62000	62001
Fallback IP	0.0.0.0	0.0.0.0
Fallback Port	0	0
Fallback Drop Time	0	0
Packet Prefix	none	none
Max Inter-Char Delay	0	0
Preferred Packet Size	0	0
Initial String	(not set)	(not set)

Hotel Dashboard

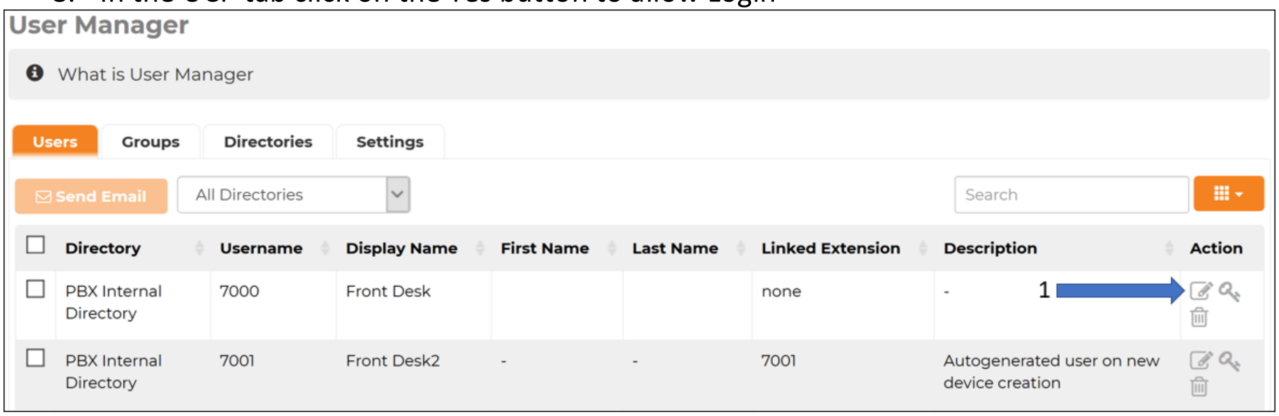
There is a web GUI that staff can use to manage wakeup calls, change a room's calling permissions, and override room status to allow for a checked-out room to place calls. The Hotel Dashboard is accessed through the ComXchange UCP (User Control Panel) with a user that is created in the User Management Module. **Note: ComXchange Admins can access the Hotel Dashboard through Reports > Hotel Dashboard.**

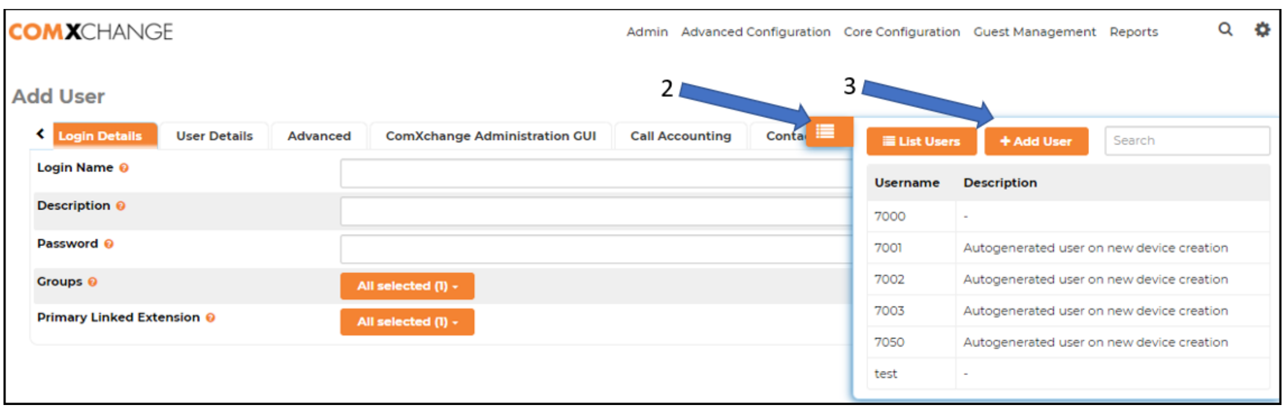
Create a User

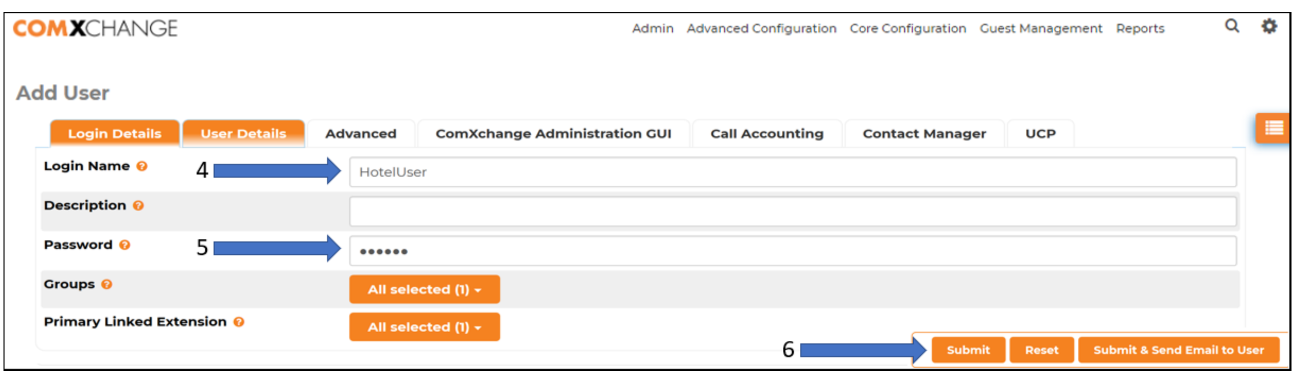
To access the Hotel Dashboard, you need a user that has access to the UCP. While this can be an extension User that is given access in the example below, we will create a User that is dedicated to access the Hotel Dashboard.

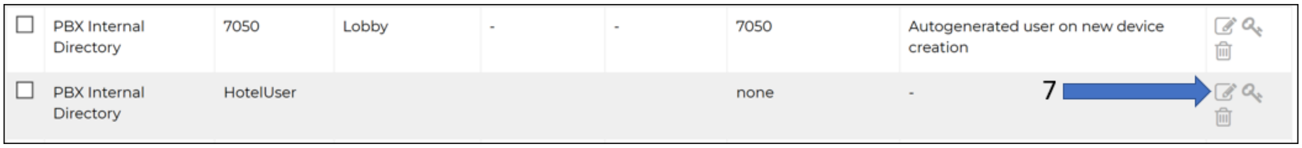
Navigate to Admin > User Management

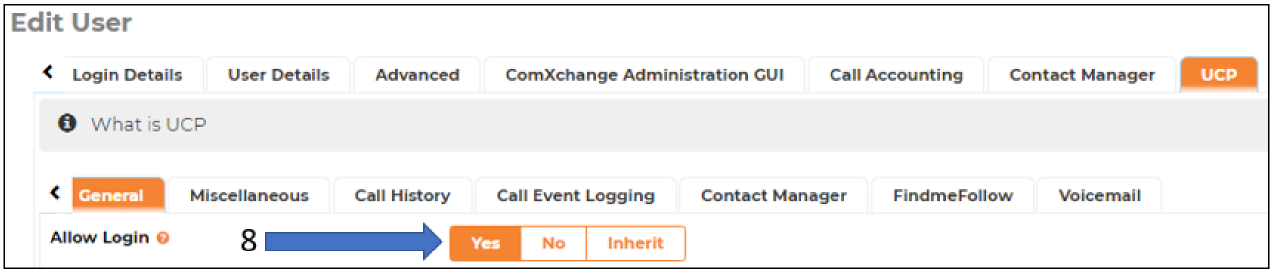
1. Choose any extension and click on the edit Action icon
2. Click on and open the Flyout Menu
3. Click on the Add User Button
4. Fill in the Login Name
5. Fill in a Password
6. Submit and Apply
7. Click on the edit Action icon of the new user
8. In the UCP tab click on the Yes button to allow Login







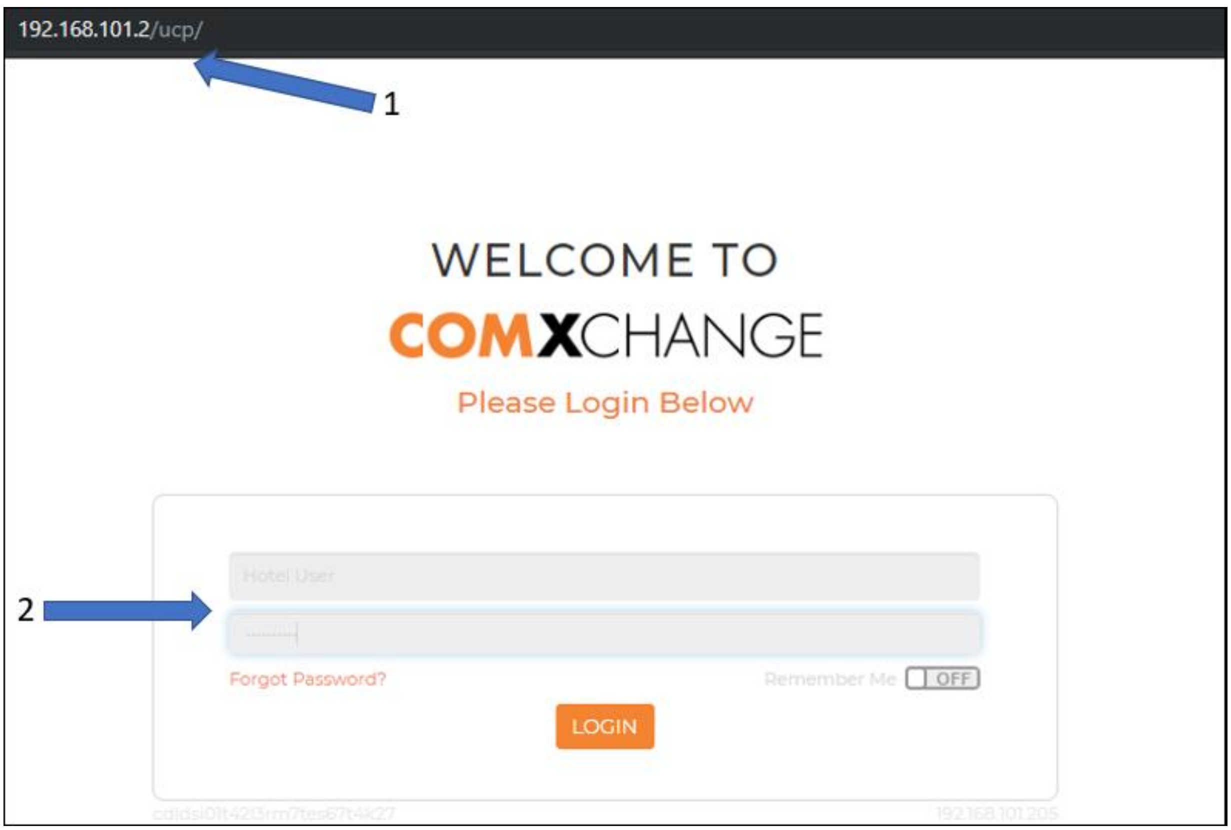


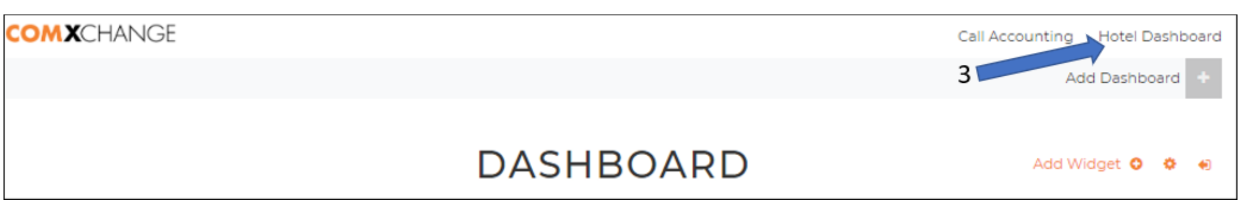


Login into the Hotel Dashboard

You can use the newly created dedicated Hotel Dashboard User to login to the Hotel Dashboard.

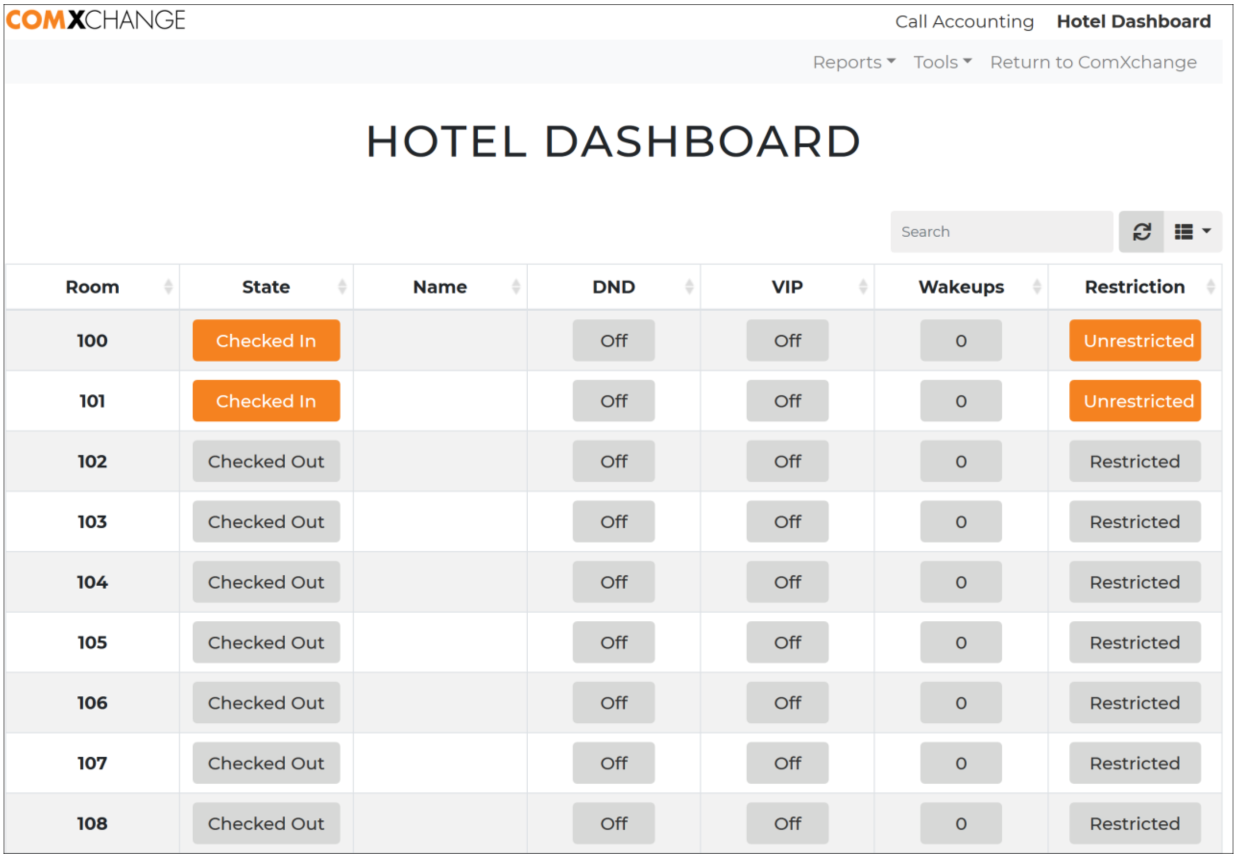
1. Enter http://192.168.101.2/UCP in a browser.
2. Login with the username and password.
3. Click on the Hotel Dashboard Link





Guest Extension Management

In the Guest Extension Management Tool you can verify and change the status of the guest extension state as it appears to ComXchange. This can be helpful when testing guest extension calls. Making changes in the Check-In State for a room with this tool only makes it appear checked in to ComXchange It does not push anything to the PMS.

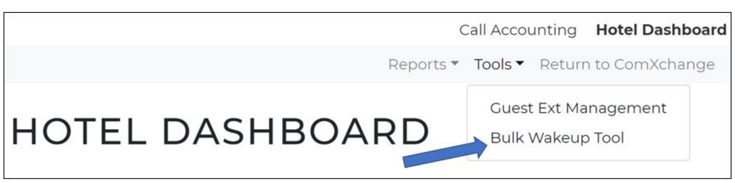


Room	State	Name	DND	VIP	Wakeups	Restriction
100	Checked In		Off	Off	0	Unrestricted
101	Checked In		Off	Off	0	Unrestricted
102	Checked Out		Off	Off	0	Restricted
103	Checked Out		Off	Off	0	Restricted
104	Checked Out		Off	Off	0	Restricted
105	Checked Out		Off	Off	0	Restricted
106	Checked Out		Off	Off	0	Restricted
107	Checked Out		Off	Off	0	Restricted
108	Checked Out		Off	Off	0	Restricted

Bulk Wakeup Tool

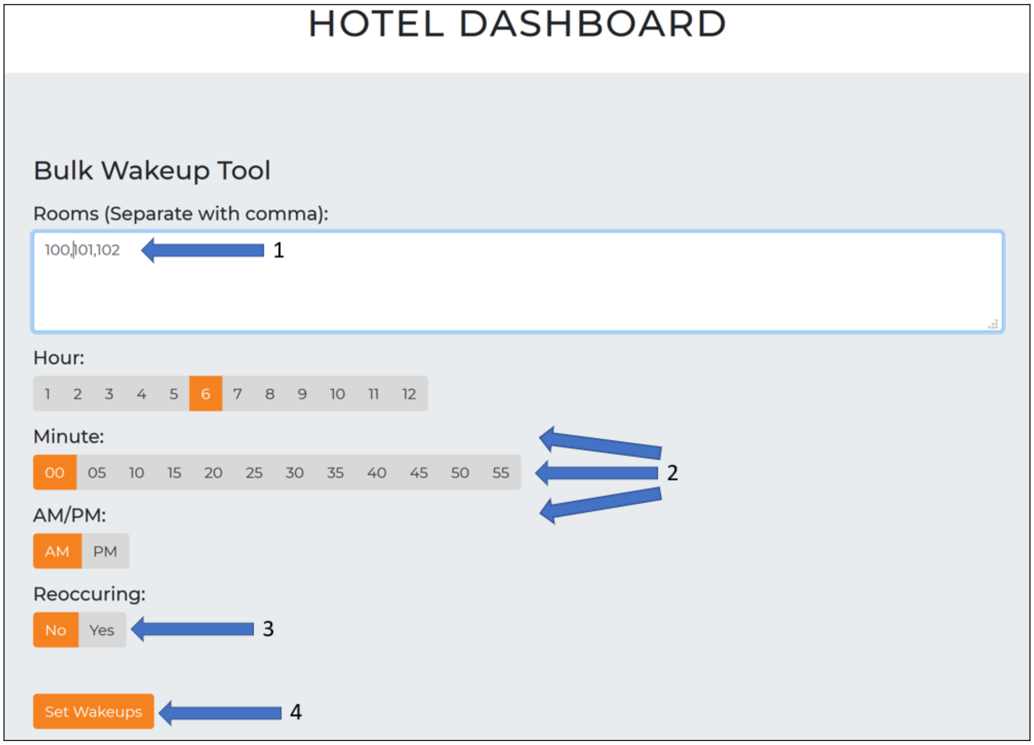
The Bulk Wakeup tool Allows you to create wakeup calls for a single or multiple guests.

Navigate to Tools > Bulk Wakeup Tool



Set Wakeup Call

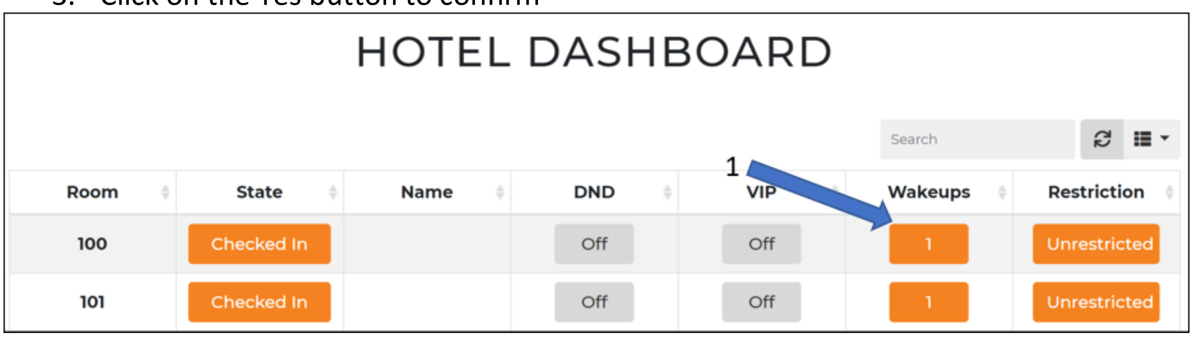
1. Enter the Room(s) number(s)
2. Set the time for the wakeup call
3. Choose if this should be reocurring for the guests stay
4. Click on Set Wakeups

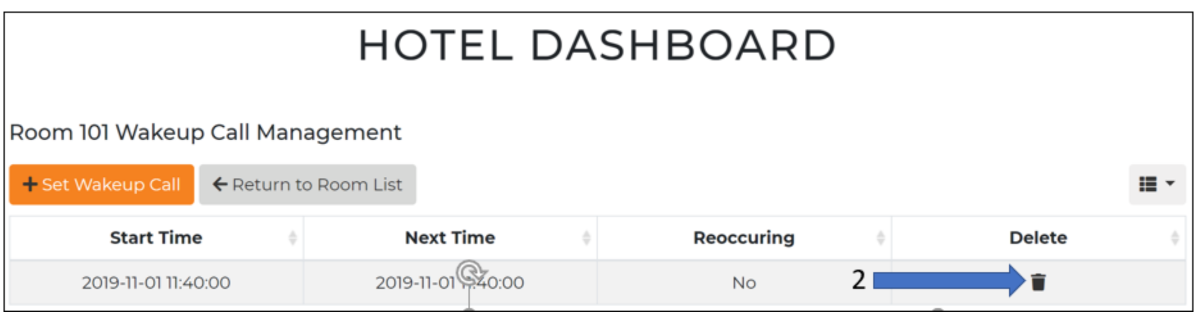


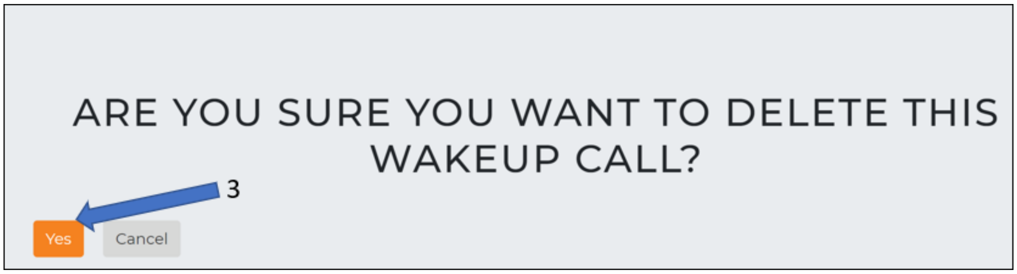
Remove Wakeup Call

Navigate to Tools > Guest Extension Management

1. Click on the Wakeup Call Button for the extension you want the wakeup call removed
2. Click on the Delete icon
3. Click on the Yes button to confirm







Guest Reports

In the Guest Tool Console there is a reports menu where a user can view reports on Emergency Notification Calls that have been placed and a report of Wakeup Calls that are Pending, Completed, or Failed.

Emergency Notification Report

HOTEL DASHBOARD

Emergency Notfication Calls Report

Search

Notification Call Time	Emergency Call Info	Attempt	Result
2019-10-25 09:17:56	Ext 7100 called 9911 (2019-10-25 09:17:24)	1 of 5	Emergency Notification call WAS Answered and ACKNOWLEDGED
2019-10-16 17:09:01	Ext 7002 called 9911 (2019-10-16 17:08:47)	1 of 5	Emergency Notification call WAS Answered and ACKNOWLEDGED
2019-10-16 17:02:20	Ext 7002 called 9911 (2019-10-16 17:01:59)	1 of 5	Emergency Notification call WAS Answered and ACKNOWLEDGED

Wakeup Reports

Pending Wakeup Calls Report

Search

Room	Wakeup Start Time	Next Attempt Time	Attempt	Reoccuring	Status
100	2019-11-01 12:45:00	2019-11-01 12:45:00	0 of 1	No	Waiting

Completed Wakeup Calls Report

Search

Log Time	Room	Wakeup Start Time	Attempt	Reoccuring	Result
2019-11-01 12:15:00	101	2019-11-01 12:15:00	1 of 1	No	Wakeup call was NOT Answered
2019-11-01 12:10:24	100	2019-11-01 12:10:00	1 of 1	No	Wakeup call WAS Answered

Completed Wakeup Calls Report

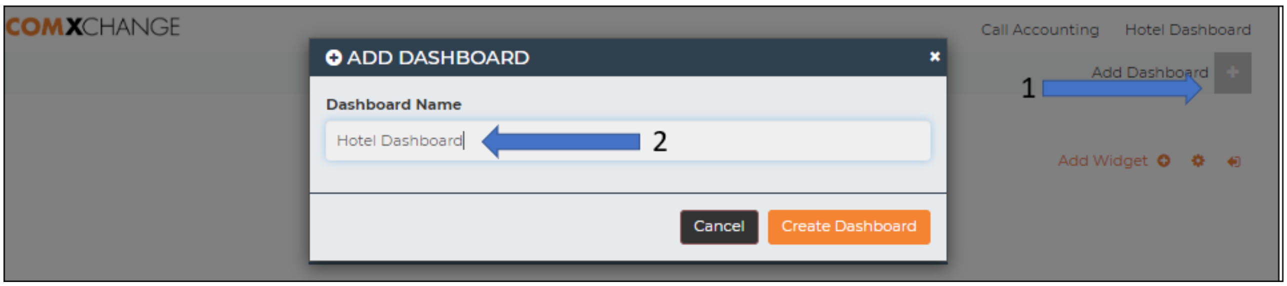
Search

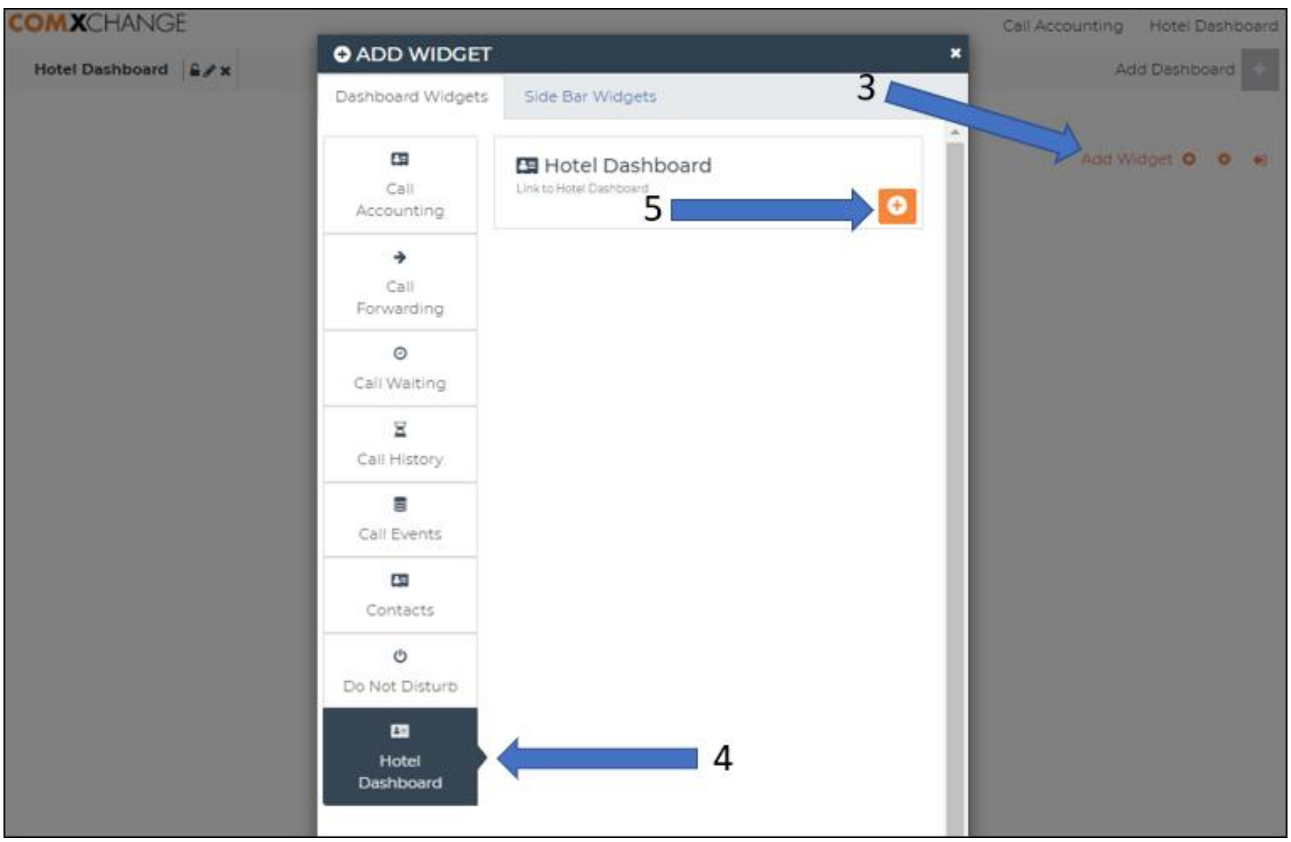
Log Time	Room	Wakeup Start Time	Attempt	Reoccuring	Result
2019-11-01 12:15:00	101	2019-11-01 12:15:00	1 of 1	No	Wakeup call was NOT Answered
2019-11-01 12:10:24	100	2019-11-01 12:10:00	1 of 1	No	Wakeup call WAS Answered

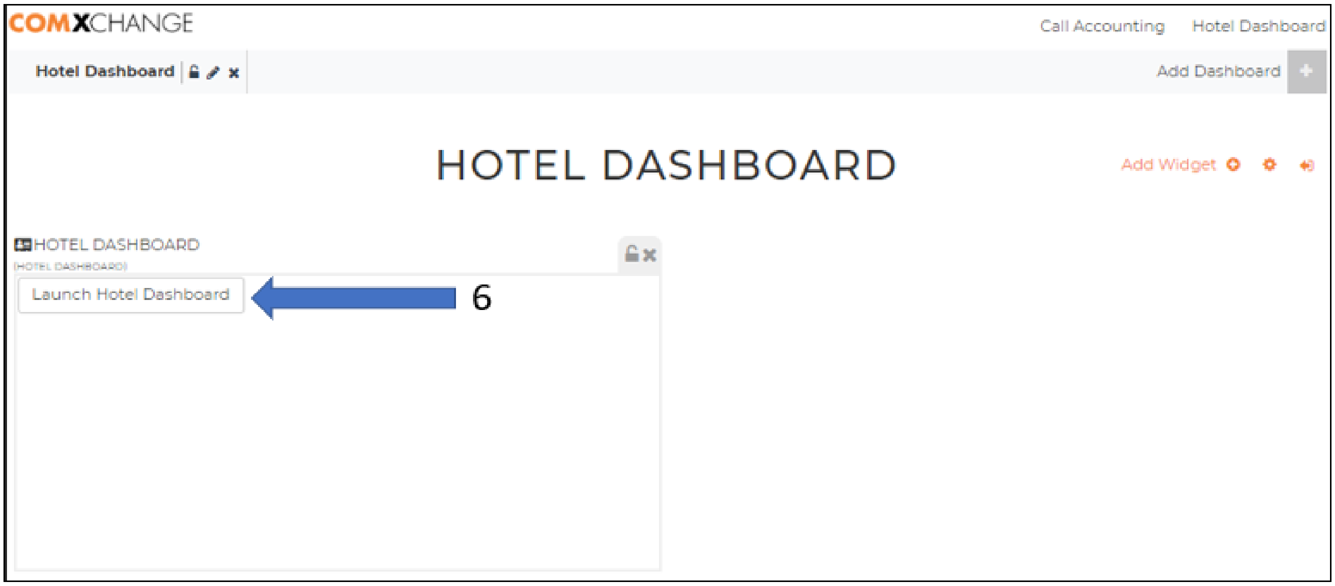
Hotel Dashboard Widget

You can launch the Hotel Dashboard through a widget created in the UCP

1. Click the Add Dashboard Button
2. Name the Dashboard
3. Click the Add Widget Button
4. Click on the Hotel Dashboard Widget
5. Click the Add button
6. Click the Launch Hotel Dashboard Button







Attendant Console

Models of Aastra and Yealink Phones can be programmed to be Attendant Console Phones. The Console can Pick-up, Park, Transfer calls, manage Night Mode, etc. It is also possible to manage Wakeup Calls as well as Guest Room Moves, DND Settings, and Call Restrictions. There are default Front Desk Phone Templates available on the ComXchange server that have many of these features configured in the Keys section. These can be set up by adding keys to other phone templates as well.

Attendant Console Keys Examples

Line Key 3 Type	XML Browser
Line Key 3 Line	Line 1
Line Key 3 Label	Wakeup
Line Key 3 Value	http://192.168.101.2/yl/wakeup.php
Line Key 3 Pickup Number	{$pickup_value}
Line Key 4 Type	XML Browser
Line Key 4 Line	Line 1
Line Key 4 Label	Guest Management
Line Key 4 Value	http://192.168.101.2/yl/guestmgmt.php
Line Key 4 Pickup Number	{$pickup_value}
Line Key 5 Type	BLF
Line Key 5 Line	Line 1
Line Key 5 Label	Mod
Line Key 5 Value	7002
Line Key 5 Pickup Number	{$pickup_value}

Line Key 6 Type	BLF
Line Key 6 Line	Line 1
Line Key 6 Label	Park
Line Key 6 Value	70
Line Key 6 Pickup Number	{$pickup_value}

Line Key 7 Type	BLF
Line Key 7 Line	Line 1
Line Key 7 Label	Park 1
Line Key 7 Value	71
Line Key 7 Pickup Number	{$pickup_value}

Line Key 8 Type	BLF
Line Key 8 Line	Line 1
Line Key 8 Label	Park 2
Line Key 8 Value	72
Line Key 8 Pickup Number	{$pickup_value}

Line Key 9 Type	BLF
Line Key 9 Line	Line 1
Line Key 9 Label	Night
Line Key 9 Value	*280
Line Key 9 Pickup Number	{$pickup_value}

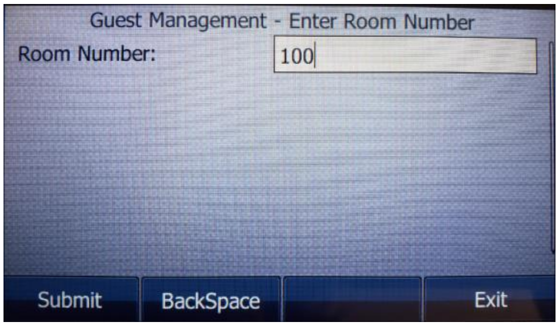
Guest Console TUI (Text-Based User Interface) Phone Applications

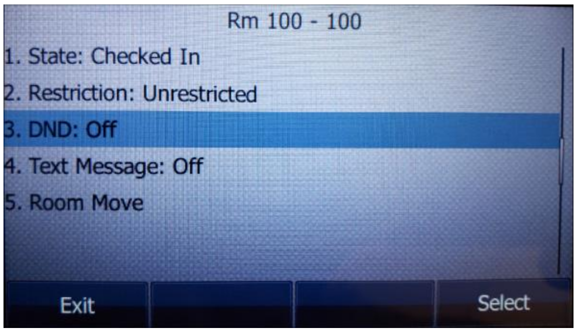
There are two hospitality applications that are used by the Attendant Console. The Guest Management App and the Guest Wakeup App.

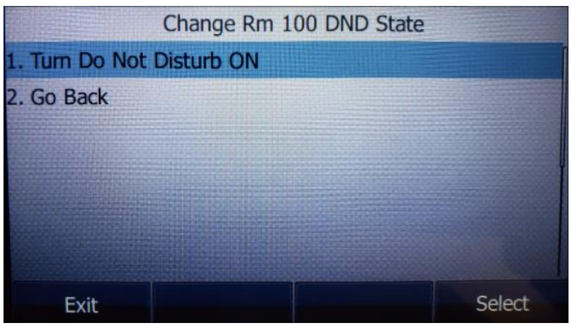
Guest Management App

1. Press the Guest Management App Button

2. Enter a Room Number - press Submit
3. Choose an action, for instance press 3 for DND - press Select
4. Select by number, DND On or Off then press Select

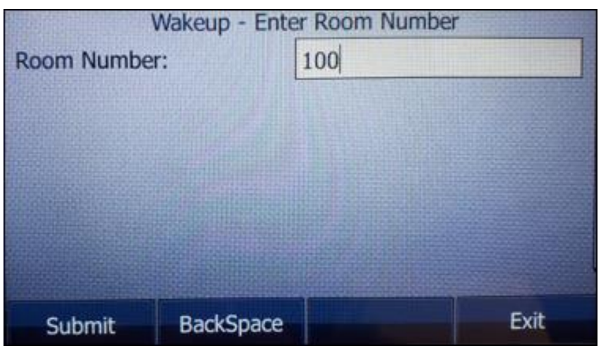


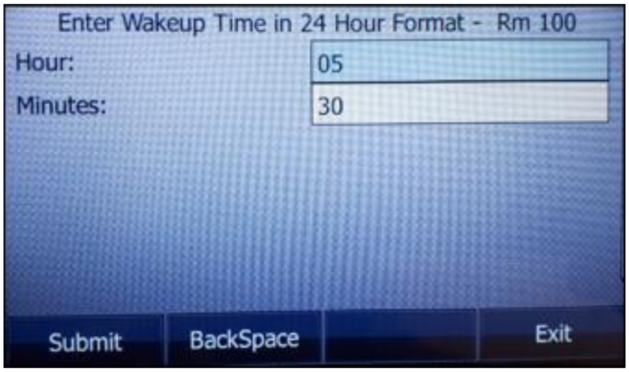


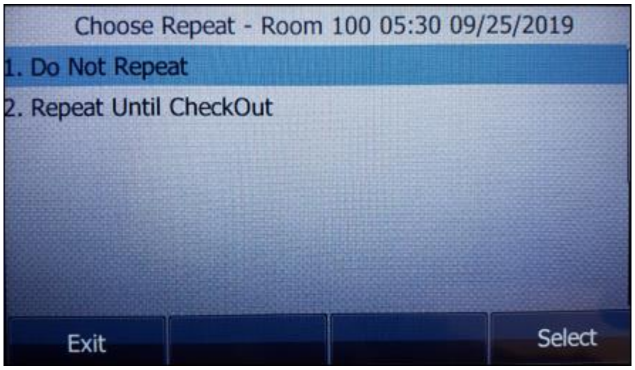


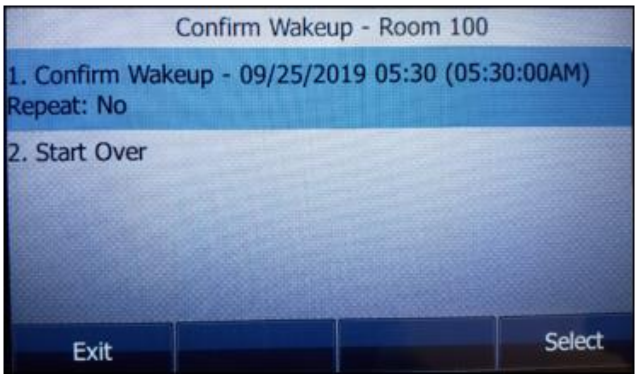
Wakeup App

1. Push the Wakeup App
2. Enter a Room Number - press Submit
3. Enter a Wakeup Time - press Submit
4. Choose to Repeat or Do Not Repeat - press Submit
5. Choose Confirm or Start Over – press Submit









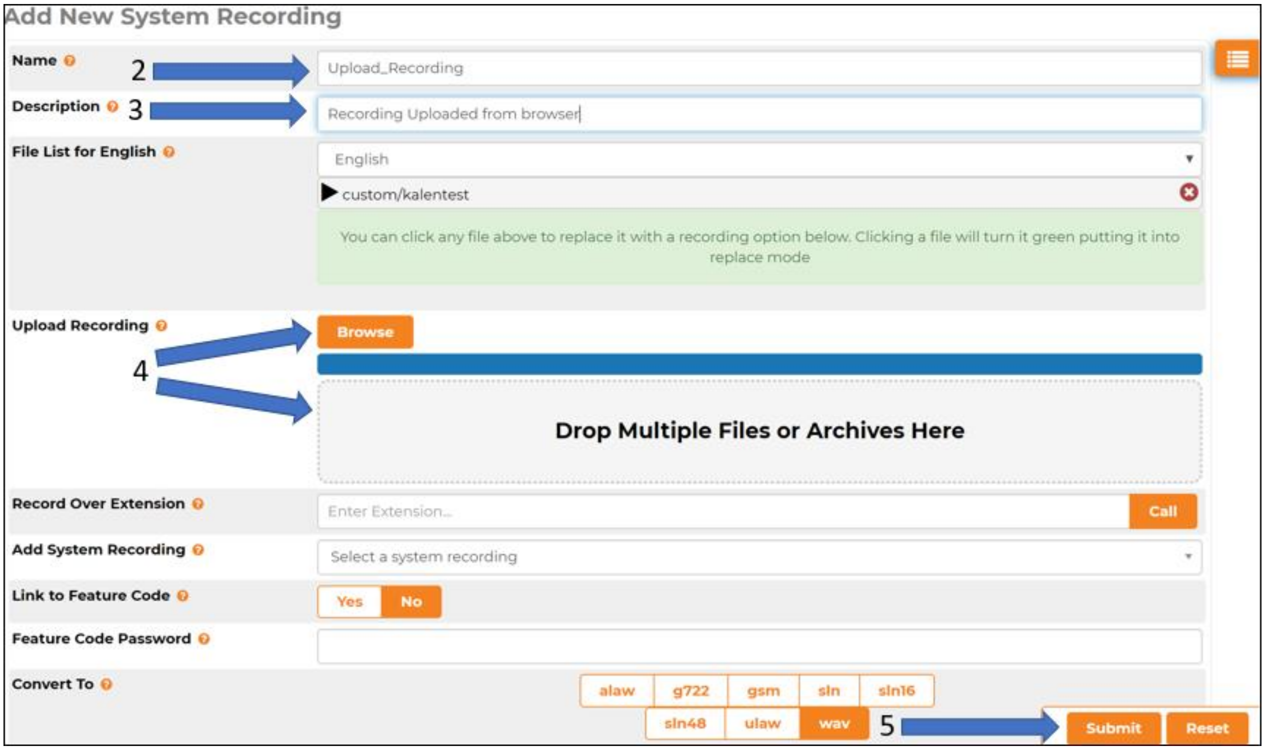
System Recordings/Prompts

The System Recoding Module allows the creation of prompts or announcements from any phone or an uploaded file. If uploading a file, it must be in the correct format. [Wav PCM Encoded, 16 bits, at 8000Hz]. You can enable a feature code on a recording to allow easy access to re-record.

Navigate to Advanced Configuration > System Recordings

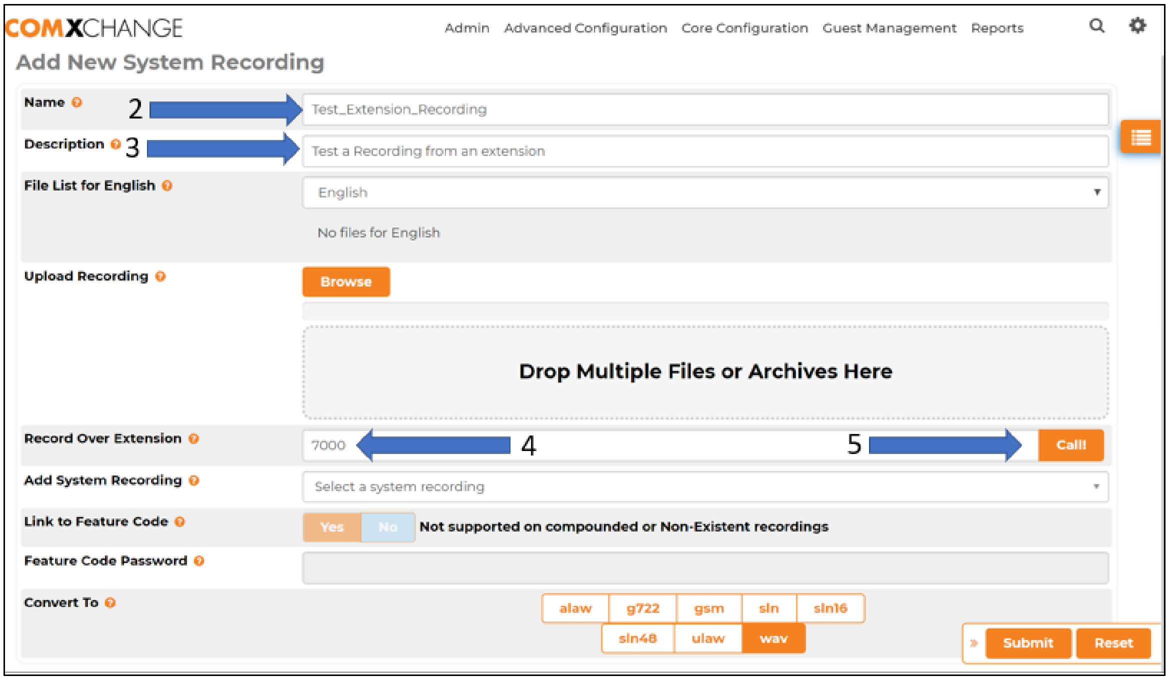
Upload a Recording

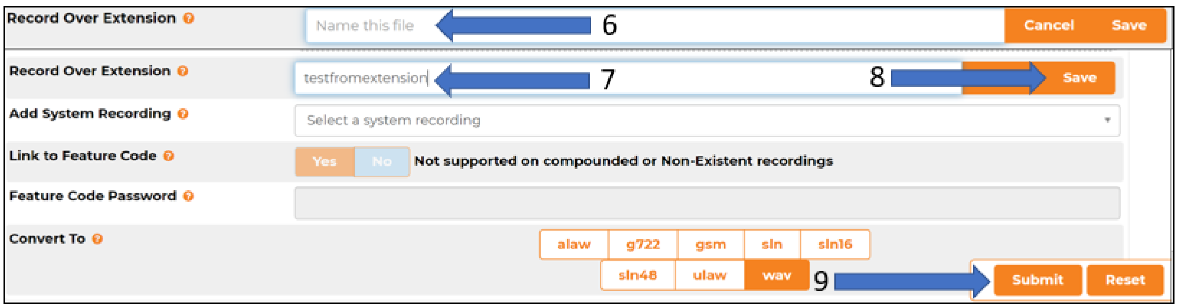
1. Click on the Add Recording Button
2. Enter a Name for the recording
3. Enter a Description of the recording
4. Upload the Recording.
    a. Drag the File to the Browser or Browse to the File by Clicking Browse
5. Submit, Click OK on the Popup, and Apply



Record Over an Extension

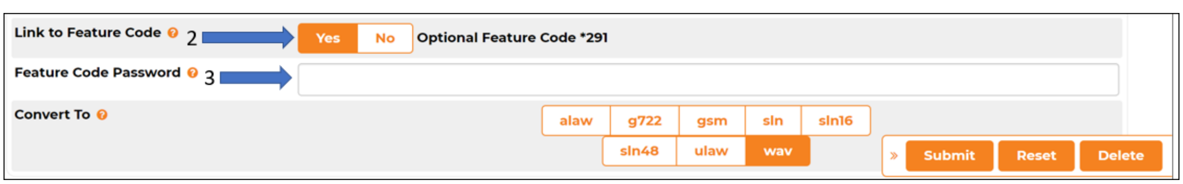
1. Click on the Add Recording Button
2. Enter a Name for the recording
3. Enter a Description of the recording
4. Enter extension 7000 to set a recording from a phone extension
5. Click on Call
    a. The phone System will Call the Extension and beep
    b. Record the Message and Hang up
6. The Record Over Extension Field changes to “Name this file”
7. Enter a name for the file
8. Click Save
9. Click Submit and Apply





System Recording Feature Code

1. Click on the Edit Actions Button of the Recording you want to assign a Feature Code
2. Link the Feature Code by selecting Yes
3. Enter a password for the Feature Code if desired



You can now dial the feature code to access the recording. This recording is now available to be used by IVRs, Queues, or Announcements.

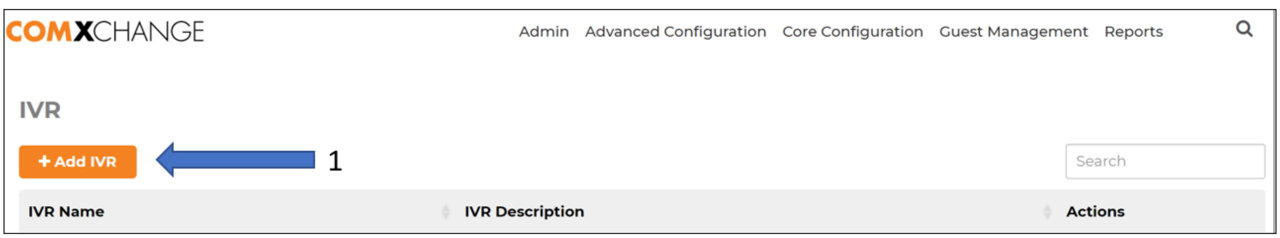
IVR Setup

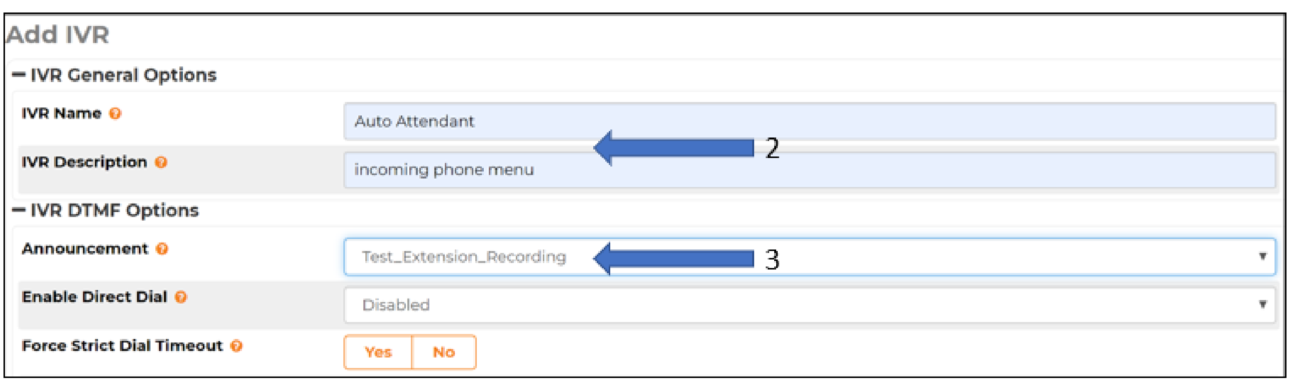
Interactive Voice Response (IVR) is a technology which allows callers to navigate a phone system before talking to an operator or other destinations on the phone server. An IVR can be set to other system modules and can be nested. The IVR can be linked to a directory to allow selective extension dialing. You can choose in the IVR menu what happens if a wrong key is pressed or if a caller doesn't press a key within a timeout period. Below we will set up an auto attendant IVR that allows a caller to choose to go to the operator leave a voicemail or be directed to extension 7000. For this to work you would need to record a greeting that offers these options to the caller.

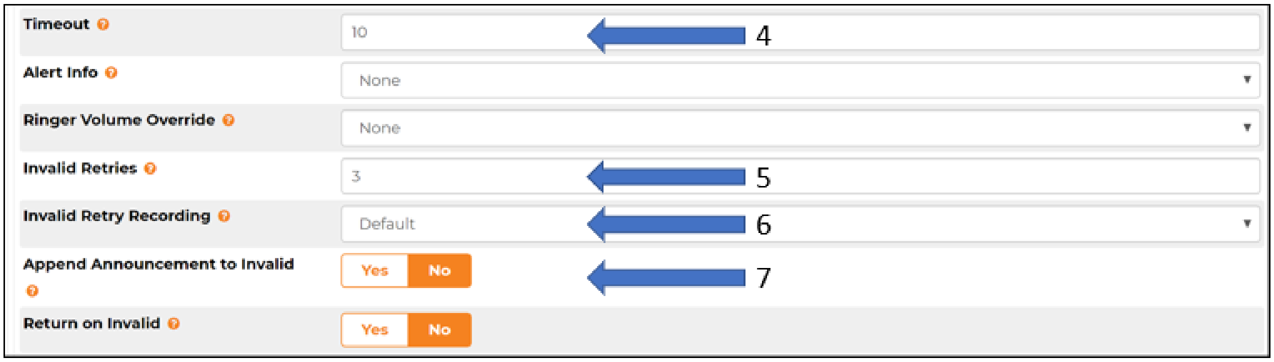
Create The IVR

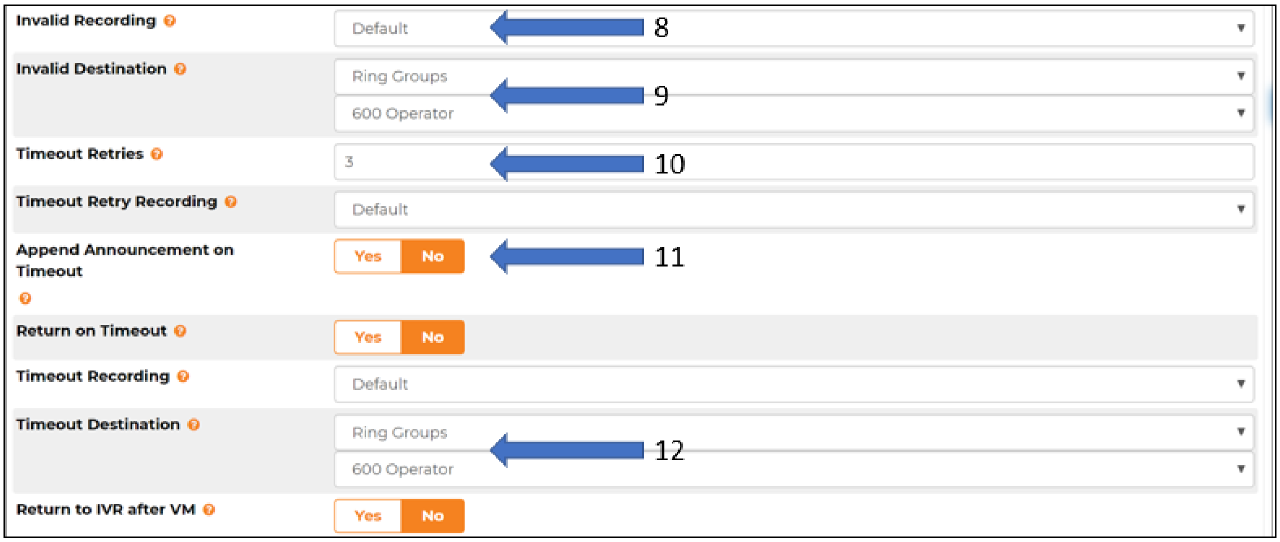
Navigate to Core Configuration > IVR.

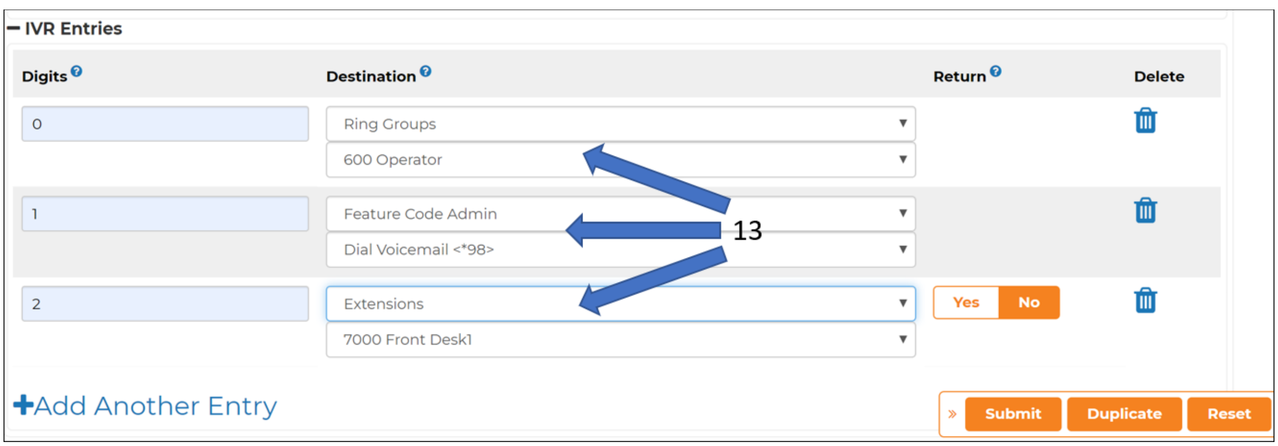
1. Click on the Add IVR Button
2. Enter an IVR name and description
3. Choose the recording to use as an announcement
4. Set the amount of time with no key press to be considered a timeout
5. Set your invalid retries, the number of invalid key presses allowed before going to Invalid Destination
6. Choose a recording to play when an invalid key is pressed
7. Choose to play the IVR announcement again when an invalid key is pressed
8. Choose a recording to play before sending a caller to an invalid destination
9. Choose a destination to send the caller when the number of invalid retries is reached
10. Choose the number of times out retries before sending the caller to the timeout destination
11. Choose a recording to play if a caller reaches a timeout
12. Choose whether to play the IVR the main announcement after a timeout
13. Choose a destination to send callers after a timeout limit is reached
14. Declare your IVR entries









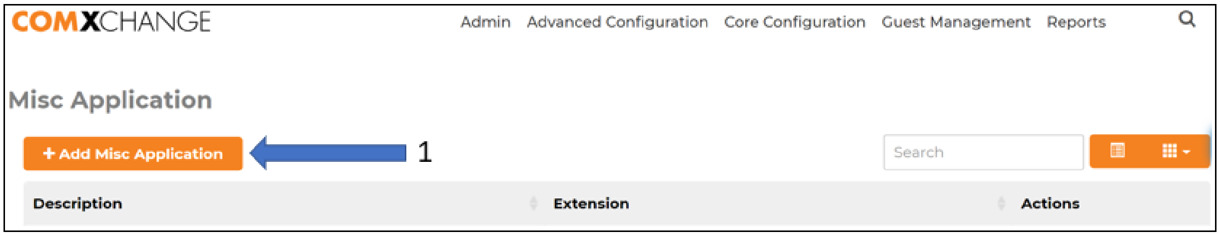


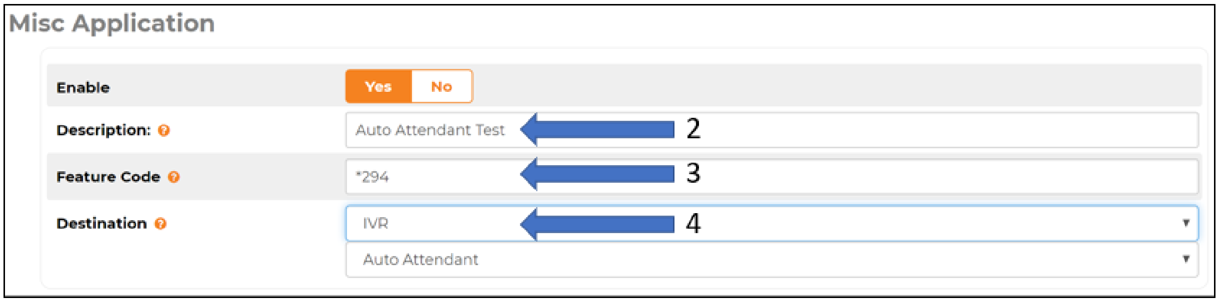
Create a Misc Application to the IVR

Create a new Misc Application to point to the Auto Attendant IVR.

Navigate to Advanced Configuration > Misc Applications

1. Click on the Add Misc Application Button
2. Fill in the Description
3. Fill in a Feature Code Number
4. Choose the Auto Attendant IVR for the destination



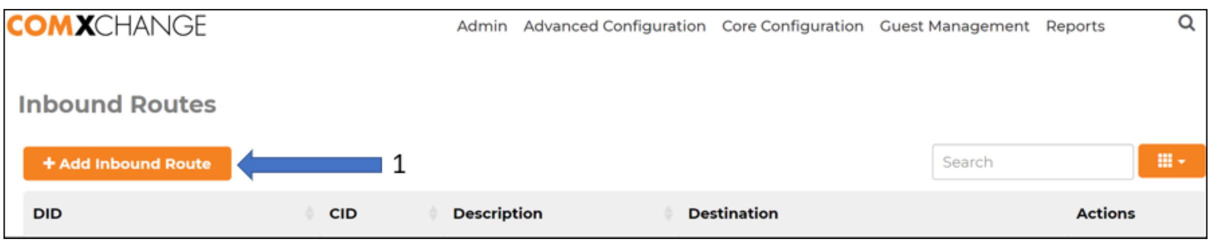


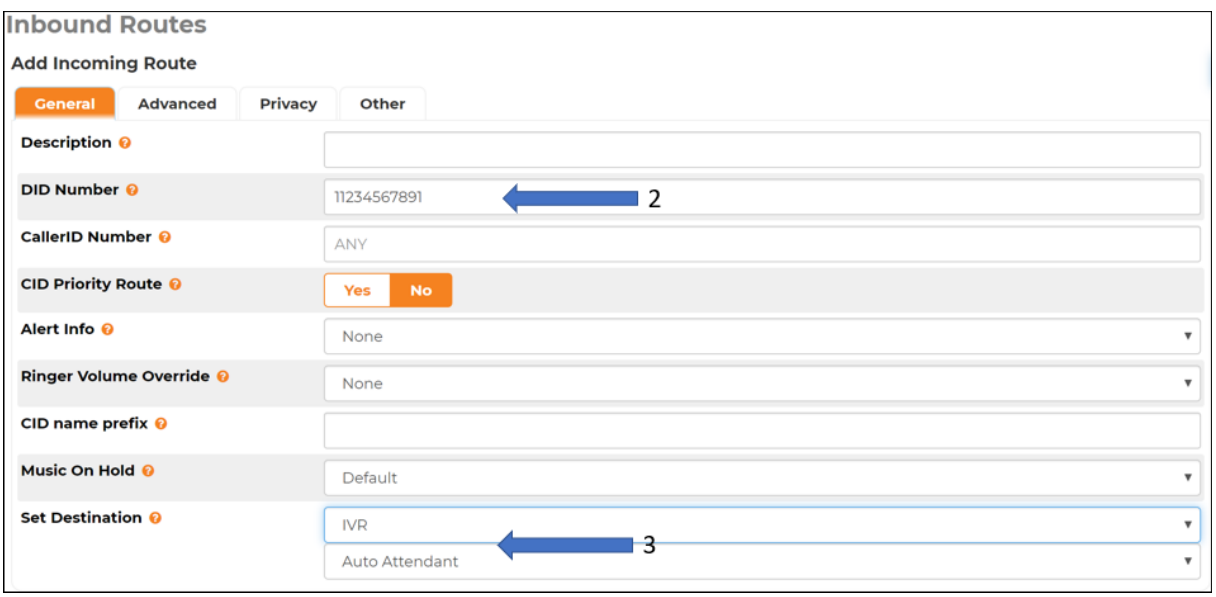
Call the Feature code and test the IVR's functions.

Point Inbound Calls to Auto Attendant IVR

Navigate to Core Configurations > Inbound Routes

1. Click on Add Inbound Route
2. Enter the DID number to match
3. Set the destination to the new IVR
4. Submit and Apply Config





Advanced Call Routing

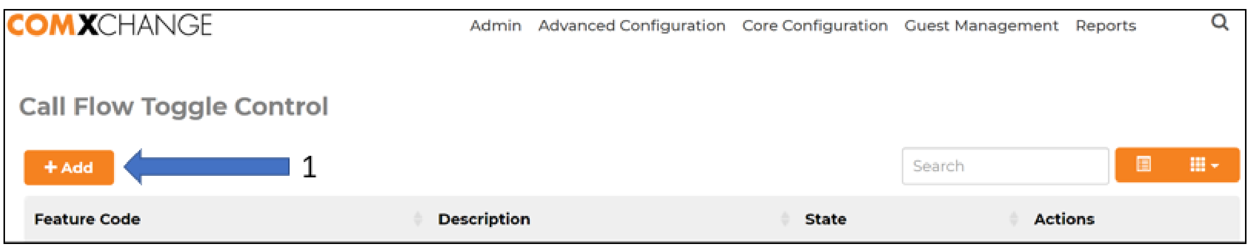
In addition to an IVR there are ways to route calls based on conditions. This can be a manual change that is managed by a user to force calls between 2 destinations such as using a Call Flow Control, or you can route calls based on time and date by using a Time Group and Time Conditions.

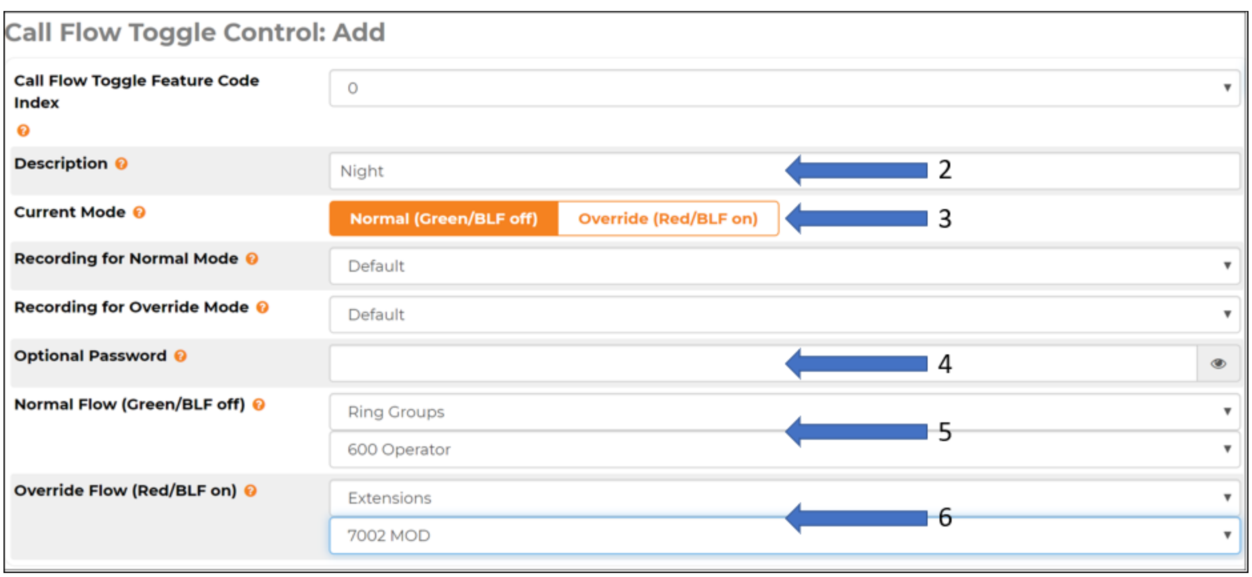
Call Flow Control

A Call Flow Control (CFC) creates feature code that can toggle between two destinations. The feature code is *28 followed by an instance number beginning with 0 so the first CFC will be *280. There can be 10 CFC instances the last being *289. The CFC can be enabled by directly dialing the feature code or it can be enabled by a using a BLF Button.   A hotel can use this for their front desk which they may want to ring different or additional extensions in the evening then just the front desk ring group used during the day.   Below we will create a CFC that will toggle between the Operator ring group and the MOD extension.

Navigate to Advanced Configuration > Call Flow Control

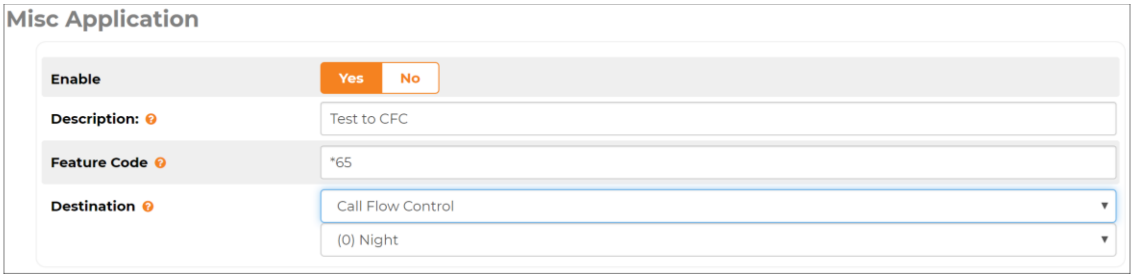
1. Click on Add
2. Fill in the Description Field. (Night)
3. Select the Normal (Green/BLF off) Button
4. Enter a password if you would like to protect the CFC mode.
5. Choose the destination of the normal call flow. (Operator)
6. Choose the destination of the override. (MOD)
7. Click on Save and Apply Config





Test Night Mode

You can test activating and deactivating the CFC by dialing *280. You can test the functionality of the CFC by creating a Misc Application that points to the Night CFC then dial the feature code number. While Night mode is activated and deactivated.



Point Inbound Route to Night Mode

Once the Night mode call flow control has been tested you can point an inbound route to the Night Mode CFC.

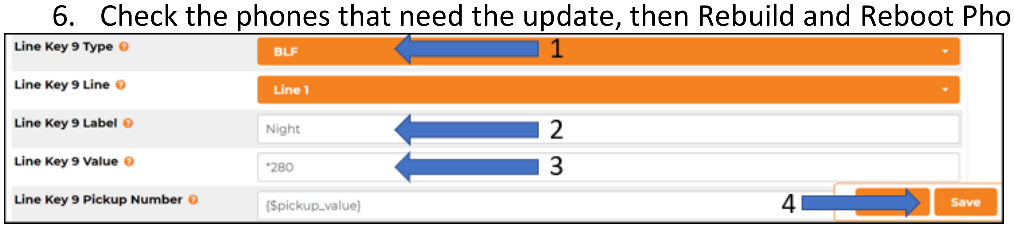
Add Night Mode CFC BLF

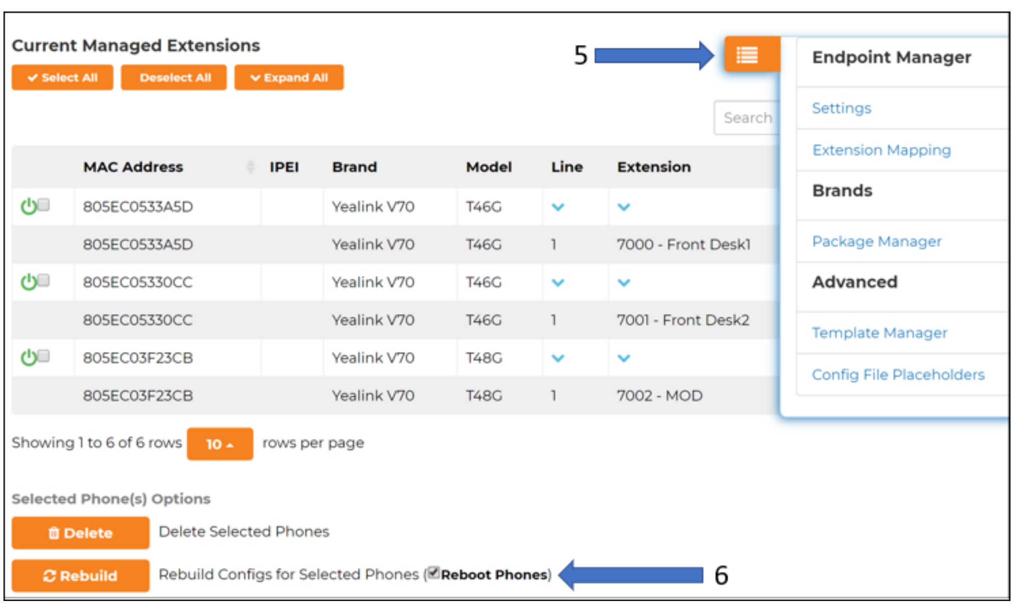
A BLF button can be added to a SIP phone to toggle the night Mode CFC on and off.

Navigate to Devices > Endpoint Template Manager.

Click on Edit for your phone template then click on The Keys Tab to add a BLF named Night.

1. Type will be BLF
2. Label will be Night
3. Value will be *280
4. Click on Save
5. Click Extension Mappings in the Fly Away Menu
6. Check the phones that need the update, then Rebuild and Reboot Phones





During Normal Flow the Night mode BLF button and label will be lit up red. During Override mode the button light will be off and the label will have a green light. Not all phones have labels that will behave this way.

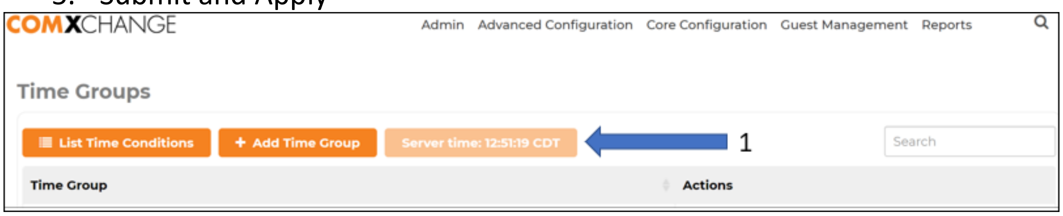
Time Based Routing

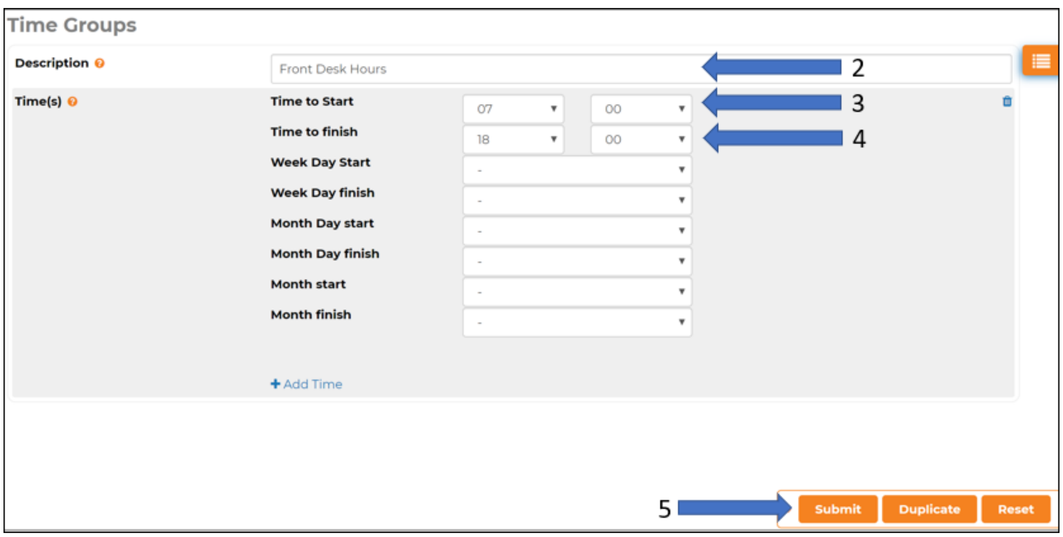
Calls coming into the ComXchange Controller can be routed to different destinations based on Time Conditions. Time Conditions work together with Time Groups or Calendars to define a time that calls will be routed to one place or another. For instance, this may be set to dial the Operator Ring Group between the hours of 7 AM to 6 PM then dial the MOD phone outside of those times.

Time Group

A Time Group defines times to be used in a Time Condition. Set up a Time Group defining Front Desk hours
Navigate to Advanced Configuration > Time Groups

1. Click Add Time Group
2. Fill in the Description Field
3. Specify the Time to Start
4. Specify the Time to Finish
5. Submit and Apply



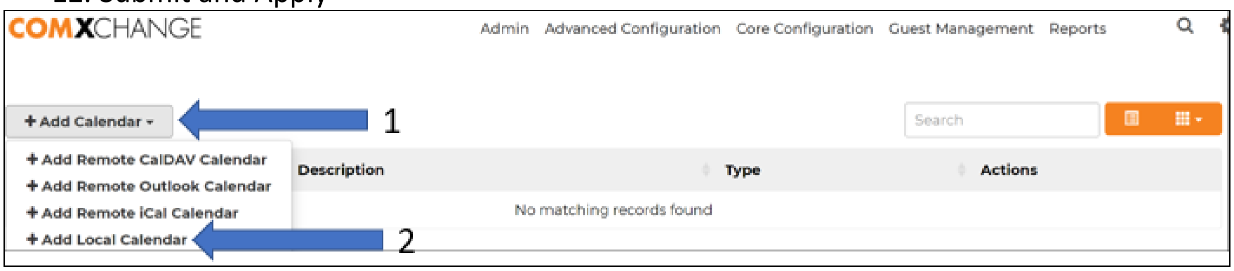


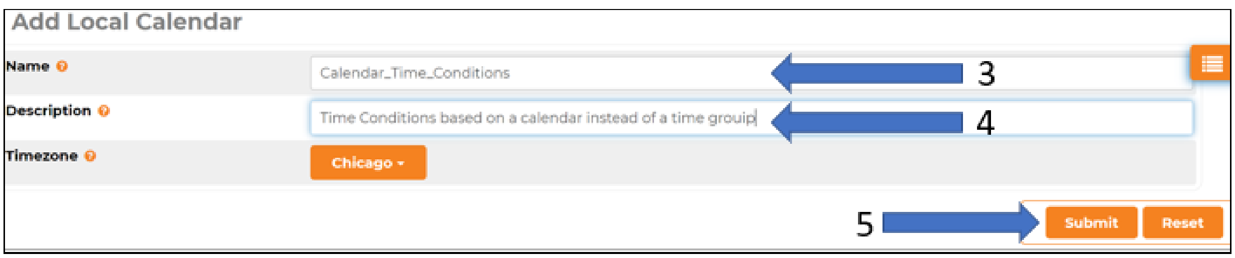
Calendar

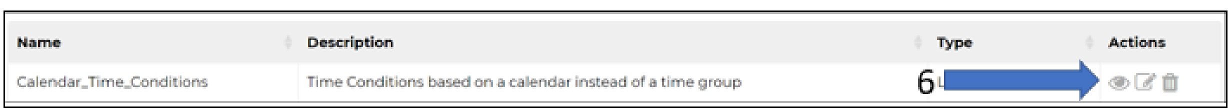
A Calendar can be used to schedule times to be used in a time Condition by creating events.

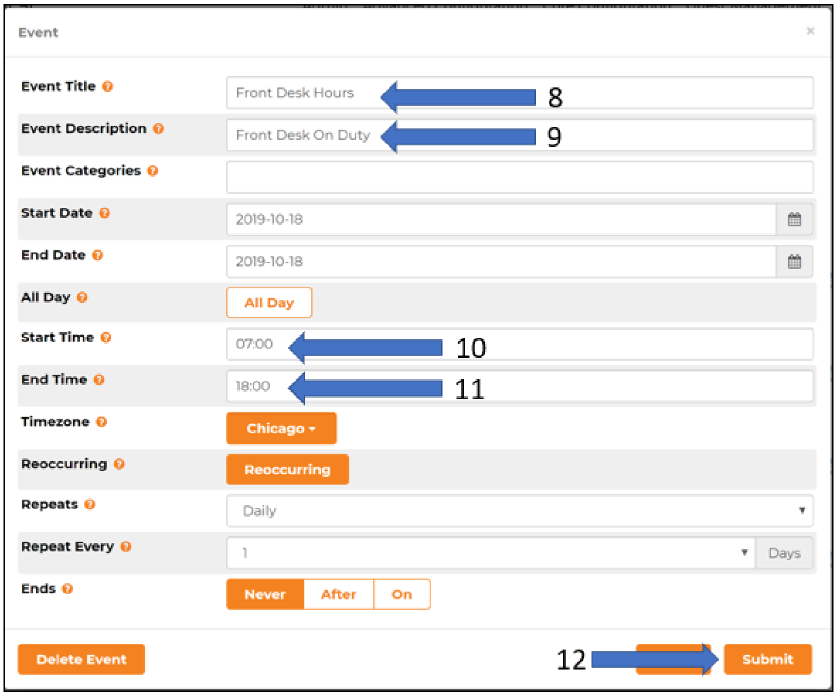
Navigate to Advanced Configuration > Calendars

1. Click on Add Calendar
2. Choose a Calendar type (local Calendar)
3. Fill in the Name Field
4. Fill in the Description Field
5. Submit and Apply
6. On the New Calendar click on the View Actions icon
7. In the Calendar View Click on a Add Event or double click in the Calendar
8. Fill in the Event Title Field
9. Fill in the Description Field
10. Specify the Time to Start
11. Specify the Time to Finish
12. Submit and Apply







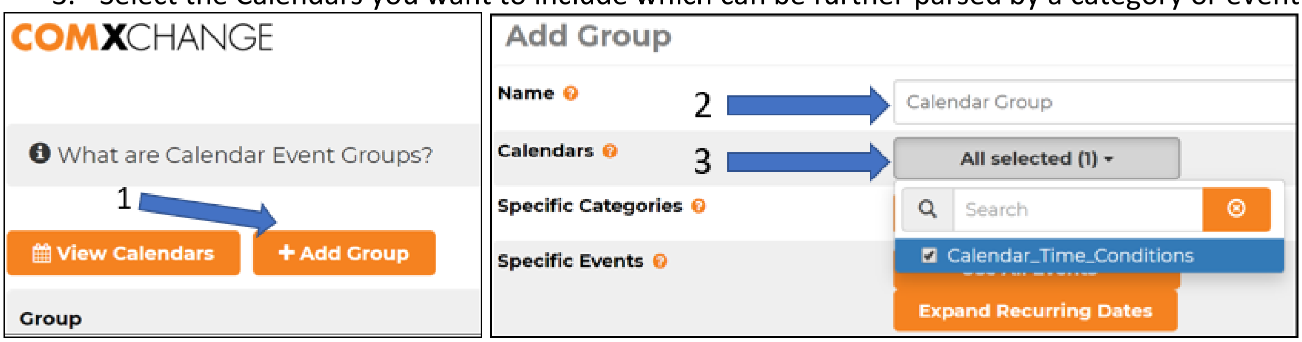


Calendar Event Groups

You can group times from multiple Calendars by creating a Calendar Group and adding multiple Calendars to it.

Navigate to Advanced Configuration > Calendars

1. Click on the Add Group Button
2. Fill in the Name field
3. Select the Calendars you want to include which can be further parsed by a category or event



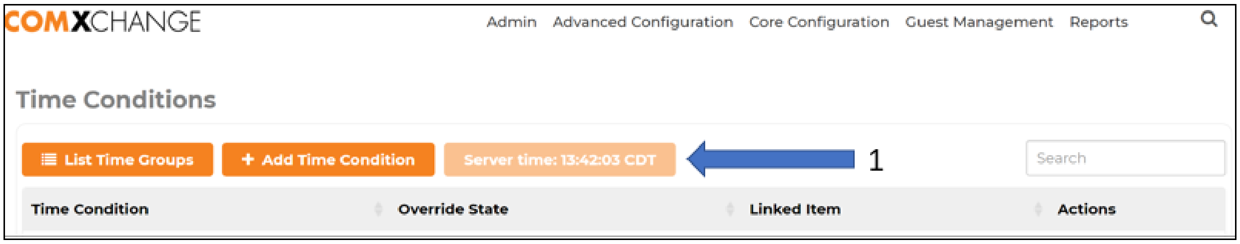
Time Conditions

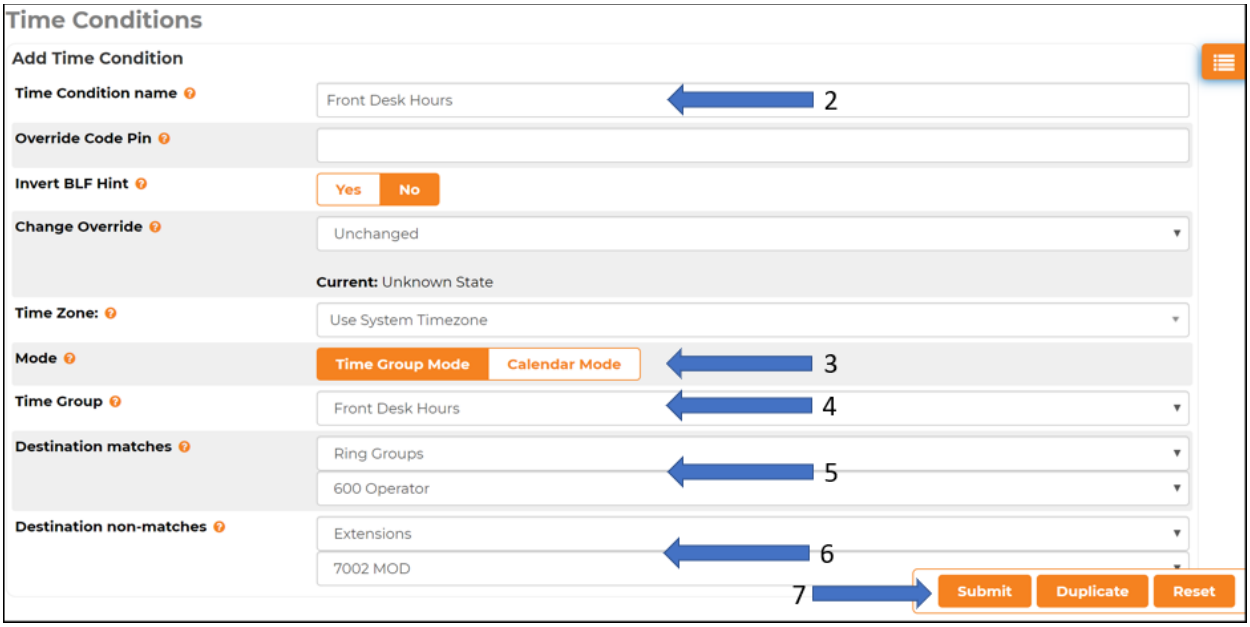
Time Condition with Time Group

A Time Condition will define the routing for calls during or outside the times defined in a Time Group or a Calendar. Below are the basic steps for setting up a Time Condition based on a Time Group.

Navigate to Advanced Configuration > Time Conditions

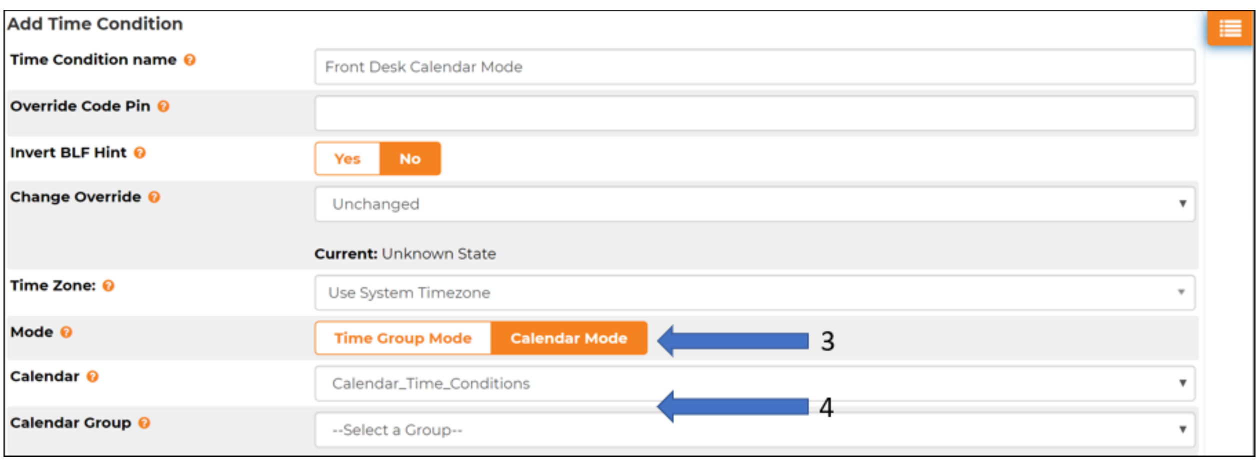
4. Click on the Add Time Condition Button
5. Fill in the Time Condition Name field
6. Choose Time Group for Mode
7. Specify the Time Group in the Drop-down
8. Choose the destination when the time matches
9. Choose a destination when the time doesn't match
10. Submit and Apply



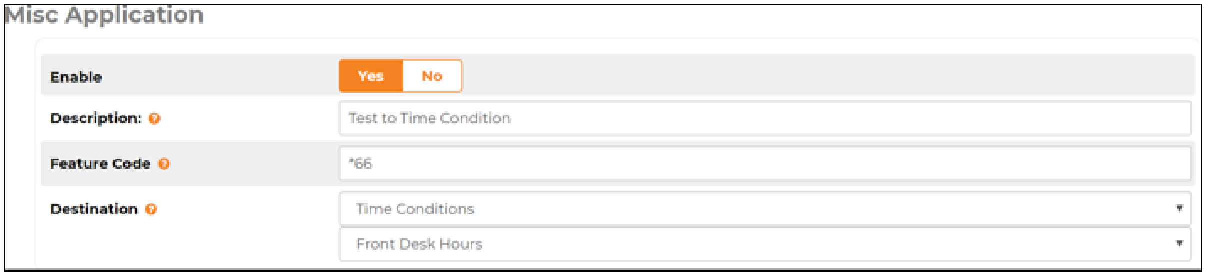


Time Group with a Calendar

To Create a Time Condition with a Calendar you would choose the Calendar Mode in step 3 and the Calendar or a Calendar Group for Step 4.



Creating a Time Condition also creates a feature code that allows you to override the current Time Condition and the normal flow will resume in the next transition. You can test the Time Condition by pointing a Misc Application to it. Once the Time Condition is tested you can set it as a destination for incoming calls. A BLF button can be created to toggle the Time Condition on or off to override the normal schedule.

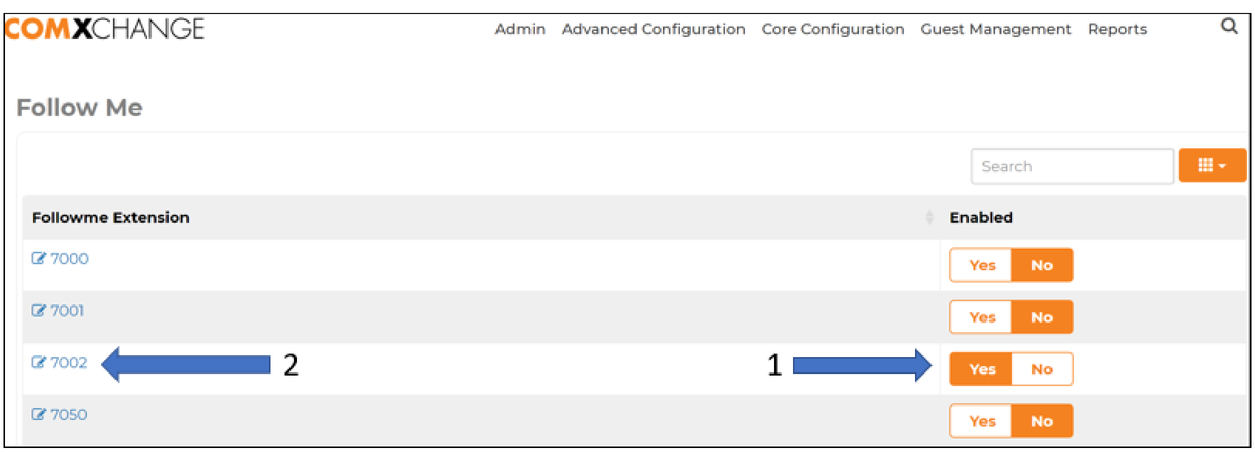


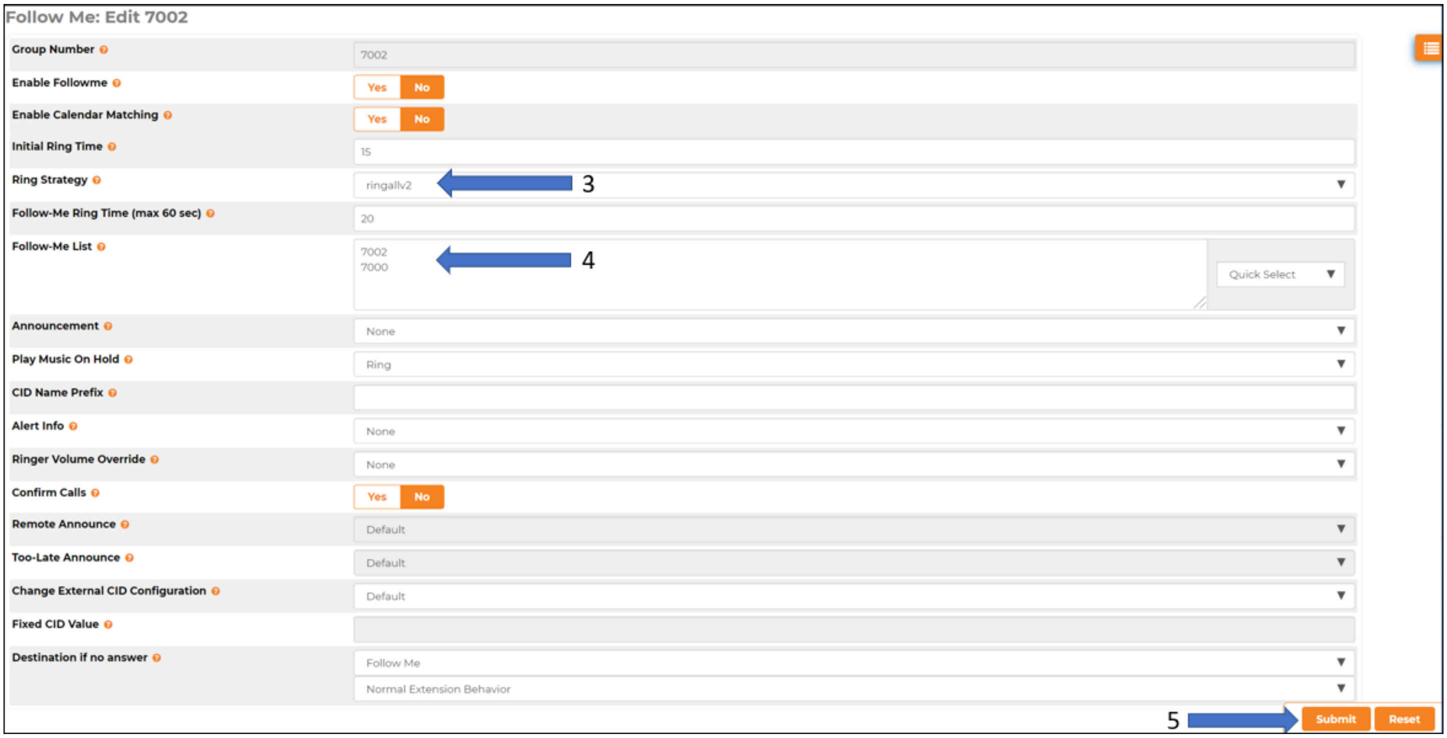
Follow Me

Follow Me Allows you to redirect a call placed to an extension to another destination. Calls received from ring groups or queues will not be redirected. You can program the Follow Me to ring an extension alone for a certain amount of time and then ring a different destination like a mobile phone (a # suffix must be used to dial an external number) or a different extension. There are also other ring strategies that can be used that are defined in the Ring Strategy Hint. This feature can also be turned on and off with a BLF. Create a Follow Me for the MOD extension to ring 7100

Navigate to Core Configuration > Follow Me

1. Enable the Follow Me on the extension on the right
2. Click on the Extension you enabled on the on the Left
3. Choose a ring strategy
4. Add numbers to call in the Follow Me List
5. Submit and Apply





The BLF Feature Code for Follow Me toggle is *74 followed by the extension. For the MOD phone you can set up a BLF key to point to *747002

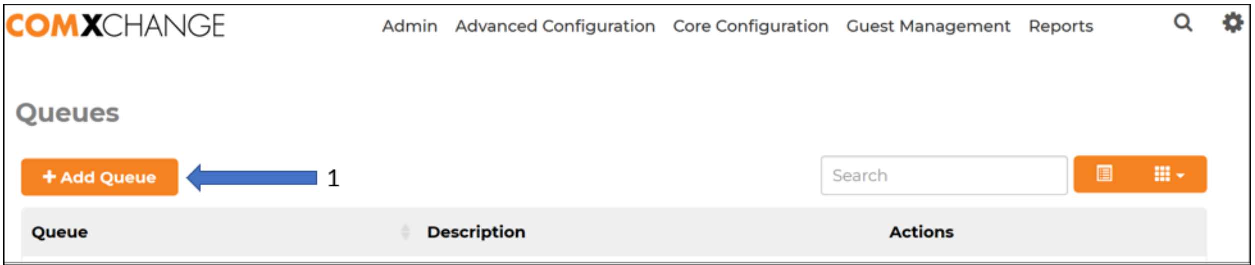
Queues

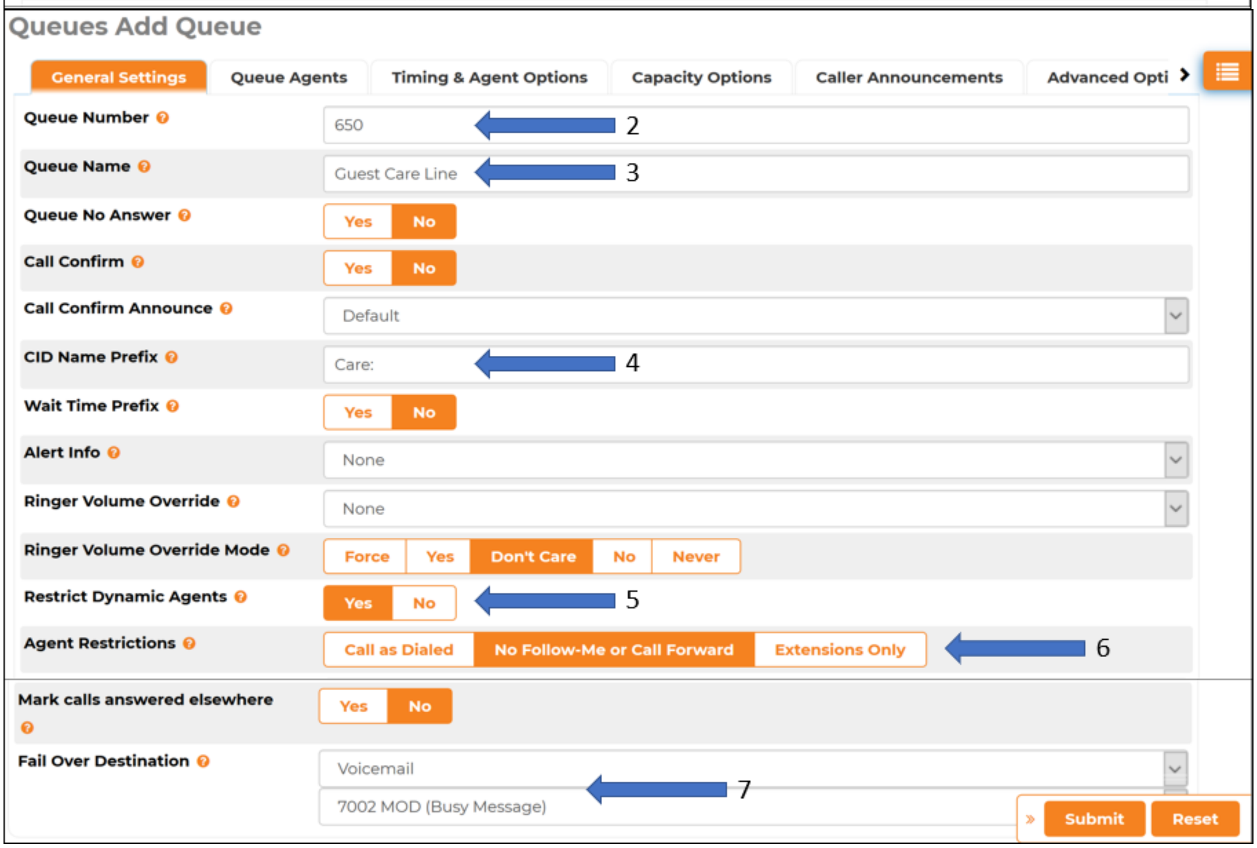
The Queues module is a more advanced version of a Ring Group. Building a Queue creates an extension and a destination that when dialed will ring multiple extensions at the same time. However, unlike a ring group, individual extensions added as Dynamic Members can be permitted to log in and out of the queue. Members of a queue can also pause their membership. A queue can be programmed to route calls to a single extension or to multiple extensions and can use various ring strategies. While waiting in the queue the system can play Music on Hold and give the caller an option to exit the queue and be routed to another destination. Calls can also time out and be routed to another destination. Below we will set up a queue for the hotels guest care line with two static and one dynamic agent.

Navigate to Core Configuration > Queues

1. Click on Add Queue
2. In the General Settings tab enter a Queue Number in the Queue Number Field (650)
3. Enter a Queue name in the Queue Name Field (Guest Care Line)
4. Enter a CID Prefix (Care: so agents know it coming from the Guest Care Line)

5. Restrict Dynamic Agents
6. Set Agent Restrictions
7. Set a Failover Destination

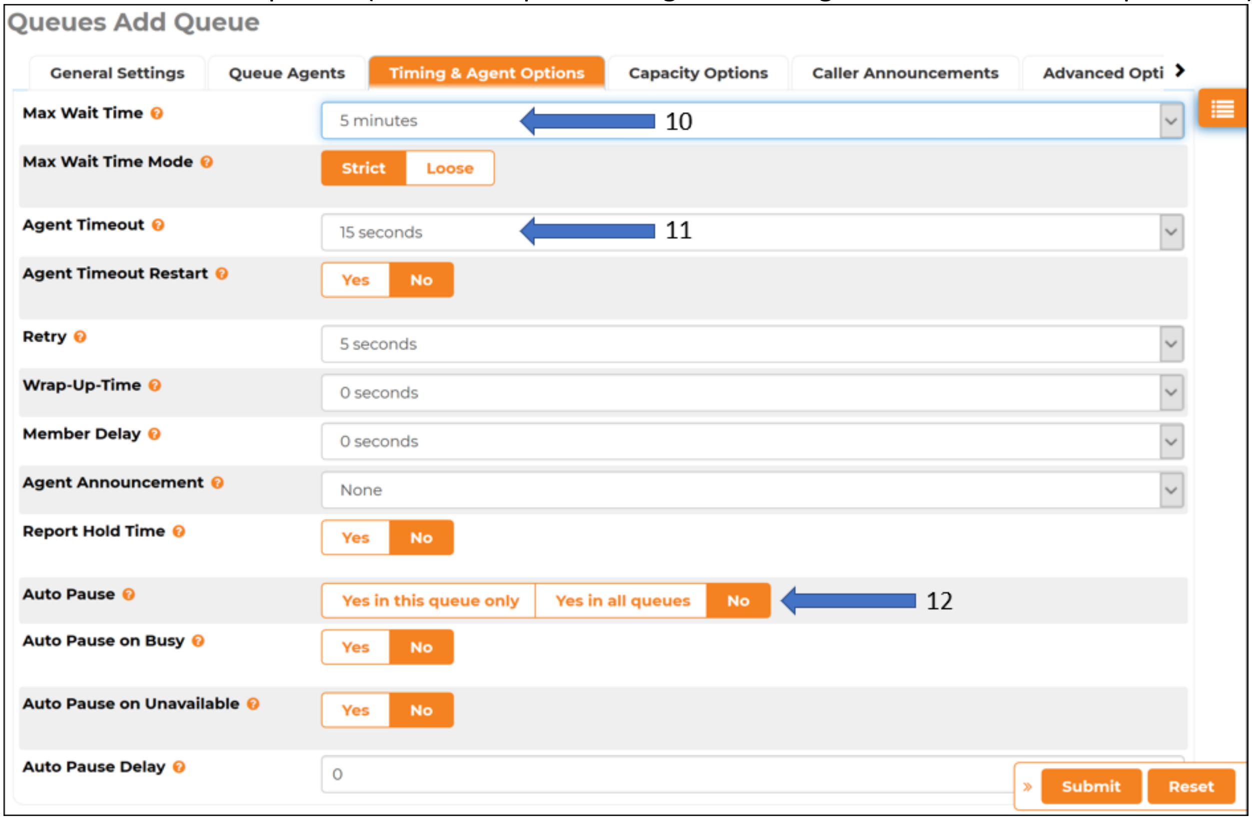




8. In the Queue Agents tab Select the desired Static Agents (7000 and 7001)
9. Select Dynamic Agents (7002)

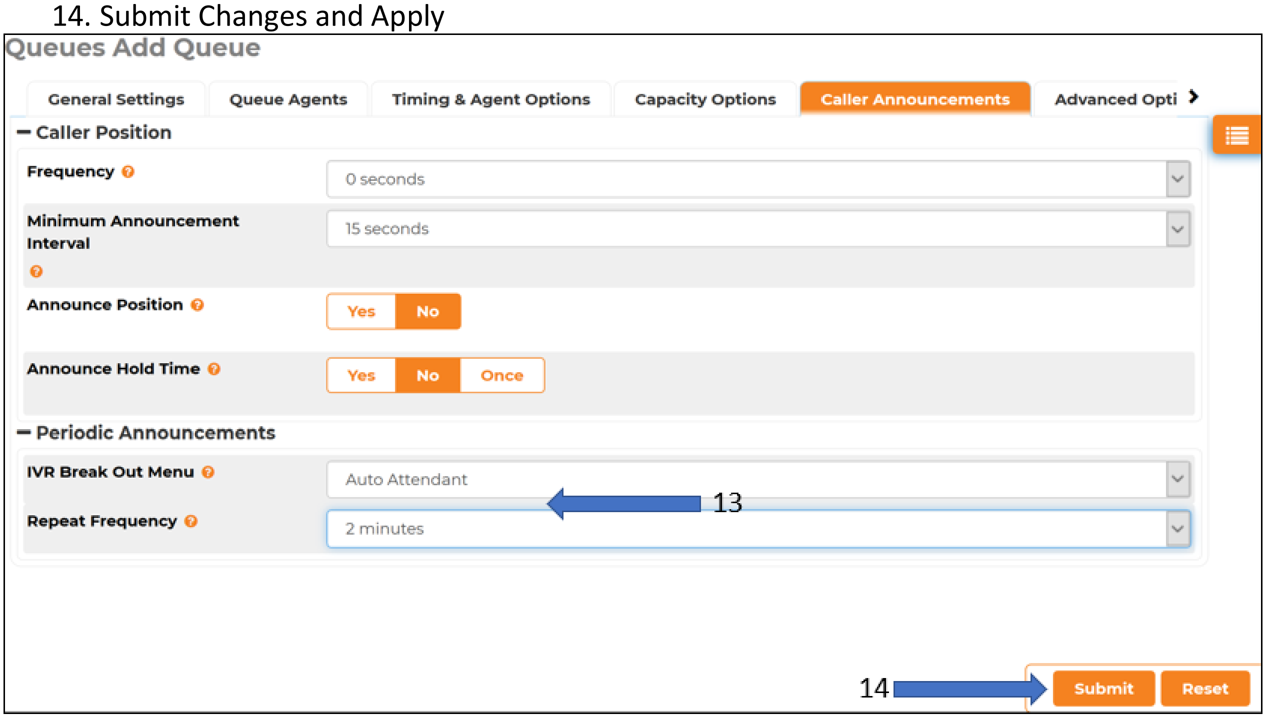
10. In the Timing and Agent Options tab set a Max Wait Time of 5 minutes. (before being sent to a failover destination)
11. Set Agent Timeout. (15 seconds is about 3 rings)

12. Auto Pause Options. (If set it will pause an agent if the agent doesn't answer a queue call)



13. You can set an IVR Breakout Menu and how often it repeats the IVR options.

14. Submit Changes and Apply



Dynamic Queue Members Login

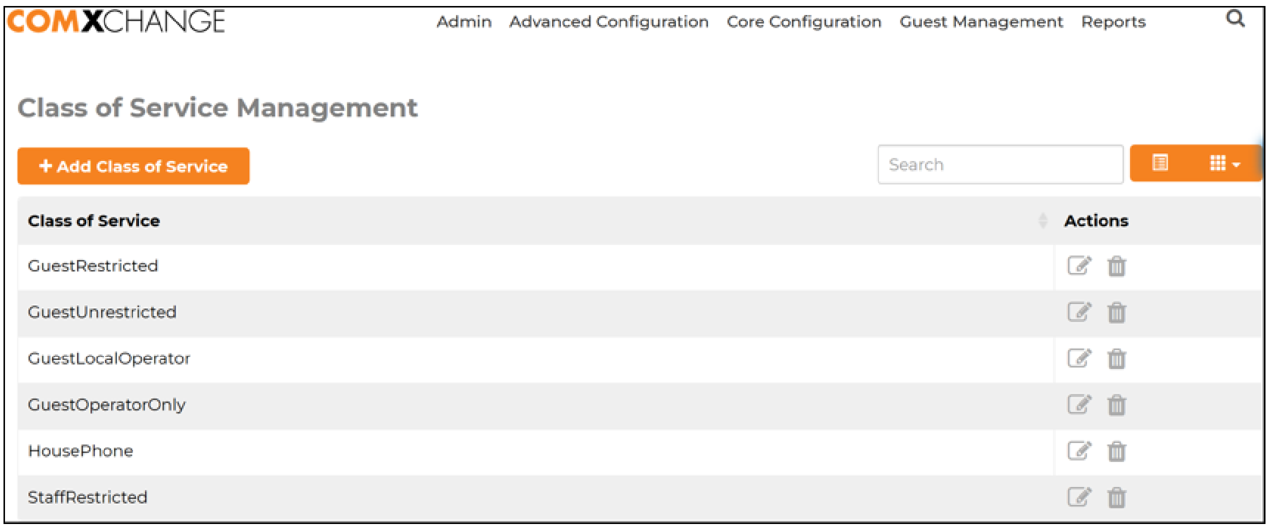
A Dynamic Queue Member can log in by dialing *45*<ext_no>*<queue_no>. For example, Extension 7002 would log in and out of queue 650 by dialing *45*7002*650. This can be simplified by programming a BLF button that dials the digits. Dialing *45 only will log in or out of all queues of which the extension on the phone is a dynamic member.

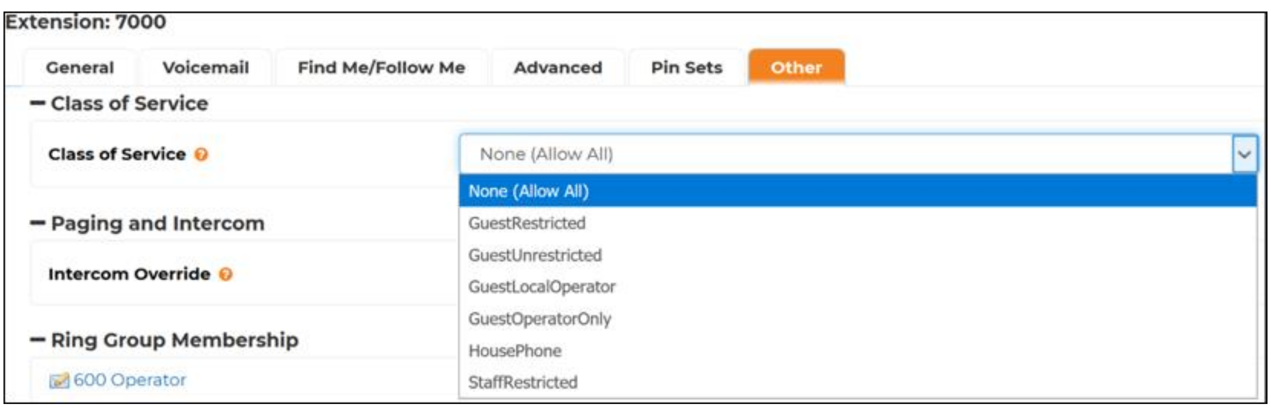
Queue Pause Activate / Deactivate

Members can activate or deactivate a pause to a queue membership by dialing *46*<ext_no>*<queue_no>. For example, Extension 7002 would pause queue 650 membership by dialing *46*7002*650. This can be simplified by programming a BLF button that dials the digits. Dialing *46 only will activate or deactivate pause for all queues of which the extension on the phone is a dynamic member.

Class of Service - Restricted Call Permissions / Staff or House phones

ComXchange has six types of Class of Service types created by default. These can be used on extensions that need restricted dialing enabled such as a house phone or a phone in a common area like a break room. A Class of Service will allow very granular access to the dial plan. A Class of Service can be applied to an extension in the Other tab - Class of Service section of the Extension.





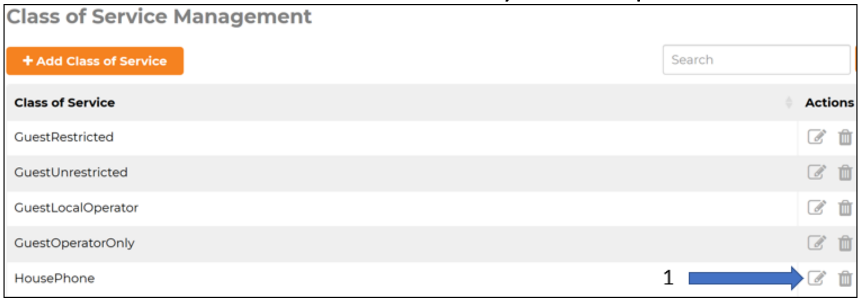
House Phone - Class of Service

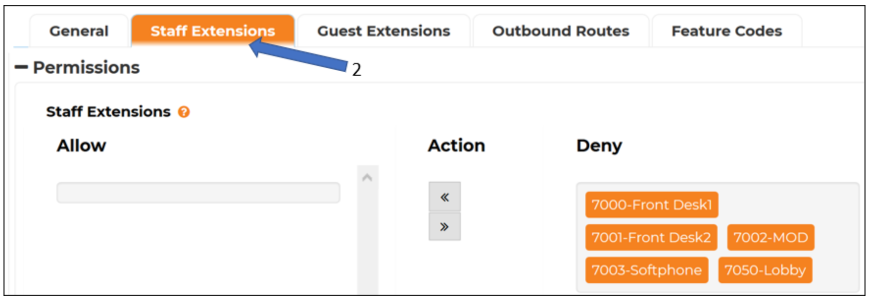
Navigate to Core Configuration > Class of Service

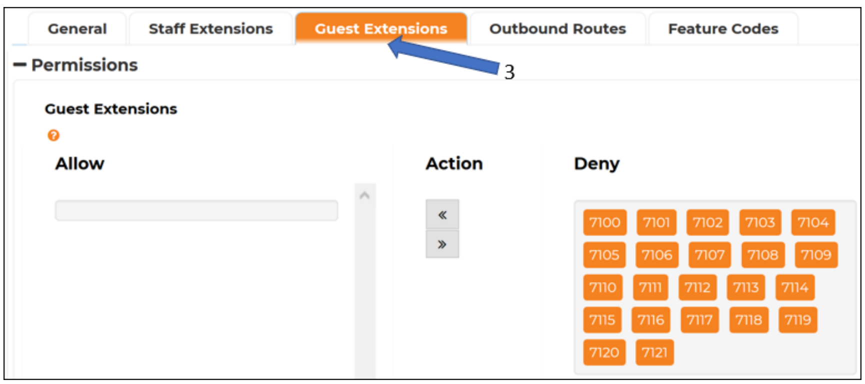
1. Click on HousePhone edit Actions icon on the right side
2. View the Staff Extensions tab
    a. This Class of Service will not allow a direct dial call to any Staff Extensions
3. View Guest Extensions tab

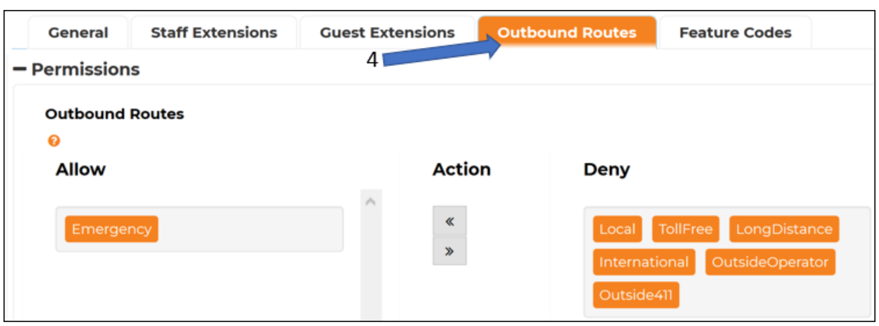
a. This Class of Service will not allow a direct dial call to any Guest Extensions

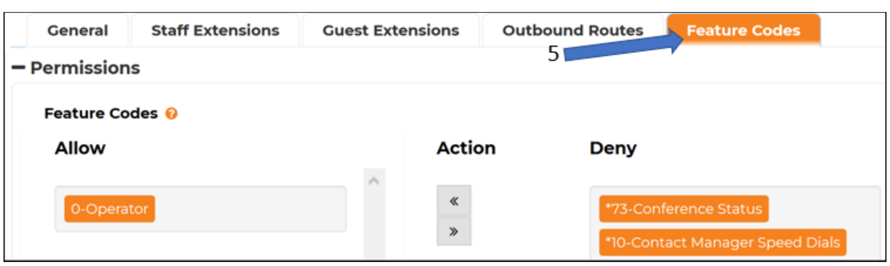
4. Outbound Routes tab
    a. This Class of Service will only allow outbound calls through the Emergency Trunk
5. Feature Codes tab
    a. This Class of Service only allows a phone to dial the Operator Feature Code 0











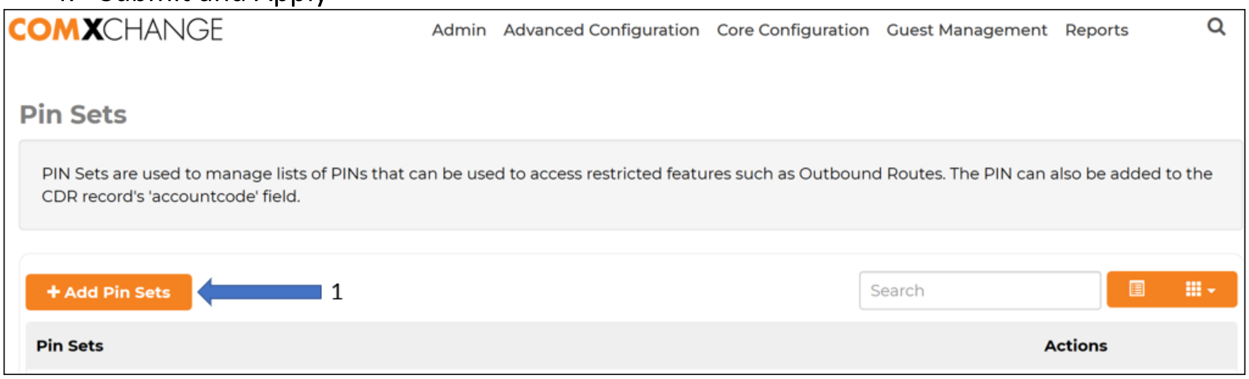
Long Distance PIN

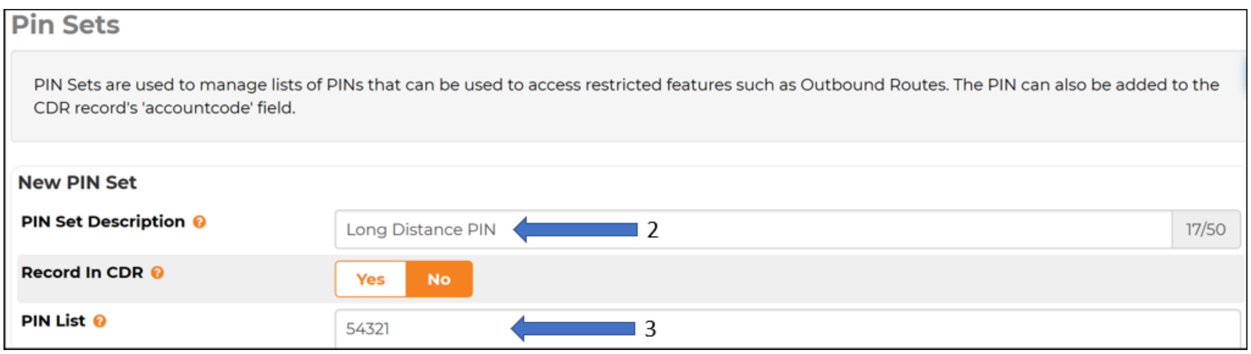
You can also require an extension to dial a PIN when dialing long distance by creating a 2nd outbound route that requires a PIN and forcing the extension to use that route by modifying and using the restricted Class of Service. You can also force a single extension or a group of extensions to use the route by using the CallerID field in the Outbound Routes Dial Patterns.

Create a PIN Set

Navigate to Advanced Configuration > PIN Sets

1. Click the Add Pin Sets button
2. Fill in the Pin Set Description field
3. Add one or more PIN's that can be used to authenticate a caller
4. Submit and Apply

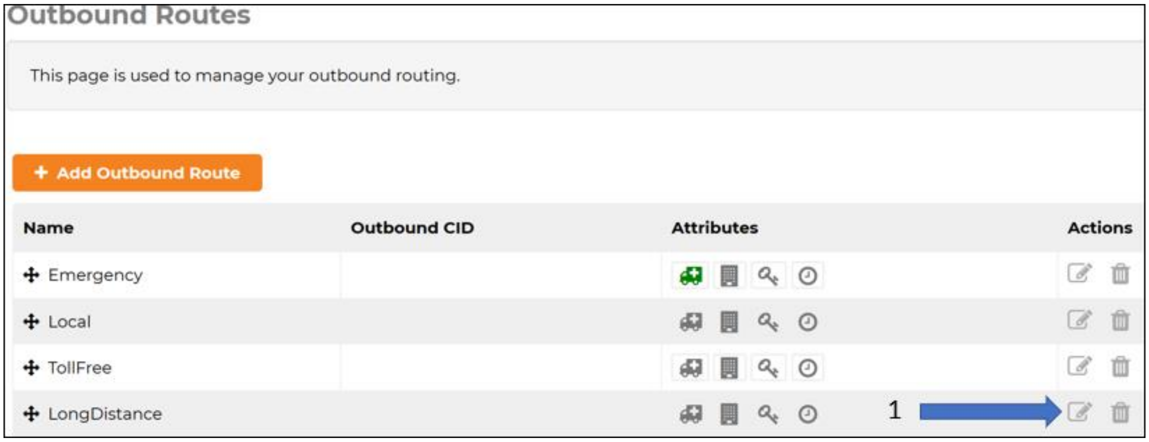


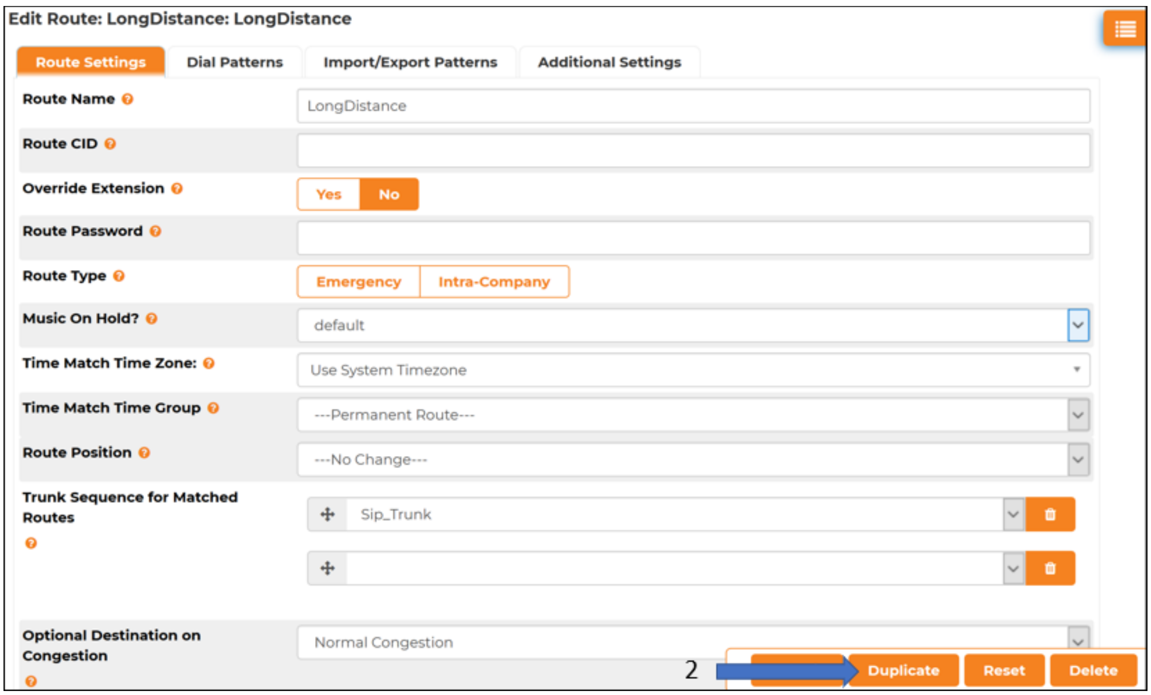


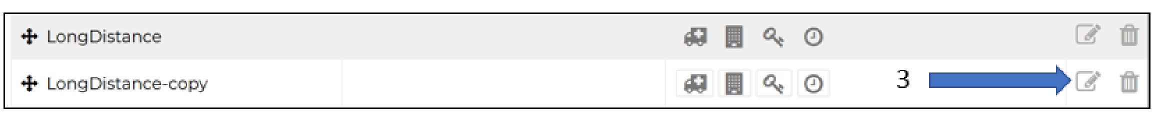
Create a 2nd Long Distance Route

Navigate to Core Configuration > Outbound Routes

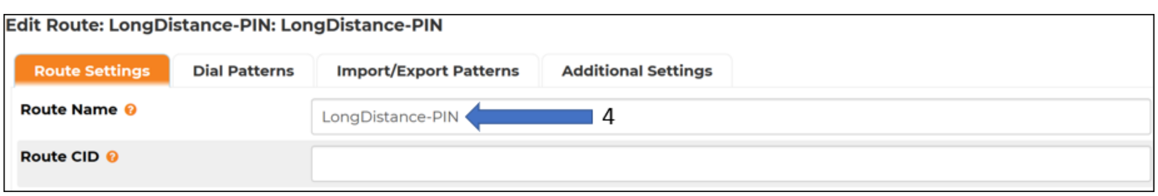
1. Click on the edit Actions icon for the LongDistance Outbound Route
2. Duplicate Route.
3. Click on the edit Actions icon of the newly Created LongDistance Route







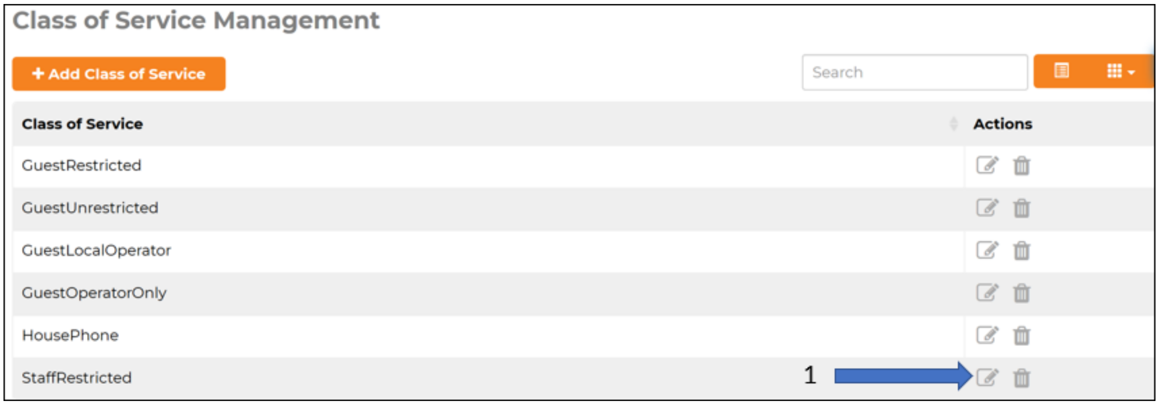
4. Change the Route Name
5. Under -Additional Settings choose the PIN Set that was previously created
6. Submit Changes and Apply

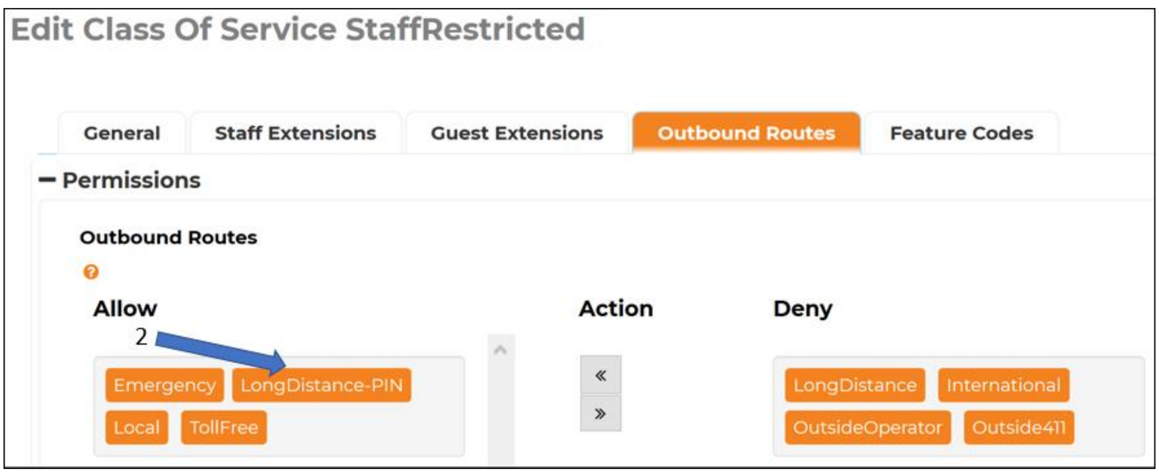


Modify Class of Service -StaffRestricted

Navigate to Advanced Configuration > Class of Service - Restricted

1. Click on the edit Actions icon of the StaffRestricted Context
2. On the Outbound Routes tab, add the LongDistance-Pin Outbound Route to the Allow group
3. Submit and Apply

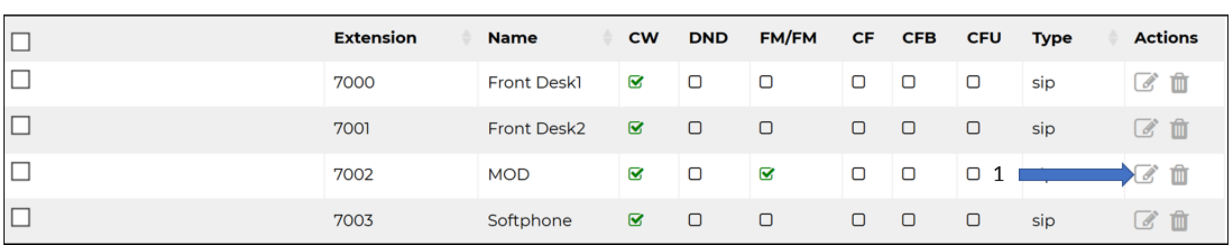


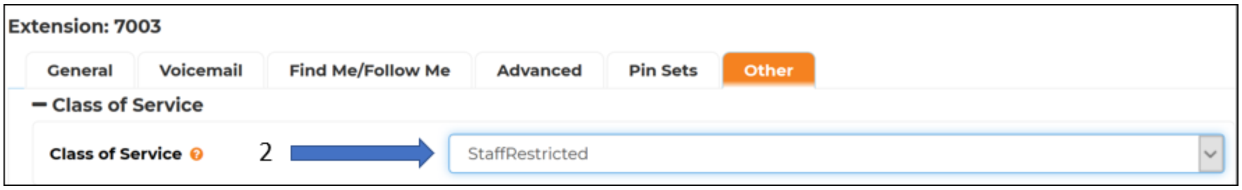


Apply StaffRestricted Class of Service to the Extension

Navigate to Core Configuration > Extensions

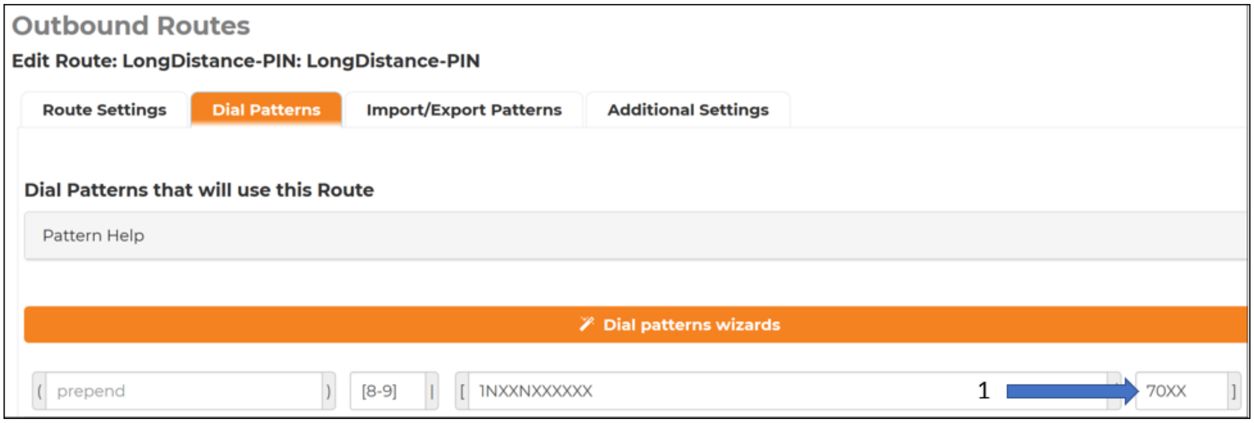
1. Click on the edit Actions icon of the extension that you want to assign the Class of Service
2. In the Other tab choose the StaffRestricted Class of Service from the dropdown





Force a PIN with CallerID

In the outbound route LongDistance-Pin you can add an extension or a wild card that will force any calls that match the pattern to use this route. Below the Pattern in the Dial Pattern tab will match any 4-digit extensions that begin with 70 and only those extensions will use this route.



Bulk Handler

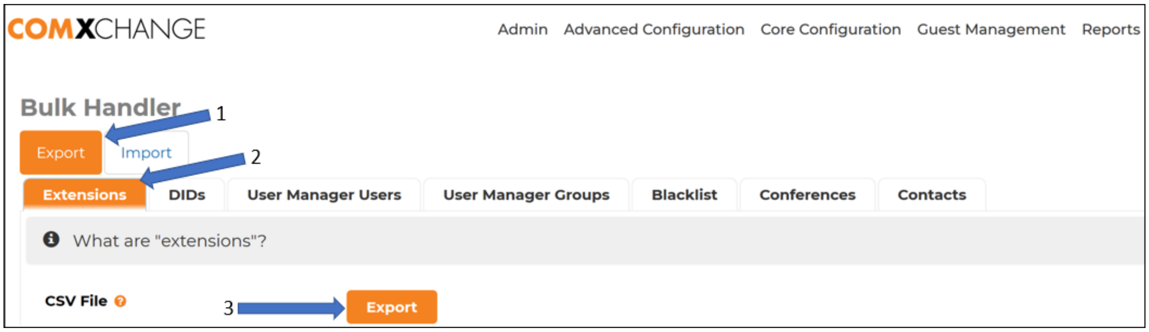
You can use the Bulk Handler Module to import Bulk Extension and Bulk DID's, among other objects The Bulk Handler Module allows for CSV Export and Import of Extension Information, Inbound Routes, Blacklist numbers, Conferences, Contacts, and Users.

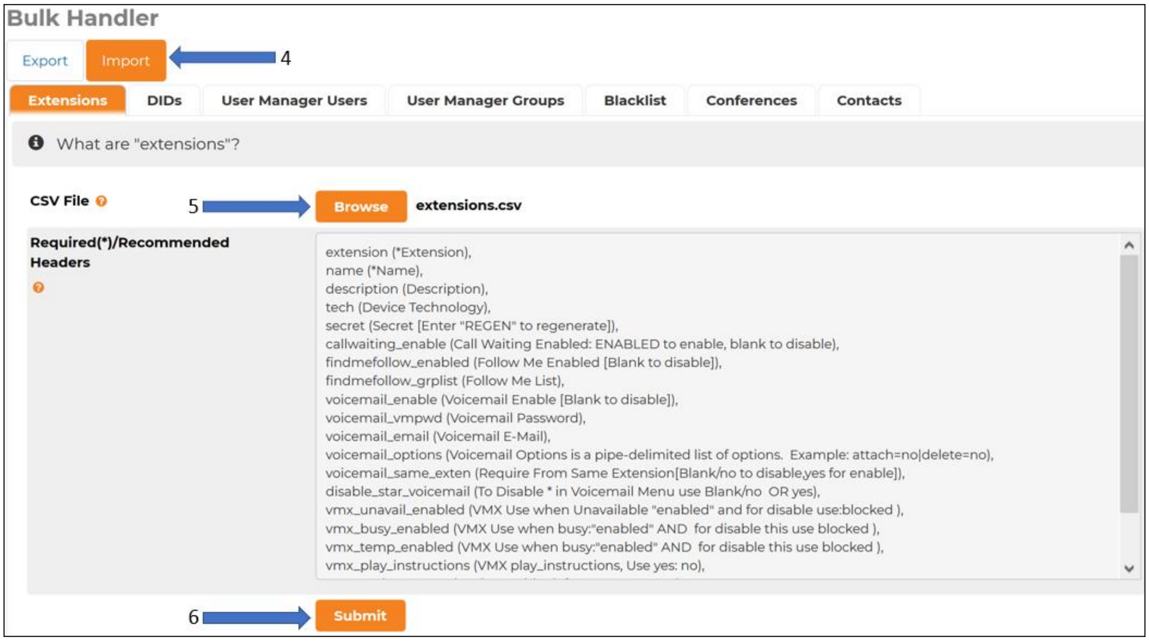
Bulk Handler - Extensions

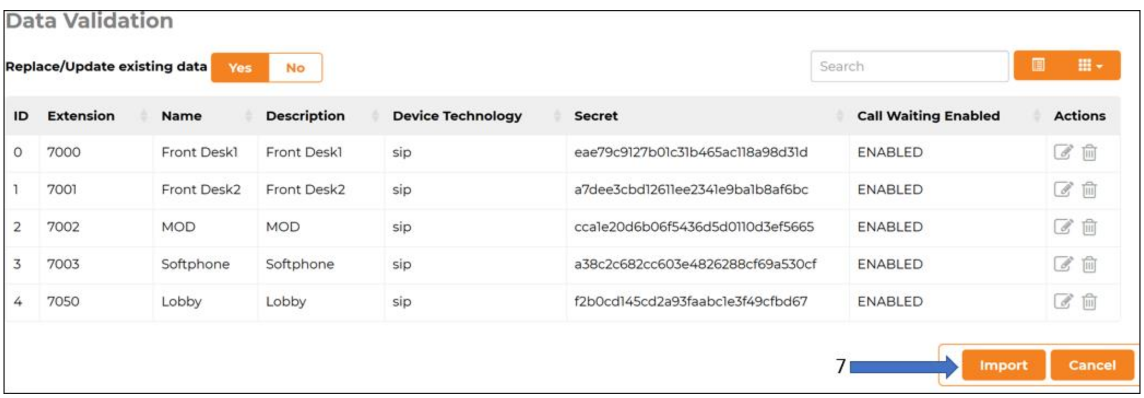
You can export a current extensions CSV file from the ComXchange Server, modify it, then reload the modified CSV file to add many extensions very quickly. If an email address is included in the extension that email will be notified of extension changes, you can place "noemail" in the Override Address to stop the notifications from being sent.

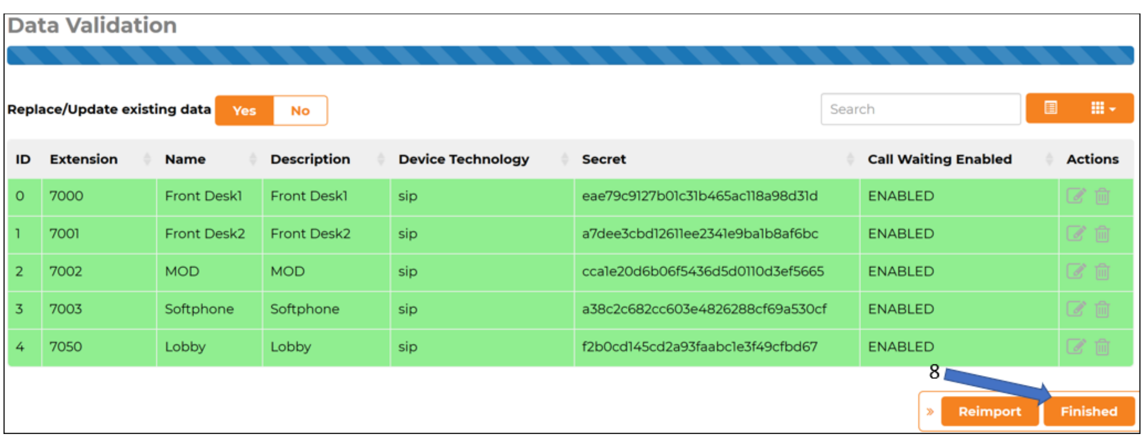
Navigate to Admin > Bulk Handler

1. Choose the Export Button
2. Click on the Extensions Tab
3. Click on Export
    a. Modify the downloaded file
4. Click on the Import Button
5. Browse to the modified file
6. Click the Submit button
7. Click the Import button
8. Click the Finished button


COMXCHANGE
Admin
Advanced Configuration
Core Configuration
Guest Management
Reports
Bulk Handler
1
Export
Import
2
Extensions
DIDs
User Manager Users
User Manager Groups
Blacklist
Conferences
Contacts
What are "extensions"?
CSV File
3
Export


Bulk Handler
Export
Import
4
Extensions
DIDs
User Manager Users
User Manager Groups
Blacklist
Conferences
Contacts
What are "extensions"?
CSV File
5
Browse
extensions.csv
Required(*)/Recommended Headers
extension (*Extension),
name (*Name),
description (Description),
tech (Device Technology),
secret (Secret [Enter "REGEN" to regenerate]),
callwaiting_enable (Call Waiting Enabled: ENABLED to enable, blank to disable),
findmefollow_enabled (Follow Me Enabled [Blank to disable]),
findmefollow_grplist (Follow Me List),
voicemail_enable (Voicemail Enable [Blank to disable]),
voicemail_vmpwd (Voicemail Password),
voicemail_email (Voicemail E-Mail),
voicemail_options (Voicemail Options is a pipe-delimited list of options. Example: attach=no|delete=no),
voicemail_same_exten (Require From Same Extension[Blank/no to disable,yes for enable]),
disable_star_voicemail (To Disable * in Voicemail Menu use Blank/no OR yes),
vmx_unavail_enabled (VMX Use when Unavailable "enabled" and for disable use:blocked),
vmx_busy_enabled (VMX Use when busy:"enabled" AND for disable this use blocked),
vmx_temp_enabled (VMX Use when busy:"enabled" AND for disable this use blocked),
vmx_play_instructions (VMX play_instructions, Use yes: no),
6
Submit


Data Validation
Replace/Update existing data
Yes
No
Search
ID | Extension | Name | Description | Device Technology | Secret | Call Waiting Enabled | Actions
0 | 7000 | Front Desk1 | Front Desk1 | sip | eae79c9127b01c31b465ac118a98d31d | ENABLED
1 | 7001 | Front Desk2 | Front Desk2 | sip | a7dee3cbd12611ee2341e9ba1b8af6bc | ENABLED
2 | 7002 | MOD | MOD | sip | cca1e20d6b06f5436d5d0110d3ef5665 | ENABLED
3 | 7003 | Softphone | Softphone | sip | a38c2c682cc603e4826288cf69a530cf | ENABLED
4 | 7050 | Lobby | Lobby | sip | f2b0cd145cd2a93faabc1e3f49cfbd67 | ENABLED
7
Import
Cancel

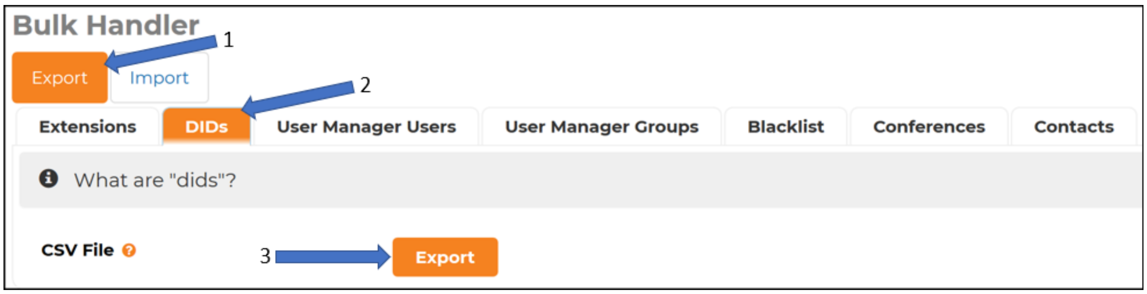

Data Validation
Replace/Update existing data
Yes
No
Search
ID | Extension | Name | Description | Device Technology | Secret | Call Waiting Enabled | Actions
0 | 7000 | Front Desk1 | Front Desk1 | sip | eae79c9127b01c31b465ac118a98d31d | ENABLED
1 | 7001 | Front Desk2 | Front Desk2 | sip | a7dee3cbd12611ee2341e9ba1b8af6bc | ENABLED
2 | 7002 | MOD | MOD | sip | cca1e20d6b06f5436d5d0110d3ef5665 | ENABLED
3 | 7003 | Softphone | Softphone | sip | a38c2c682cc603e4826288cf69a530cf | ENABLED
4 | 7050 | Lobby | Lobby | sip | f2b0cd145cd2a93faabc1e3f49cfbd67 | ENABLED
8
Reimport
Finished

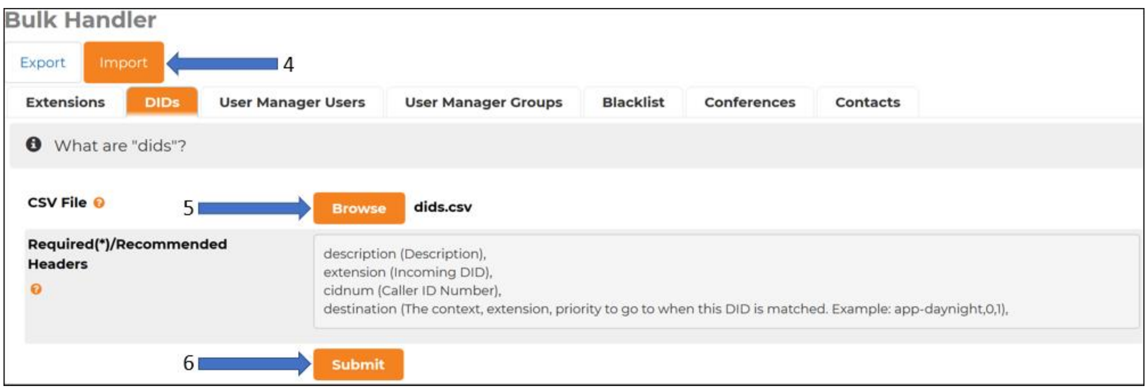
Bulk DID's

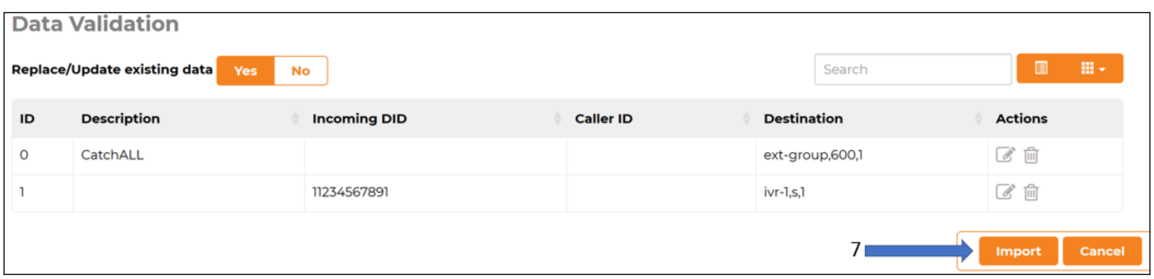
You can download a current List of inbound DID routes from the ComXchange server, modify it, then reload the modified CSV file to add inbound DID routes very quickly.

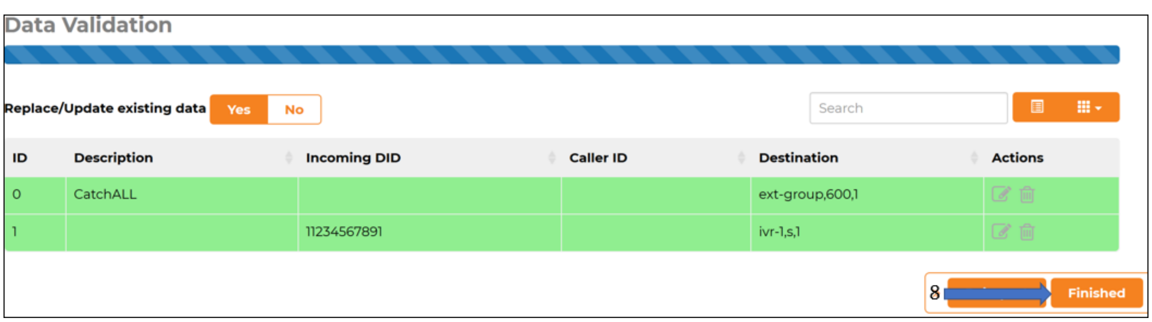
Navigate to Admin > Bulk Handler

1. Choose the Export Button
2. Click on the Extensions Tab
3. Click on Export
    a. Modify the downloaded file
4. Click on the Import Button
5. Browse to the modified file
6. Click the Submit button
7. Click the Import button
8. Click the Finished button







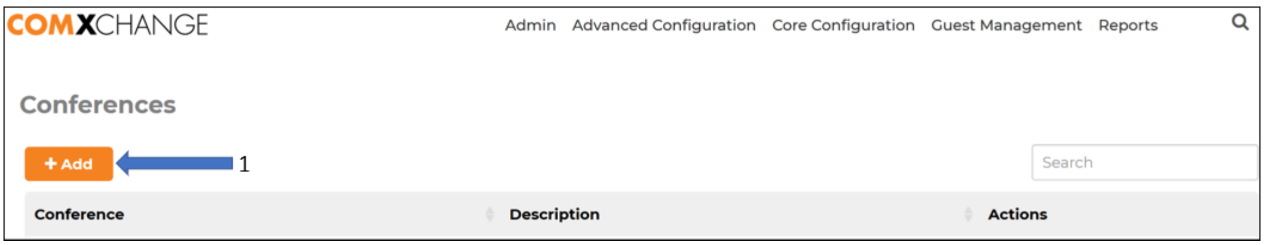


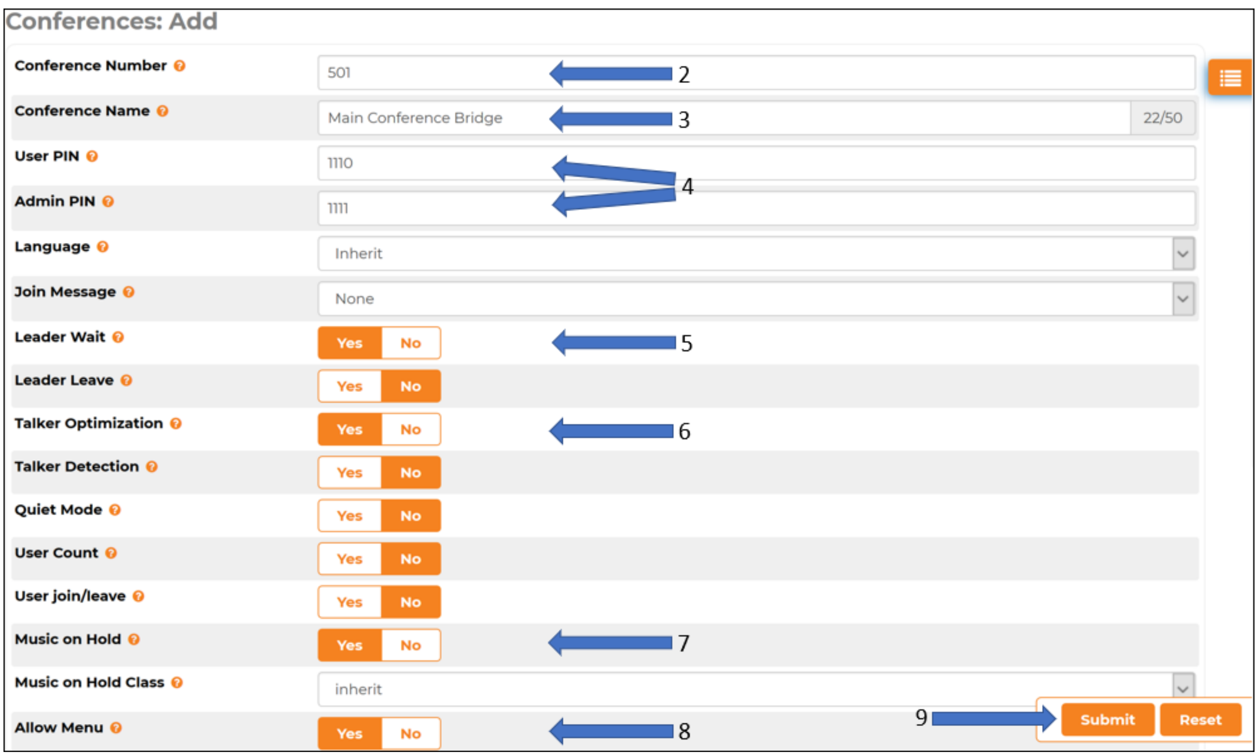
Conference Rooms

A conference room is a single extension number that users can dial so they can talk to each other in a conference call. It also creates a destination that calls can be sent to.

Navigate to Core Configuration > Conferences.

1. Click the Add Button
2. Fill in the Conference Number field.
3. Fill in the Conference Name field.
4. Assign User and Admin PINs.
5. Change the Leader Wait to Yes.
6. Change Talker Optimization to Yes.
7. Change Music on Hold to Yes.
8. Change Allow Menu to Yes.
9. Submit Changes and Apply.





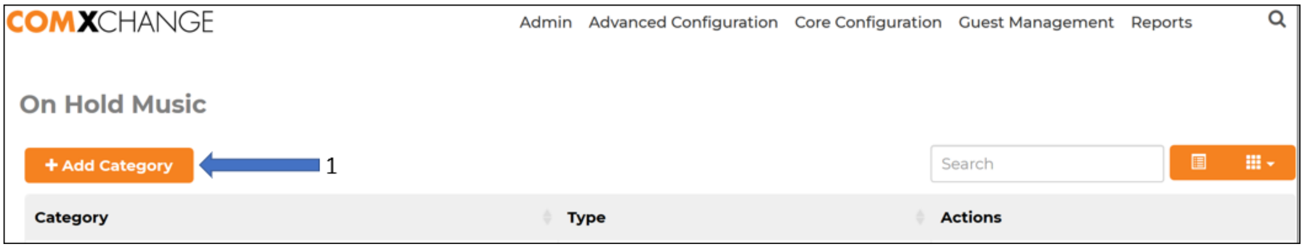
Music on Hold

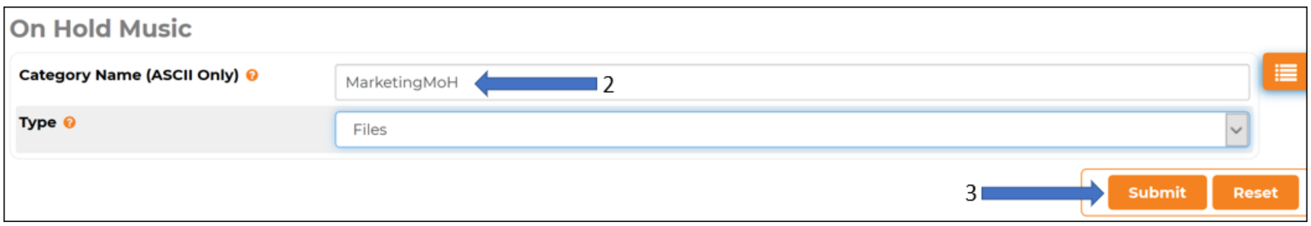
Music on Hold (MoH) allows the Hotel to play music to fill the silence when callers are placed on hold. MoH can be used in Queues, Inbound Routes, Conference Rooms, ETC. Customized MoH can be uploaded as .wav and .mp3 files, or you can create a category to use a streaming service.

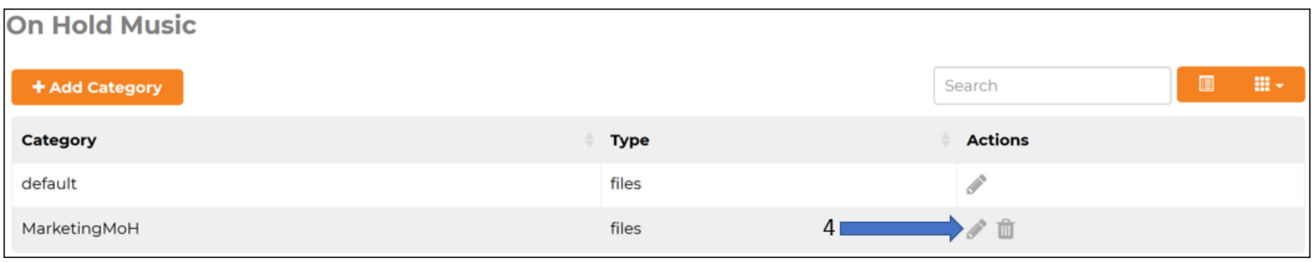
Upload Wav or Mp3

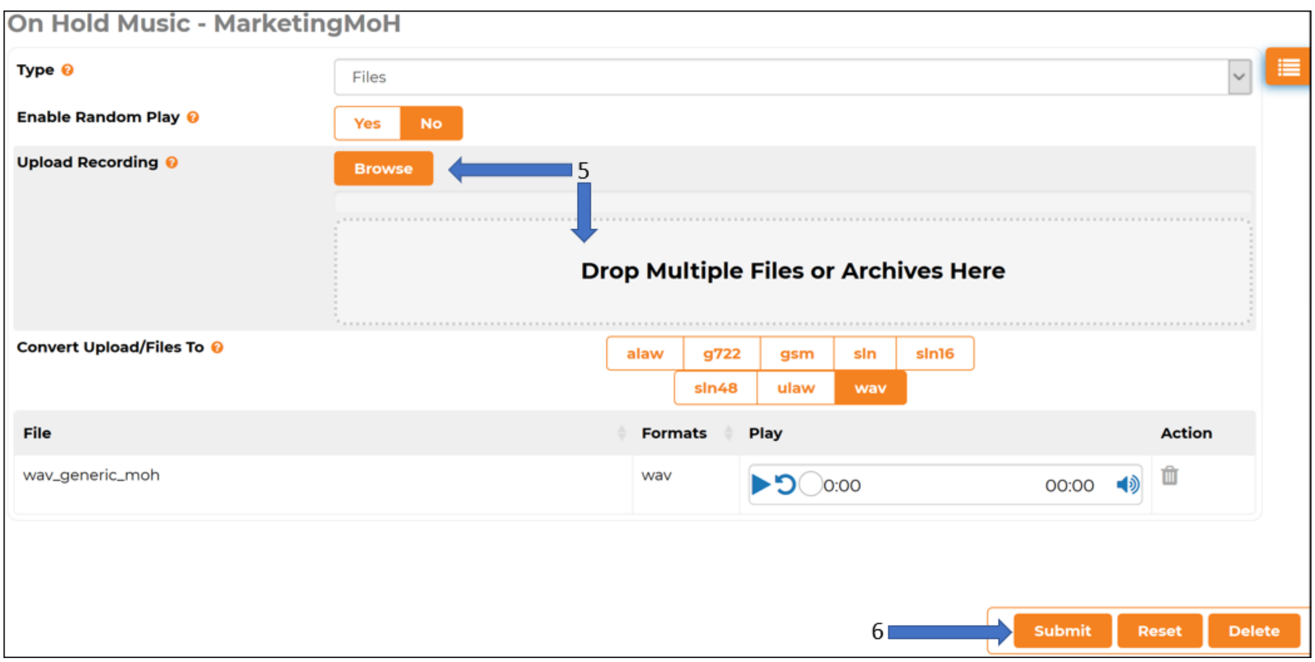
Navigate to Advanced Configuration > Music on Hold

1. Click on the Add Music Button
2. Fill in the Category Name field
3. Click on Submit
4. Click on the edit Actions icon for the new Category
5. Click the Browse button and navigate to the file you want to upload or drag the file to the page
6. Click on Submit

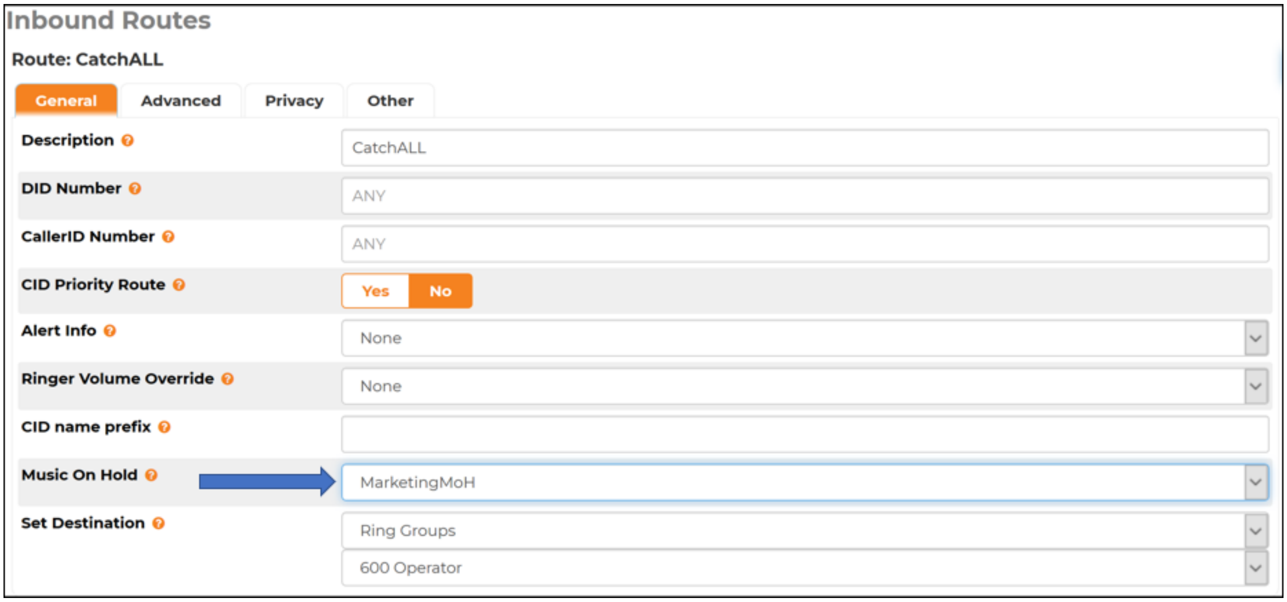








The new Category of music is ready and can be used for an Incoming Route, etc.



Warning: If you replace the wav_generic_moh.wav file in the reserved default category the wav_generic_moh.wav music file will be deleted from the system.

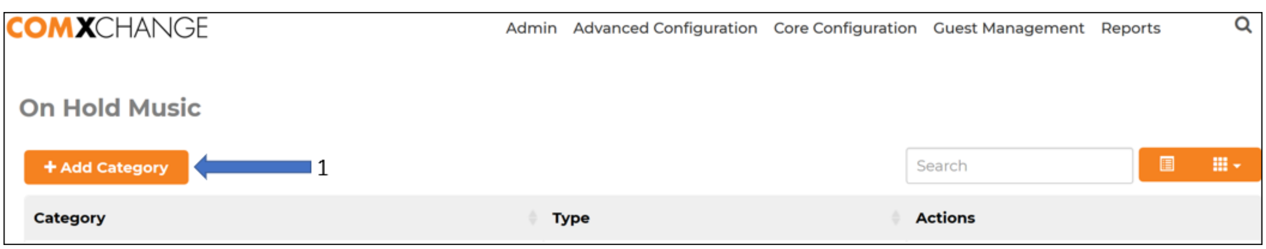
Streaming Music on Hold

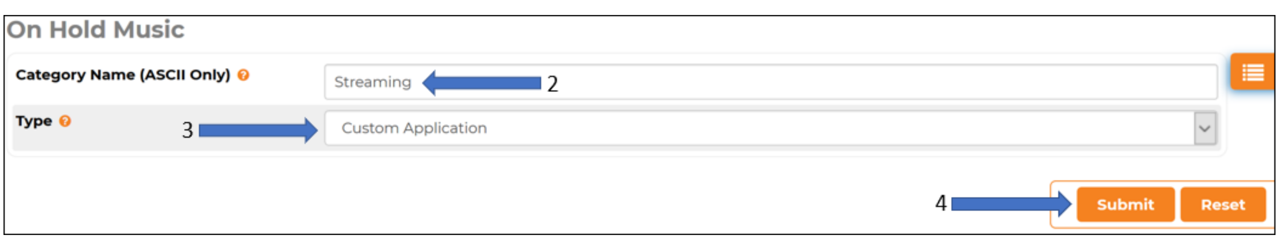
Music on hold can also be supplied by a streaming music device.

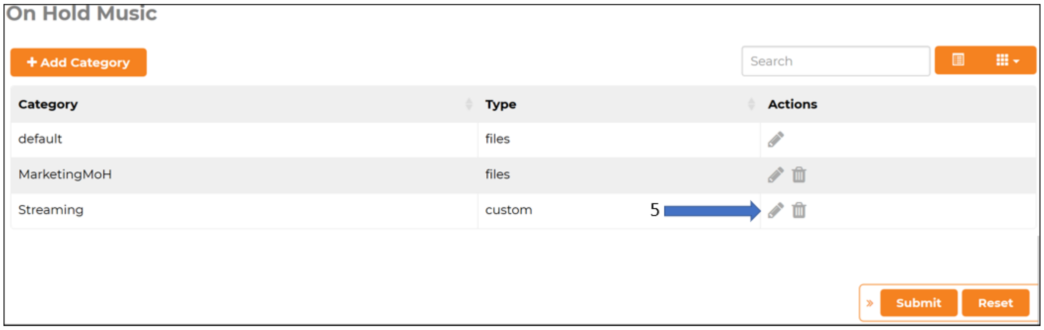
Navigate to Advanced Configuration > Music on Hold

1. Click on the Add Category Button
2. Fill in the Category Name field
3. Choose Custom Application in the Type dropdown
4. Click on Submit
5. Click on the edit Actions icon for the new Category
6. Fill in the Application Field in the format of:
    a. /usr/bin/mpg123 -q -r 8000 -f 8192 --mono -s http://<ip address>/xstream

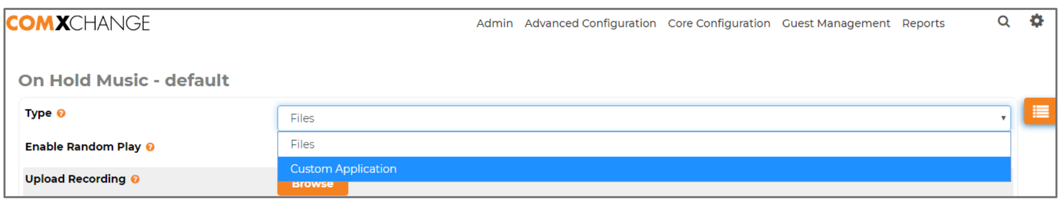
7. Click on Submit







To have a streaming service be used as a default MoH edit the default on hold music and choose Custom Application as the type then use the same string discussed above in the Application field

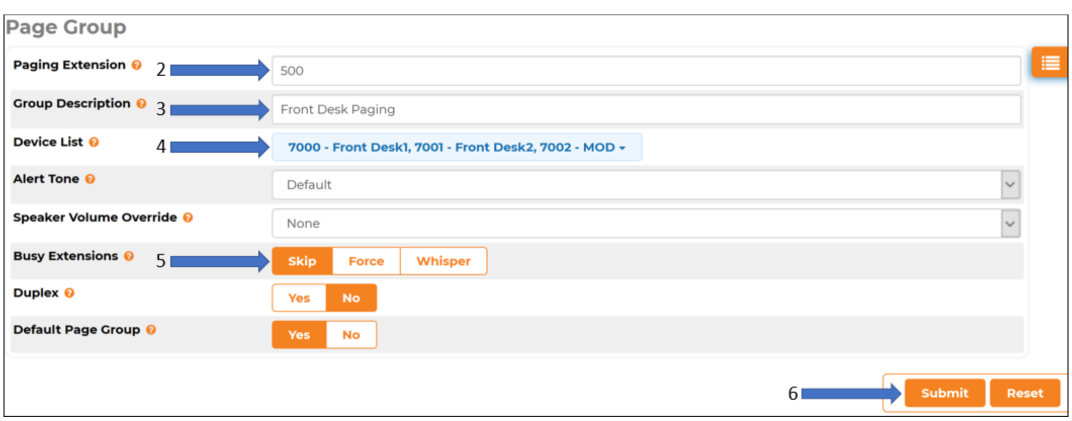


Paging

Allows the setup of a group of phones that will auto answer a call made to the paging extension. This requires SIP auto answer which doesn't work on most soft phones.

Navigate to Advanced Configuration > Paging and Intercom.

1. Click on Add Page Group Button
2. Fill in the Paging Extension Field
3. Fill in the Group Description Field
4. Click in the Device List and check the extensions to be part of the group
5. Choose what to do if the Extension is busy
6. Submit and Apply



Logs / Reports

ComXchange has built in logging and reports that are valuable to monitor the system or for help with troubleshooting issues with making or receiving calls. The reports can give information on registered extensions and trunks as well as provide access to the asterisk and call logs.

Asterisk Info

The Asterisk Summary Page will open with a brief chart of the Asterisk Uptime, Active SIP Channels, SIP Registries, and SIP Peer Status. There are many other reports such as SIP Peers that will list the SIP Name/Username and whether they are registered or not with an IP address.

Navigate to Reports > Asterisk Info

Summary

Asterisk Info

This page supplies various information about Asterisk
Current Asterisk Version: 13.28.1

Summary

Asterisk System uptime: 1 day, 7 hours, 9 minutes, 33 seconds
Last reload: 4 hours, 10 minutes, 49 seconds

Active SIP Channel(s): 4
Sip Registry: 1
Sip Peers:
- Online: 6
- Online-Unmonitored: 0
- Offline: **21**
- Offline-Unmonitored: **0**

Active IAX2 Channel(s): 0
IAX2 Registry: 1
IAX2 Peers:
- Online: 0
- Offline: **0**
- Unmonitored: 0

SIP Peers

The peers report will now have both the staff and the guest extensions

Peers

Chan_Sip Peers

Name/username	Host	Dyn	Forcerport	Comedia	ACL	Port	Status	Description
7000/7000	192.168.101.192	D	No	No	A	5062	OK (6 ms)	
7001/7001	192.168.101.191	D	No	No	A	5062	OK (76 ms)	
7002/7002	192.168.101.60	D	No	No	A	5062	OK (3 ms)	
7003	(Unspecified)	D	No	No	A	0	UNKNOWN	
7050/7050	192.168.101.21	D	No	No	A	5084	OK (1 ms)	
7100/7100	192.168.101.21	D	No	No	A	5101	OK (1 ms)	
7101/7101	192.168.101.21	D	No	No	A	5099	OK (1 ms)	

SIP Registry

Registries

Chan_Sip Registry

```
Host                              dnsmgr Username    Refresh State          Reg.Time
sip2.graymatternetworks.com:5060       Y   1742         120 Request Sent
1 SIP registrations.
```

Asterisk Logfiles

The Asterisk Log files can provide details into a call's events. In the examples below you can see the logs for the Full Log and the Fail2ban Log.

Navigate to Reports > Asterisk Logfiles

Full Log

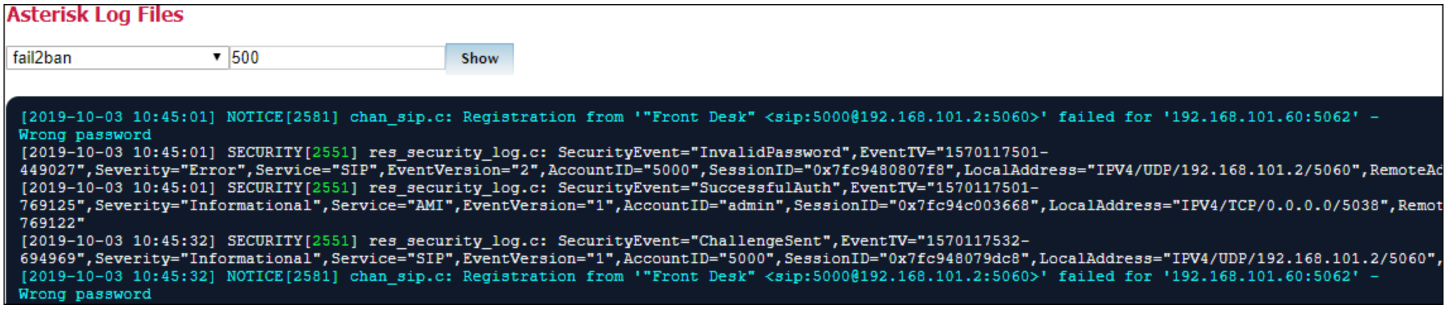
Asterisk Log Files

File	full
Lines	500
Filter	

Show

```
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [s@macro-user-callerid:20] Set("SIP/7001-00000000", "__TTL=64") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [s@macro-user-callerid:21] GotoIf("SIP/7001-00000000", "1?continue") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx_builtins.c: Goto (macro-user-callerid,s,37)
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [s@macro-user-callerid:37] Set("SIP/7001-00000000", "CALLERID(number)=7001") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [s@macro-user-callerid:38] Set("SIP/7001-00000000", "CALLERID(name)=Front Desk2") in new
stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [s@macro-user-callerid:39] GotoIf("SIP/7001-00000000", "0?cnum") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [s@macro-user-callerid:40] Set("SIP/7001-00000000", "CDR(cnam)=Front Desk2") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [s@macro-user-callerid:41] Set("SIP/7001-00000000", "CDR(cnum)=7001") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [s@macro-user-callerid:42] Set("SIP/7001-00000000", "CHANNEL(language)=en") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:4] ExecIf("SIP/7001-00000000", "0?Set(CHANNEL(language)=)") in new
stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:5] Set("SIP/7001-00000000", "MEETME_ROOMNUM=501") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:6] Set("SIP/7001-00000000", "MAX_PARTICIPANTS=0") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:7] Set("SIP/7001-00000000", "MEETME_MUSIC=") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:8] ExecIf("SIP/7001-00000000", "1?Set(MAX_PARTICIPANTS=0)") in new
stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:9] ExecIf("SIP/7001-00000000", "0?Set(MEETME_MUSIC=inherit)") in new
stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:10] GotoIf("SIP/7001-00000000", "0?ANSWERED") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:11] Answer("SIP/7001-00000000", "") in new stack
[2019-10-30 10:04:29] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:12] Wait("SIP/7001-00000000", "1") in new stack
[2019-10-30 10:04:30] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:13] GotoIf("SIP/7001-00000000", "0?USER:CHECKPIN") in new stack
[2019-10-30 10:04:30] VERBOSE[5748][C-00000000] pbx_builtins.c: Goto (from-internal,501,14)
[2019-10-30 10:04:30] VERBOSE[5748][C-00000000] pbx.c: Executing [501@from-internal:14] GotoIf("SIP/7001-00000000", "1?READPIN") in new stack
```

Fail2Ban Log

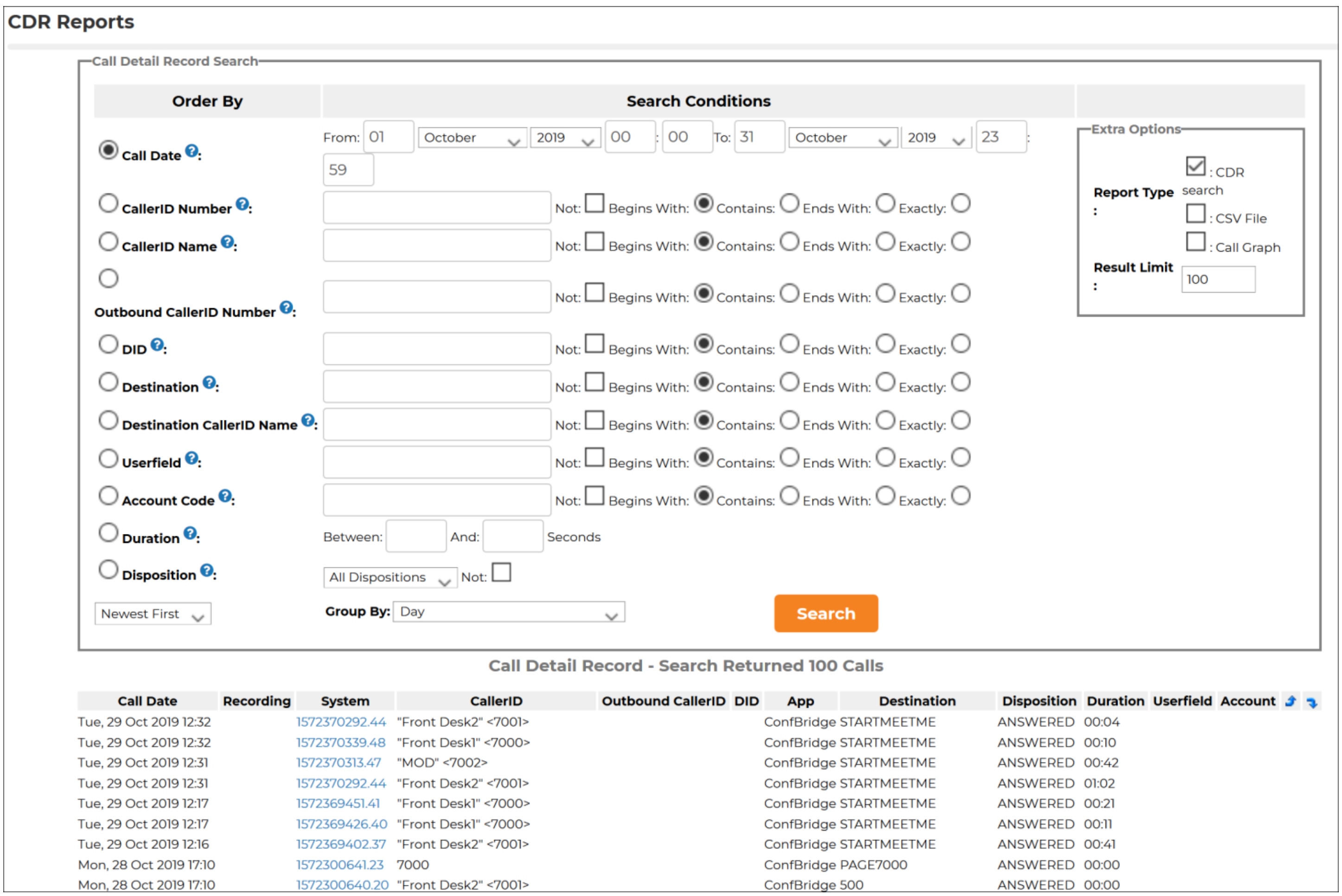


CDR Reports

The CDR Reports allow you to view or download a report of calls made and received by ComXchange. There are many options to search and filter by including Date, Date Range, Number Called, Caller ID, etc.

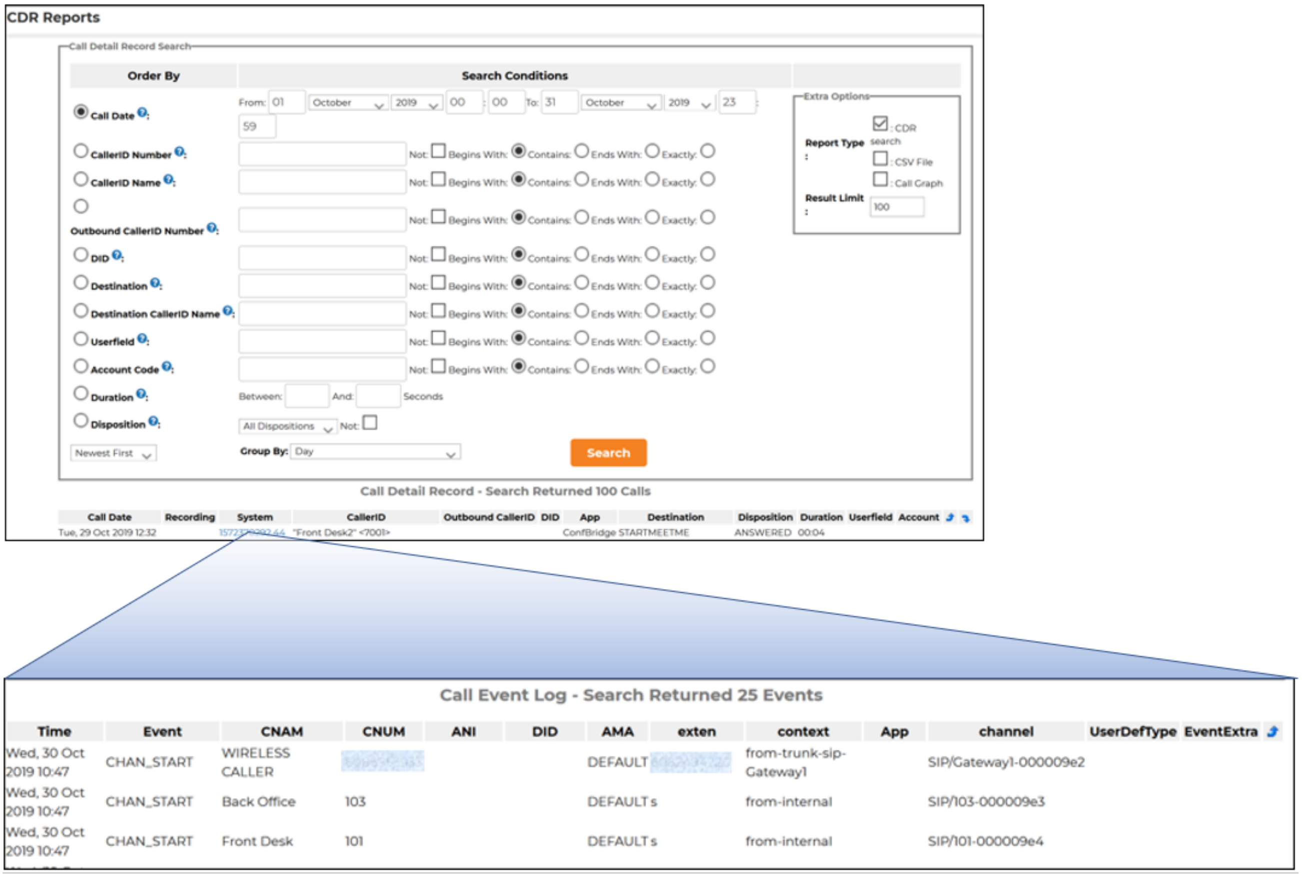
Navigate to Reports > CDR Reports

Place your search criteria and click on Search.



Call Date	Recording	System	CallerID	Outbound CallerID	DID	App	Destination	Disposition	Duration	Userfield	Account
Tue, 29 Oct 2019 12:32		1572370292.44	"Front Desk2" <7001>			ConfBridge	STARTMEETME	ANSWERED	00:04		
Tue, 29 Oct 2019 12:32		1572370339.48	"Front Desk1" <7000>			ConfBridge	STARTMEETME	ANSWERED	00:10		
Tue, 29 Oct 2019 12:31		1572370313.47	"MOD" <7002>			ConfBridge	STARTMEETME	ANSWERED	00:42		
Tue, 29 Oct 2019 12:31		1572370292.44	"Front Desk2" <7001>			ConfBridge	STARTMEETME	ANSWERED	01:02		
Tue, 29 Oct 2019 12:17		1572369451.41	"Front Desk1" <7000>			ConfBridge	STARTMEETME	ANSWERED	00:21		
Tue, 29 Oct 2019 12:17		1572369426.40	"Front Desk1" <7000>			ConfBridge	STARTMEETME	ANSWERED	00:11		
Tue, 29 Oct 2019 12:16		1572369402.37	"Front Desk2" <7001>			ConfBridge	STARTMEETME	ANSWERED	00:41		
Mon, 28 Oct 2019 17:10		1572300641.23	7000			ConfBridge	PAGE7000	ANSWERED	00:00		
Mon, 28 Oct 2019 17:10		1572300640.20	"Front Desk2" <7001>			ConfBridge	500	ANSWERED	00:00		

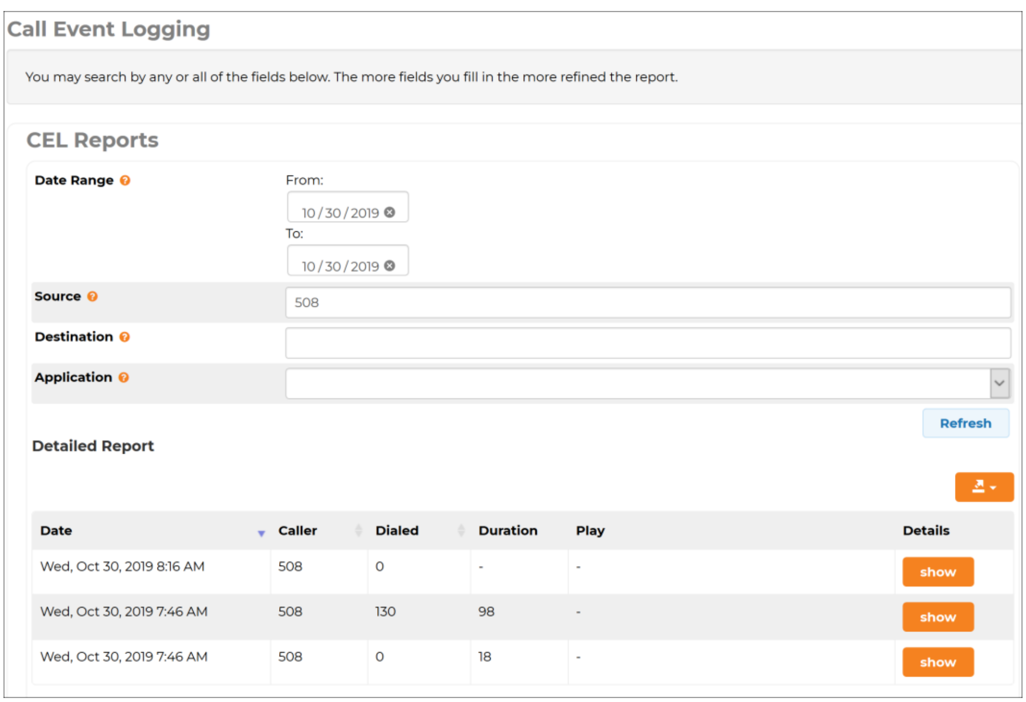
You can click on the links under system to see more details.



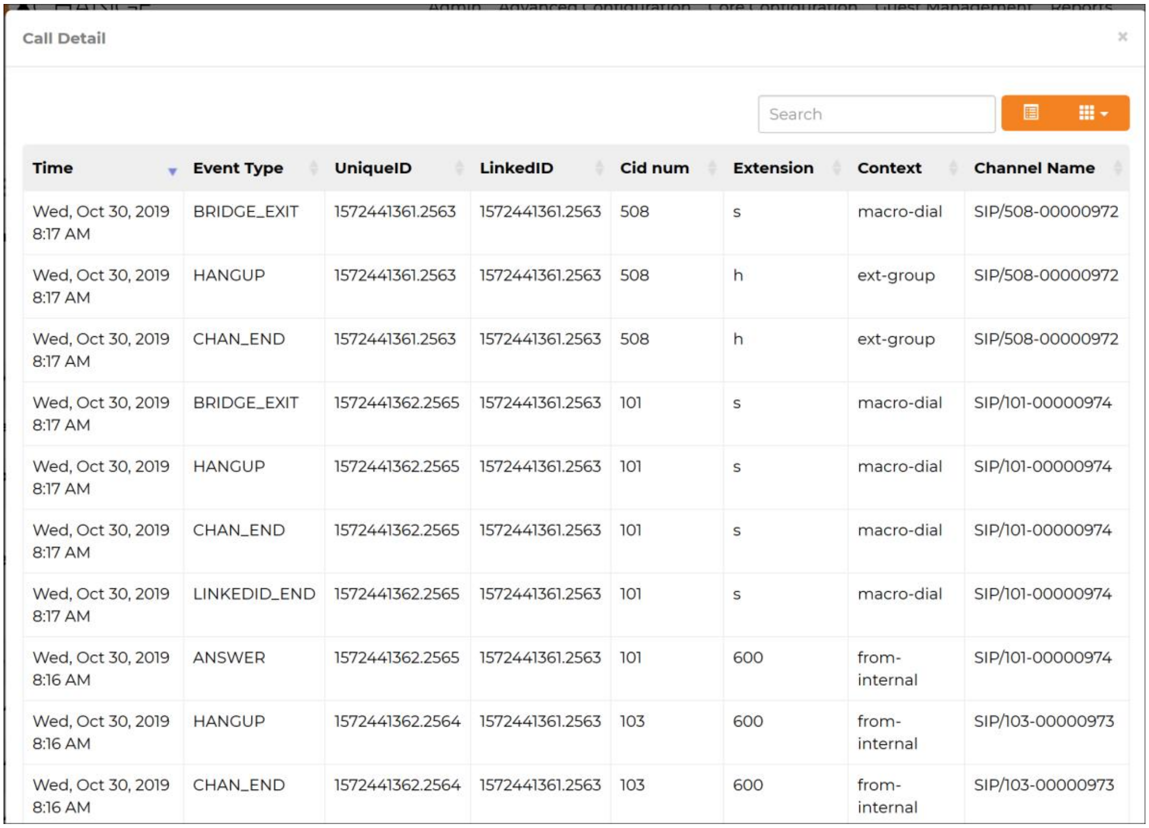
Call Event Logging

The CEL Reports is a Beta module that will allow you to view the detailed events of calls made and received by ComXchange. The information is similar-to the information found when you click on the system link found in a CDR record but can be searched based on the source and destination digits.

Navigate to Reports > Call Event Logging



Click on the Show button of the call to see details such as Context, and Channel Name



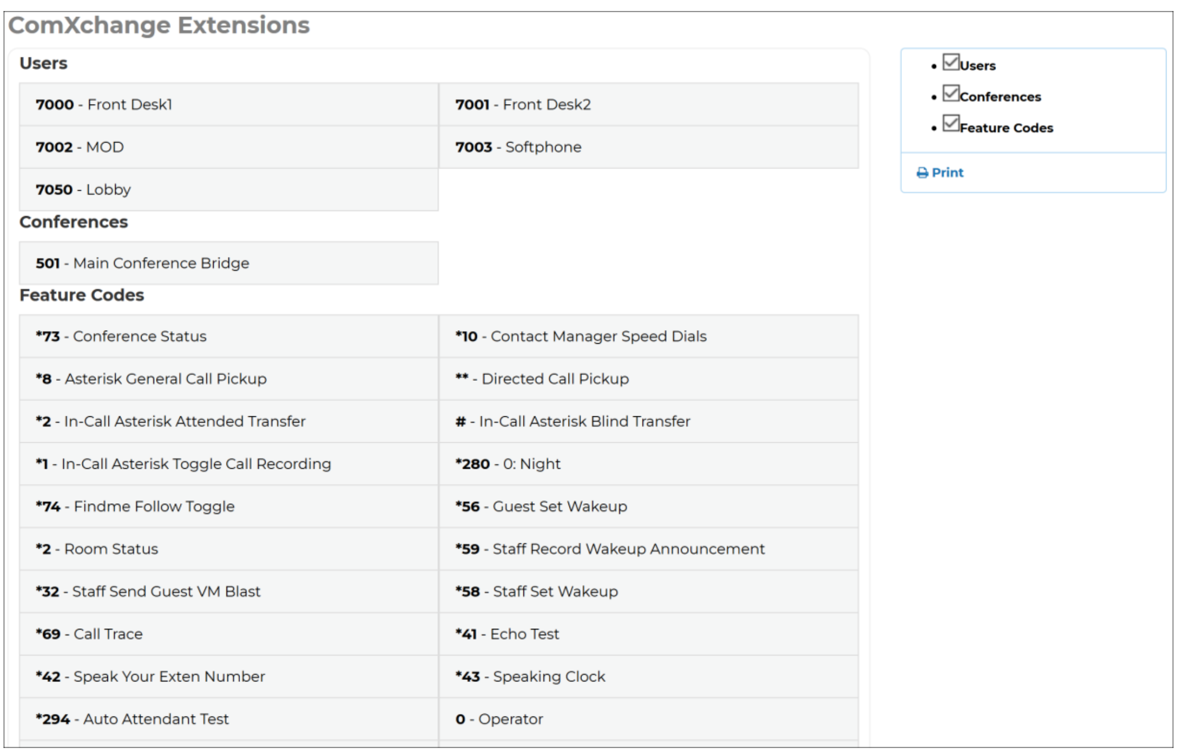
Call Detail

Time	Event Type	UniqueID	LinkedID	Cid num	Extension	Context	Channel Name
Wed, Oct 30, 2019 8:17 AM	BRIDGE_EXIT	1572441361.2563	1572441361.2563	508	s	macro-dial	SIP/508-00000972
Wed, Oct 30, 2019 8:17 AM	HANGUP	1572441361.2563	1572441361.2563	508	h	ext-group	SIP/508-00000972
Wed, Oct 30, 2019 8:17 AM	CHAN_END	1572441361.2563	1572441361.2563	508	h	ext-group	SIP/508-00000972
Wed, Oct 30, 2019 8:17 AM	BRIDGE_EXIT	1572441362.2565	1572441361.2563	101	s	macro-dial	SIP/101-00000974
Wed, Oct 30, 2019 8:17 AM	HANGUP	1572441362.2565	1572441361.2563	101	s	macro-dial	SIP/101-00000974
Wed, Oct 30, 2019 8:17 AM	CHAN_END	1572441362.2565	1572441361.2563	101	s	macro-dial	SIP/101-00000974
Wed, Oct 30, 2019 8:17 AM	LINKEDID_END	1572441362.2565	1572441361.2563	101	s	macro-dial	SIP/101-00000974
Wed, Oct 30, 2019 8:16 AM	ANSWER	1572441362.2565	1572441361.2563	101	600	from-internal	SIP/101-00000974
Wed, Oct 30, 2019 8:16 AM	HANGUP	1572441362.2564	1572441361.2563	103	600	from-internal	SIP/103-00000973
Wed, Oct 30, 2019 8:16 AM	CHAN_END	1572441362.2564	1572441361.2563	103	600	from-internal	SIP/103-00000973

Print Extensions

The Print Extensions report can be used to easily verify extensions, feature codes, and other call destinations.

Navigate to Reports > Print Extensions



ComXchange Extensions

Users

7000 - Front Desk1	**7001** - Front Desk2
7002 - MOD	**7003** - Softphone
7050 - Lobby	

Conferences

501 - Main Conference Bridge

Feature Codes

***73** - Conference Status	***10** - Contact Manager Speed Dials
***8** - Asterisk General Call Pickup	****** - Directed Call Pickup
***2** - In-Call Asterisk Attended Transfer	**#** - In-Call Asterisk Blind Transfer
***1** - In-Call Asterisk Toggle Call Recording	***280** - 0: Night
***74** - Findme Follow Toggle	***56** - Guest Set Wakeup
***2** - Room Status	***59** - Staff Record Wakeup Announcement
***32** - Staff Send Guest VM Blast	***58** - Staff Set Wakeup
***69** - Call Trace	***41** - Echo Test
***42** - Speak Your Exten Number	***43** - Speaking Clock
***294** - Auto Attendant Test	**0** - Operator

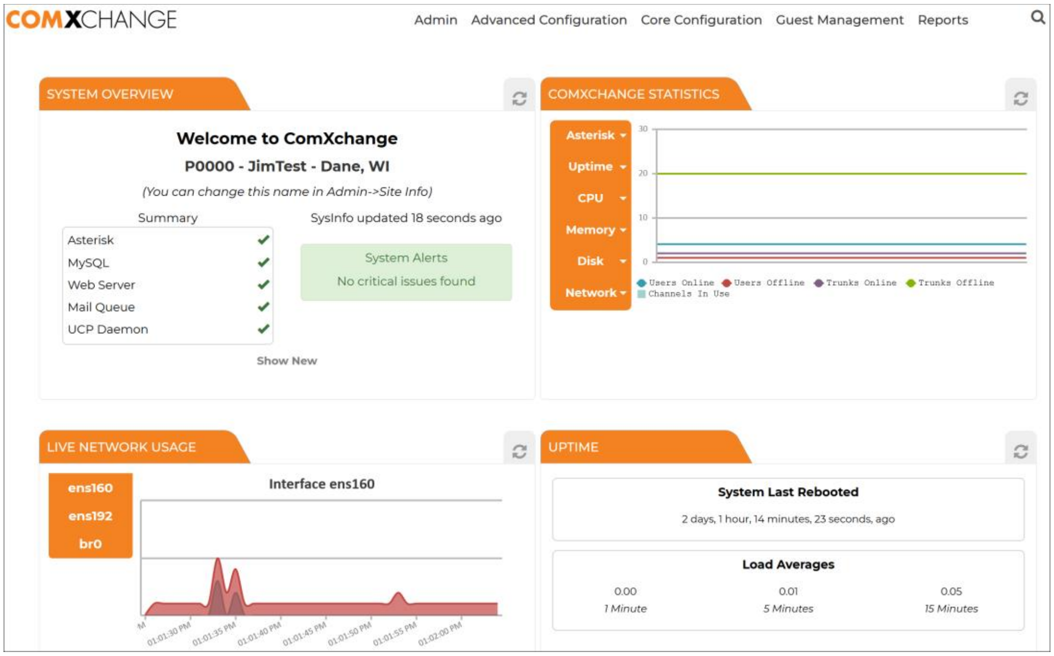
- Users
- Conferences
- Feature Codes

Print

System Dashboard

The System Dashboard is the default log in page of the ComXchange Phone System. The page displays Server Uptime, Status and Statistics.

Navigate to Report > Dashboard



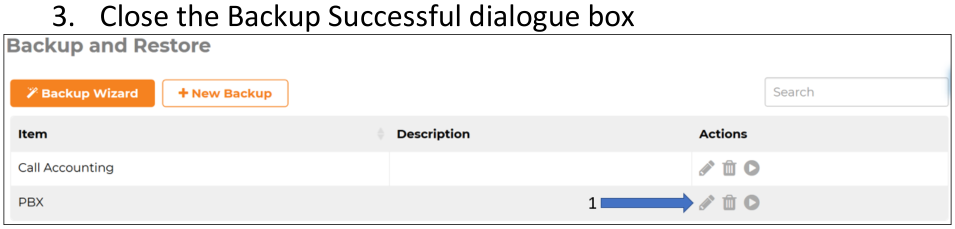
BACKUP & RESTORE

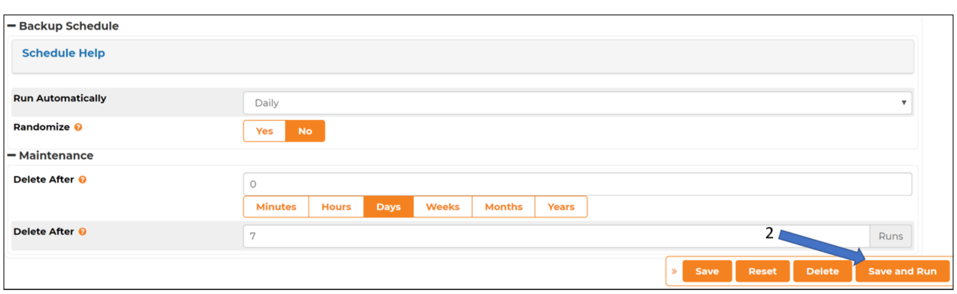
The ComXchange Phone System comes with 2 default backup jobs running that will perform a full backup of the ComXchange server and one that backs up the call accounting data. Backups are stored on the local hard drive as well as a local USB Storage Drive. Other backup destinations can be set by customizing the backup job. The backup is set to run daily and will save 7 backups, removing the oldest.

Performing a Backup on Demand

Navigate to Admin > Backup and Restore.

1. Click on the Call Accounting or PBX backup that job that you want to run
2. Click on the Save and Run button at the bottom of the page to perform a backup
3. Close the Backup Successful dialogue box

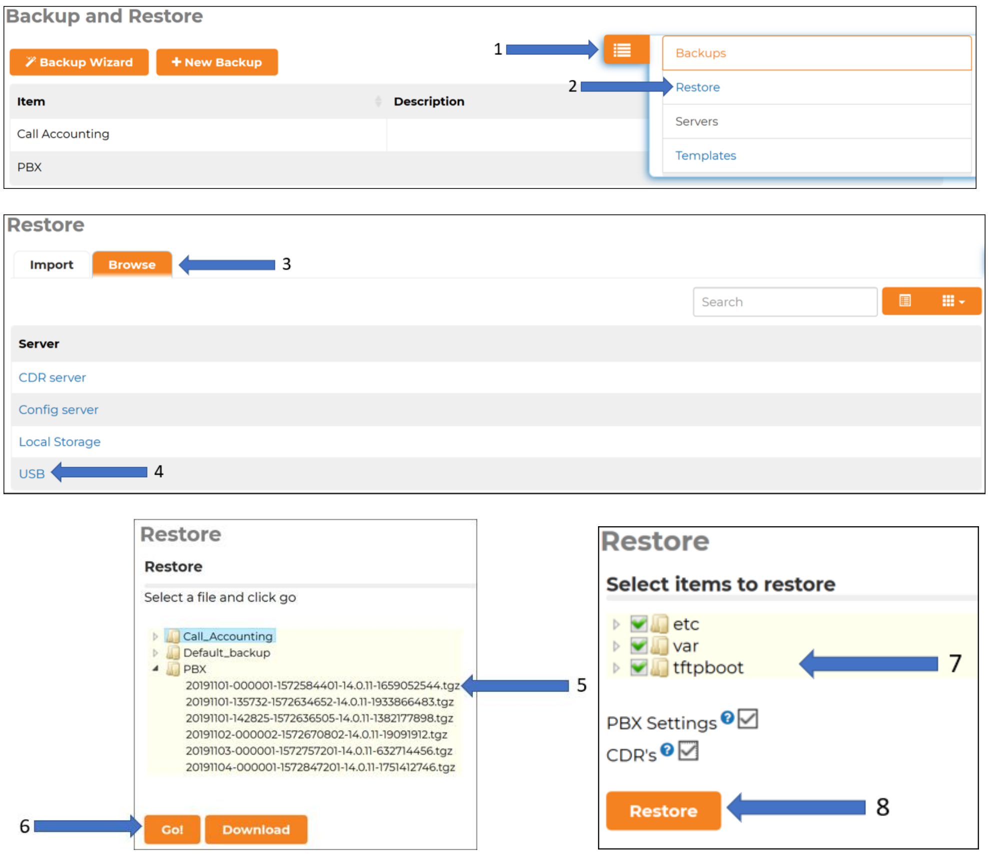




Restore from a Backup

Navigate to Admin > Backup and Restore.

1. Click on the fly out menu on the right-side
2. Click on Restore
3. Click on the Brose Tab (or import from a file)
4. Click on Local Storage or USB in the right-side menu.
5. Expand the Backup Job folder and choose the backup file.
6. Click on Go.
7. Select the items to restore.
8. Click on restore.



User Mangement

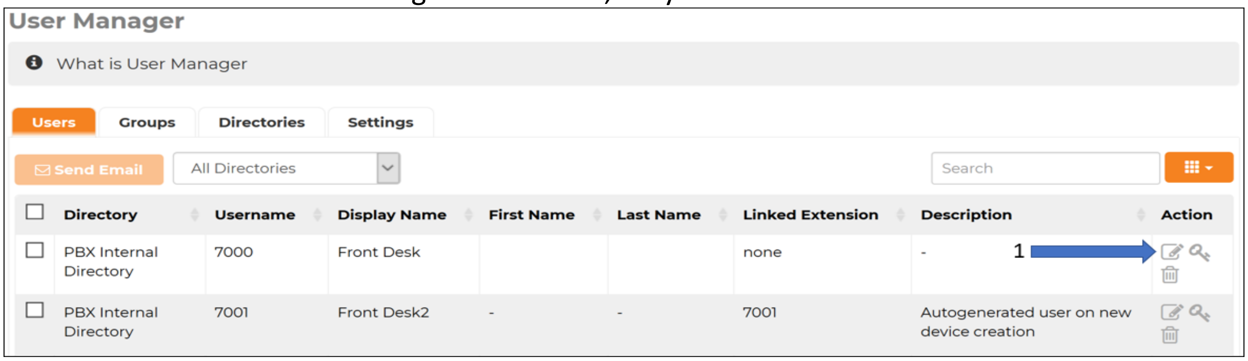
The User Management Module is where you manage user's passwords and permissions for other modules like Call Accounting and UCP. Users are automatically created and associated with an extension when an extension is made. You can also create users that aren't attached to an extension to do managerial activities in Call Accounting and the Hotel Dashboard. Below we will make changes to allow a user to log into the User Control Panel.

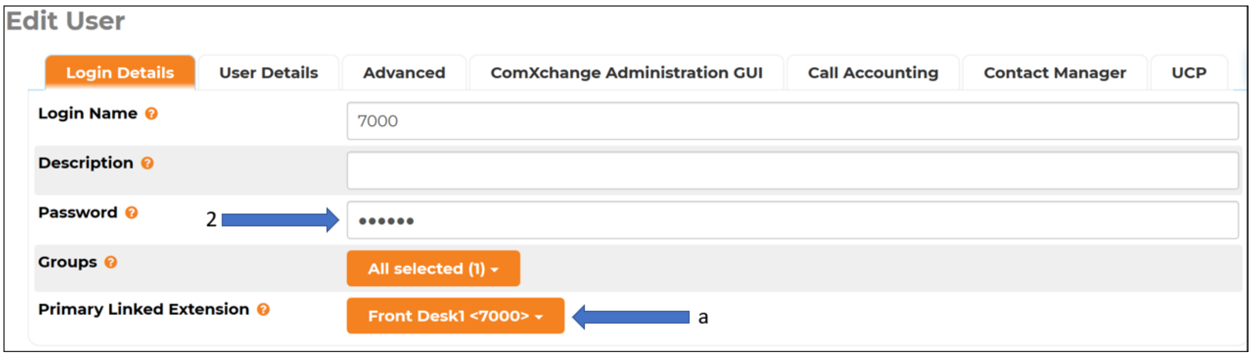
Navigate to Admin > User Management - Users

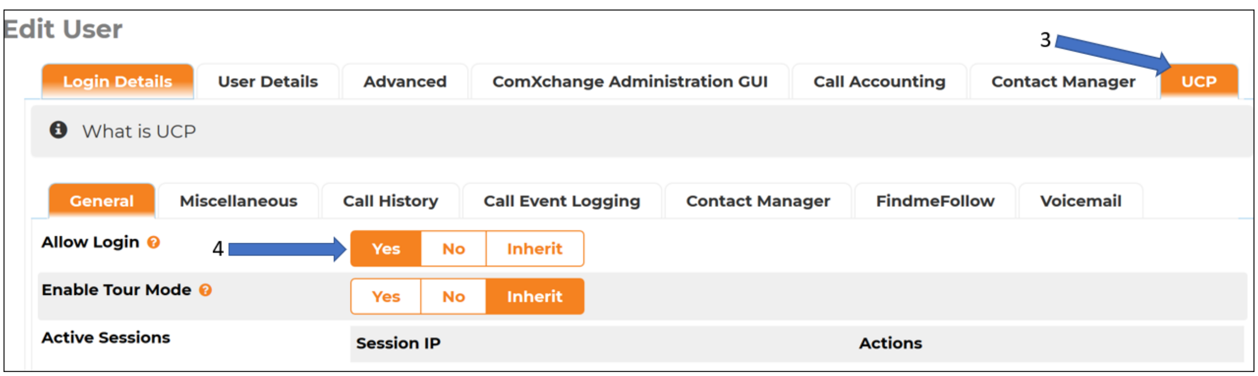
Users

1. Click on the edit Action icon for the extension that you want to grant access to the UCP
2. Change the password to a default password for the user to log in with
   a. Note this is where you can see the Primary Linked Extension
3. The Call Accounting tab is where you allow access and permissions to the Call Accounting features

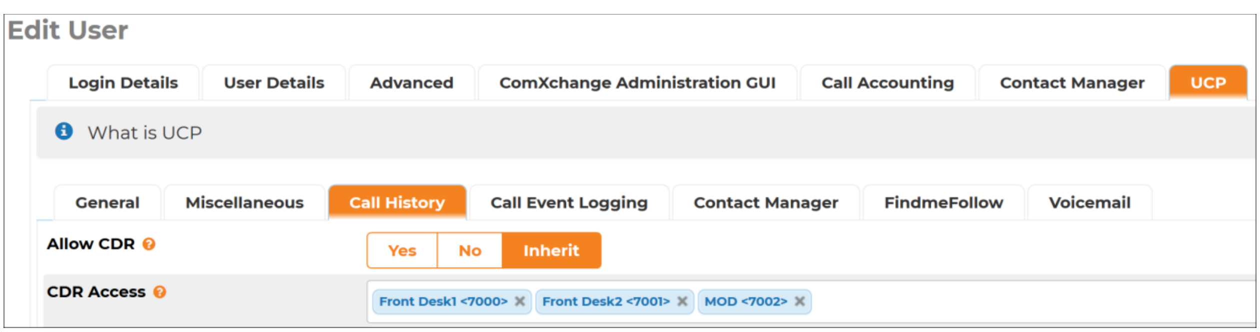
a. Typically, you will set this up for a user that isn't linked to an extension
4. UCP tab is where you would enable access and grant permissions for a user to access their UCP and manage their extensions features like voicemail and call history
5. Set the Allow Login to Yes
    a. Once the user is granted access, they should be able to access the UCP features







In the UCP tab you can give users access to other extensions for example below User 7002 will be allowed to access the Call History for extension 7000, 7001, and 7002.

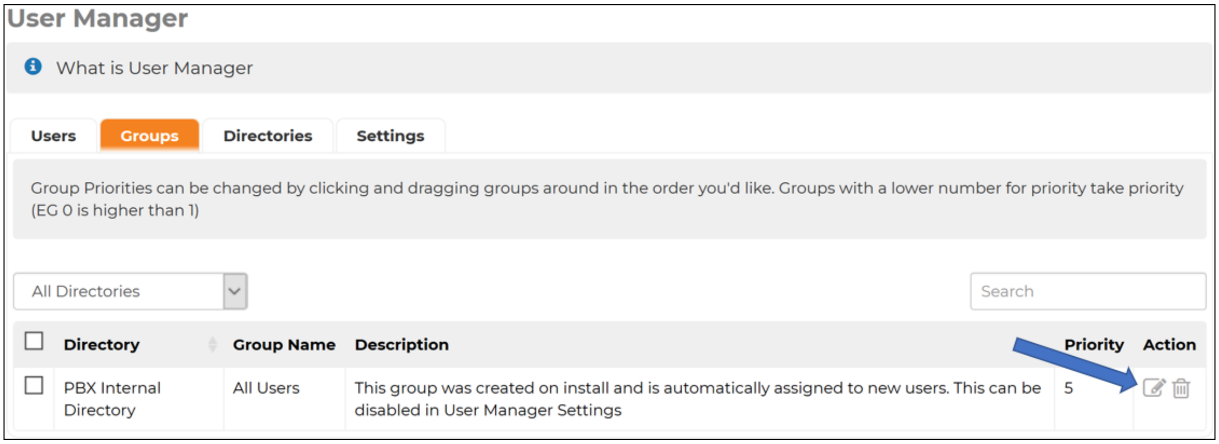


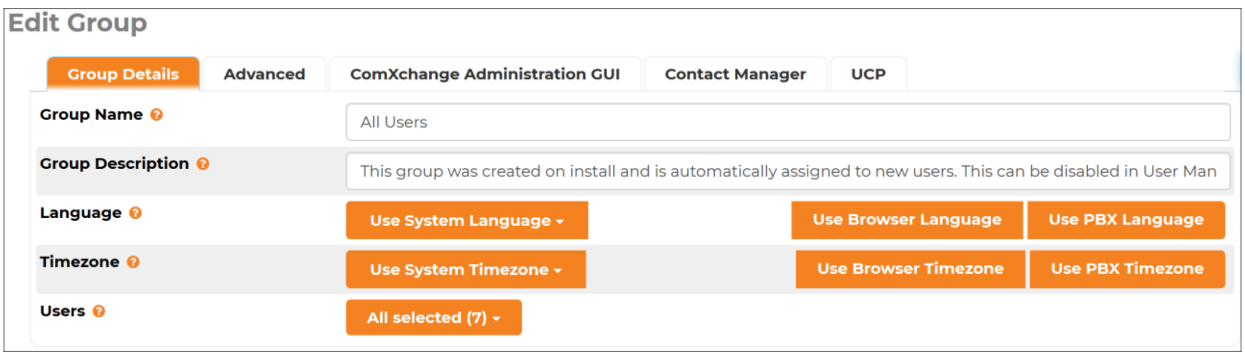
Groups

Can be used to put users with similar permissions together. By default, there is an All Users group that all new users (with or without an extension) is added to. This Group defines the default permissions that are assigned to these users at creation.

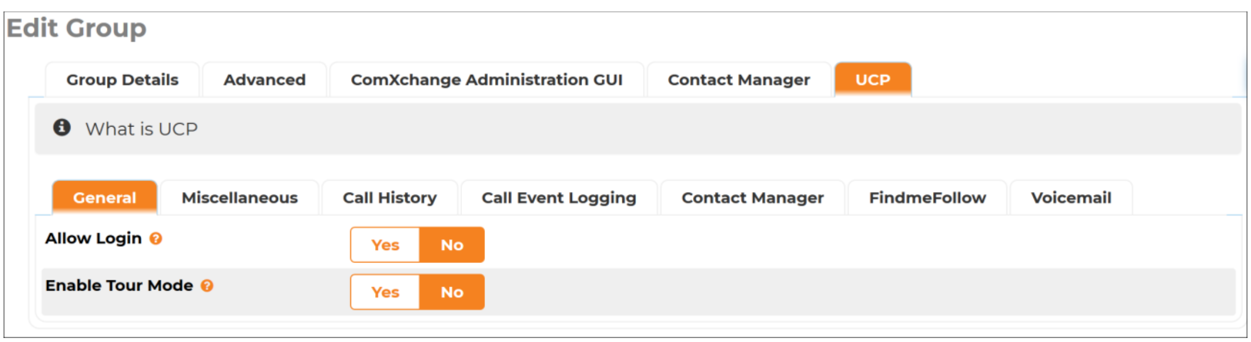
Navigate to Admin > User Management – Groups

Click on the edit Action icon to view the default permissions that Users of this group will be assigned





You can also change what features are accessible to the users by default.

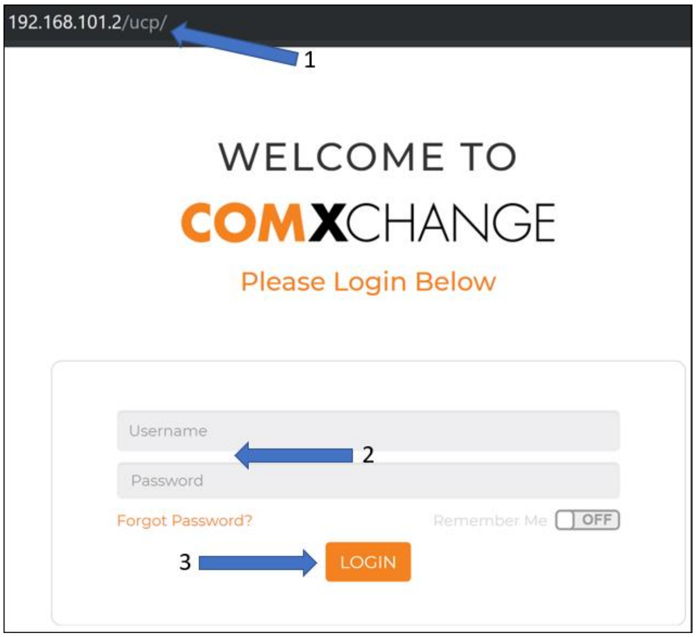


User Control Panel

The User Control Panel is a web menu that allows users to do some basic management of their extension. The User logs in by typing “http://< LAN IP address of the ComXchange>/ucp” into the address bar of a web browser. Or an easier method for users to access it is to create a desktop shortcut with the location of http://192.168.101.2/ucp which will directly open the login page. The User Control Panel allows users to log into a portal with their extension (by default) and password set in the User Management Module. In the UCP users can set up a dashboard and add widgets that allow them to see the Extensions call history, listen to and manage their Voicemail, setup Follow Me rules, Call forwarding, and other settings. Users must first be set up in the User Management Module to be allowed to log in to the UCP.

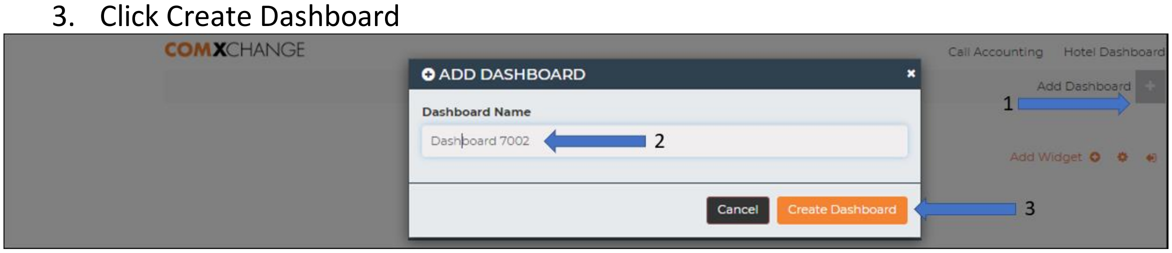
UCP Login

1. Navigate to the web login by entering http:192.168.101.2/ucp
2. Enter the user credentials and password
3. Click Login



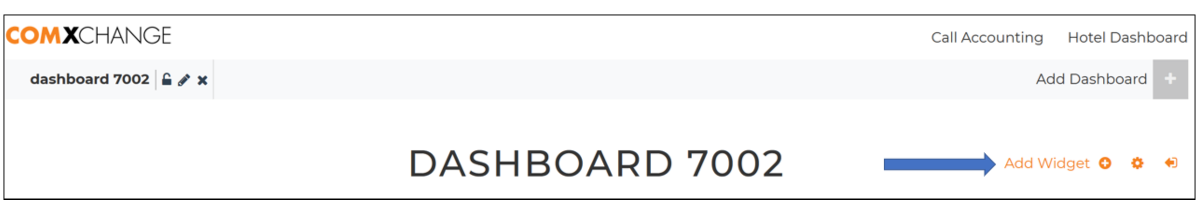
Create a UCP Dashboard

1. Click on add a Dashboard
2. Fill in Dashboard Name
3. Click Create Dashboard



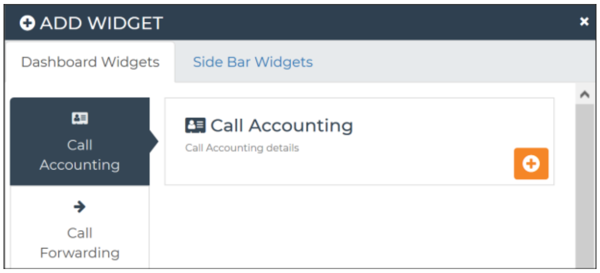
UCP Widgets

Once a dashboard is created a user can add Widgets to access different functions. To add the different functions Click on the Add a Widget Button on the dashboard.



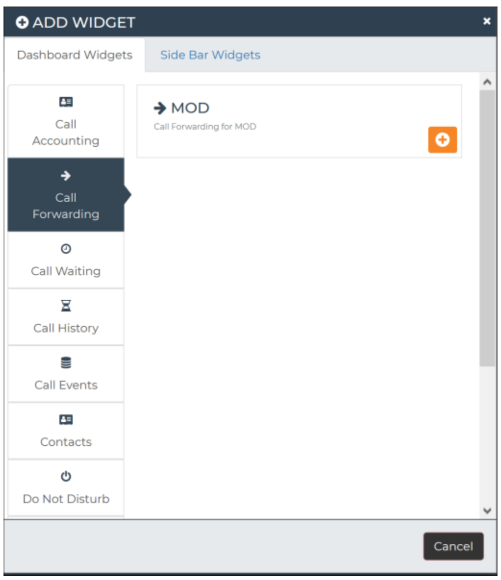
Call Accounting Widget

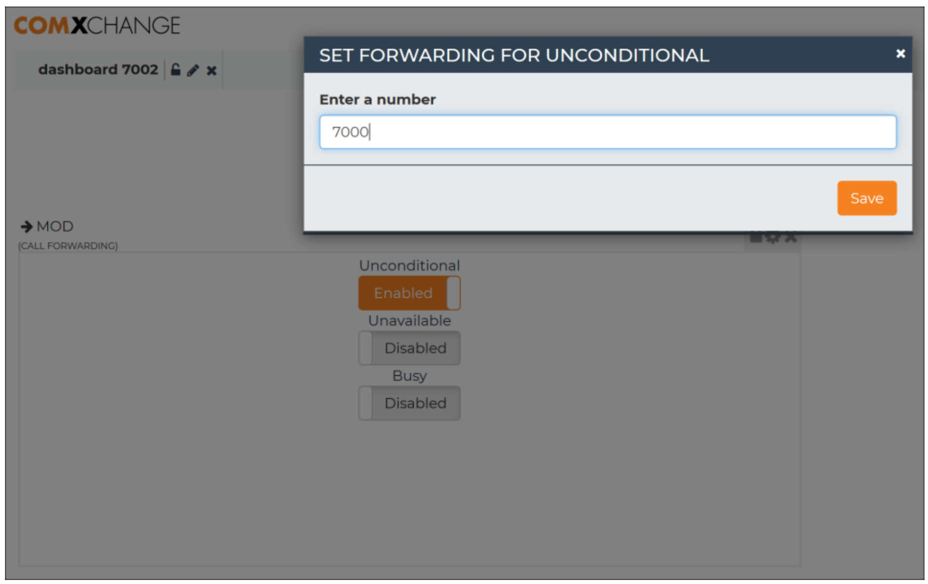
The Call Accounting Widget adds a link to Launch the Call Accounting Module for ComX14.



Call Forwarding Widget

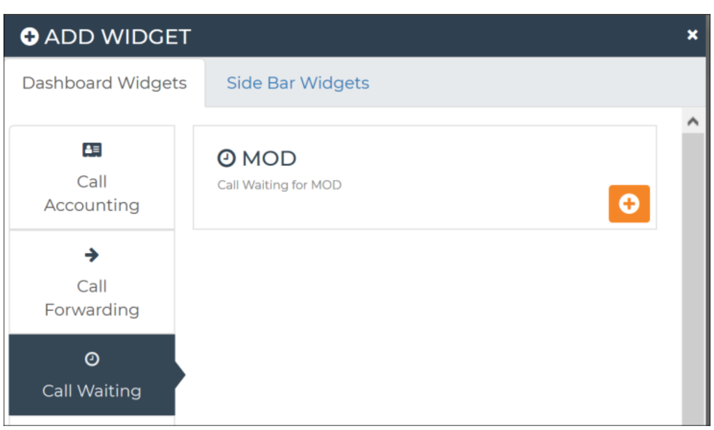
The Call Forwarding Widget allows a User to control the forwarding functionality of the extension. This can be set as unconditional (always forwarding), Unavailable (if the phone endpoint becomes unresponsive), and Busy (if the Extension has an active call).

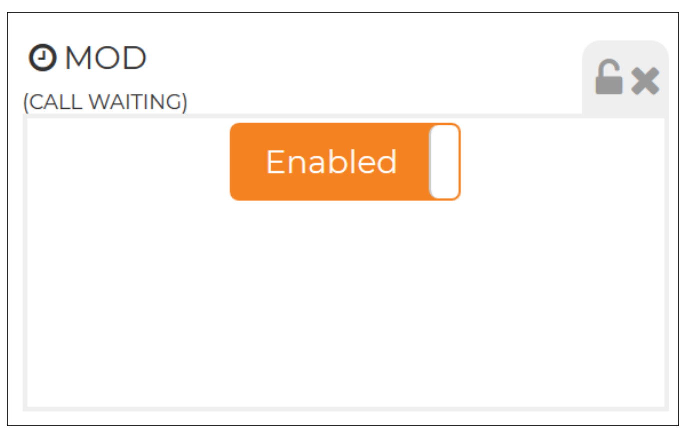




Call Waiting Widget

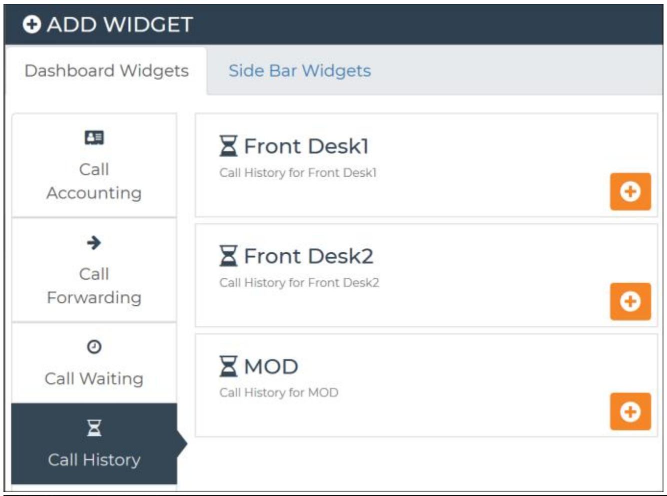
The Call Waiting Widget controls whether Call Waiting is enabled or disabled for an extension.

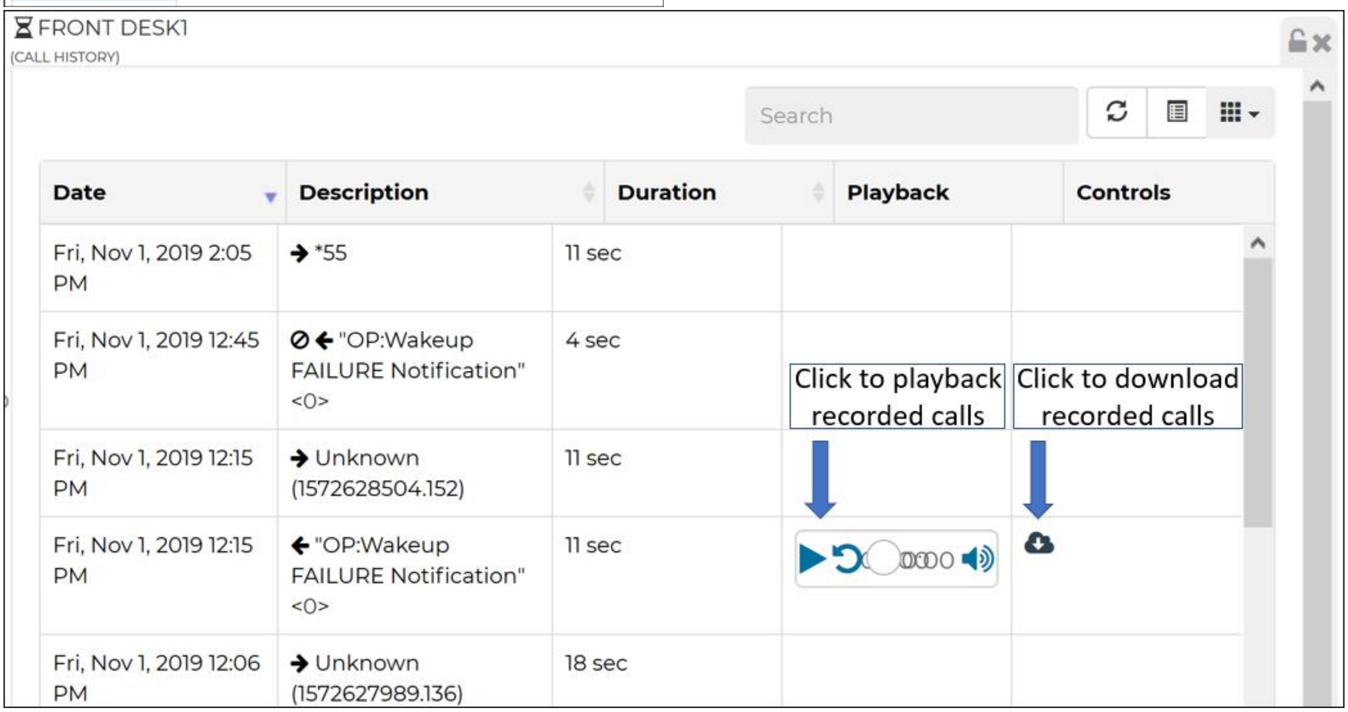




Call History

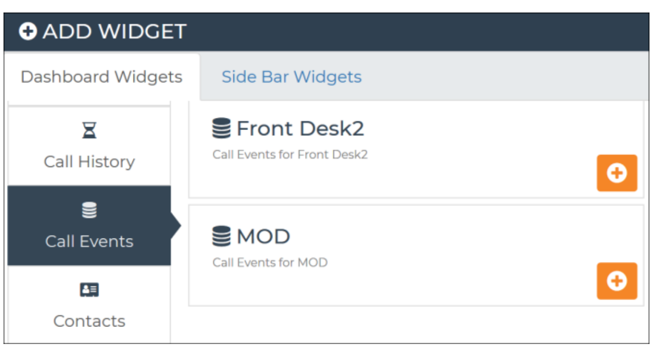
The Call History Widget allows a User to view and search for calls and listen to recorded calls to and from the Extension(s) that the user is given access to in the UCP section for the user in the User Management Module. Below User 7002 has access to both Front Desk extensions.

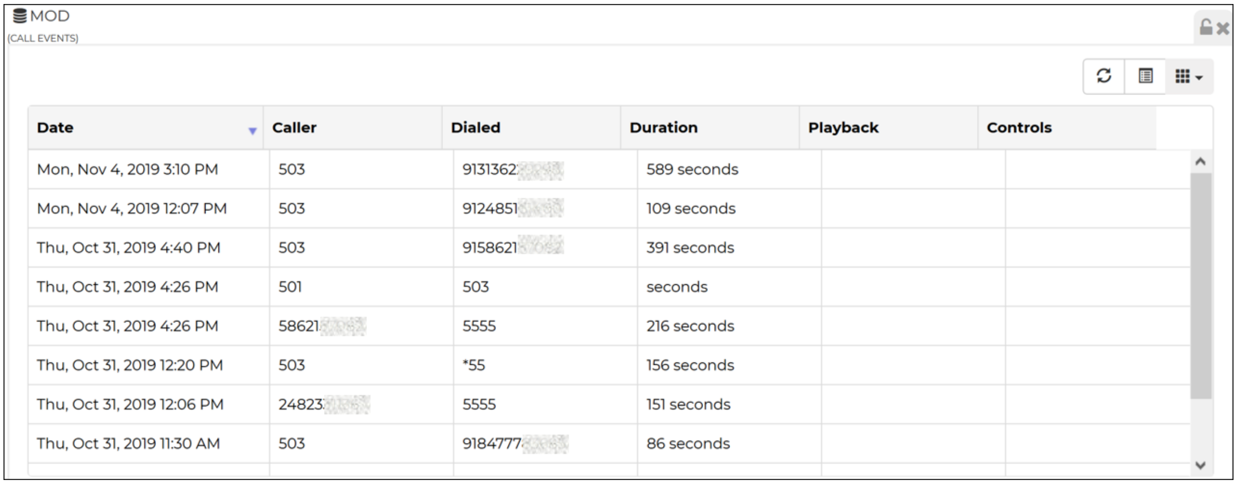




Call Events Widget

The Call Events Widget allows a user view the Date, Caller, Dialed Number, and Duration of a call.

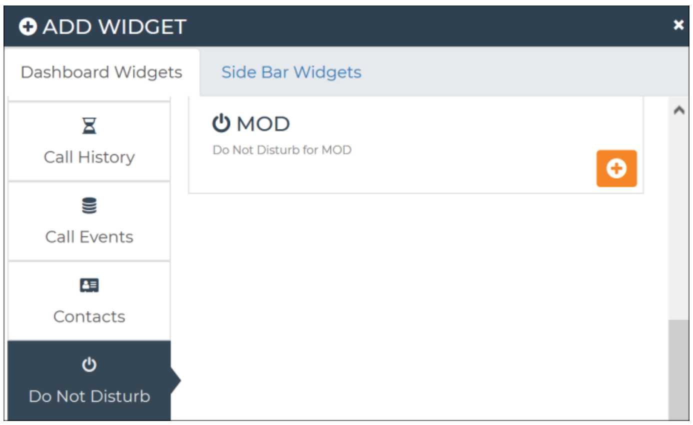


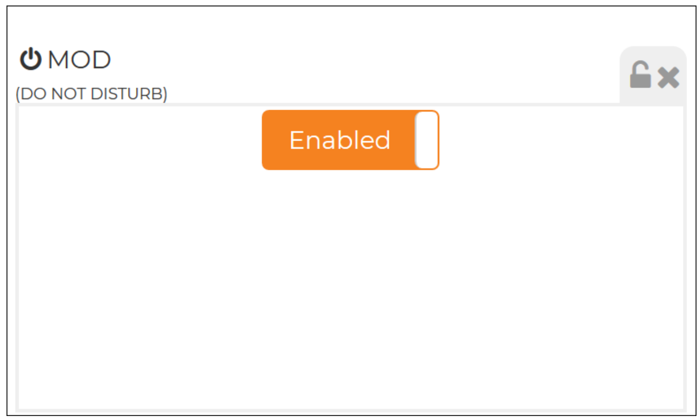


Date	Caller	Dialed	Duration	Playback	Controls
Mon, Nov 4, 2019 3:10 PM	503	9131362	589 seconds		
Mon, Nov 4, 2019 12:07 PM	503	9124851	109 seconds		
Thu, Oct 31, 2019 4:40 PM	503	9158621	391 seconds		
Thu, Oct 31, 2019 4:26 PM	501	503	seconds		
Thu, Oct 31, 2019 4:26 PM	58621	5555	216 seconds		
Thu, Oct 31, 2019 12:20 PM	503	*55	156 seconds		
Thu, Oct 31, 2019 12:06 PM	24823	5555	151 seconds		
Thu, Oct 31, 2019 11:30 AM	503	9184777	86 seconds		

Do Not Disturb

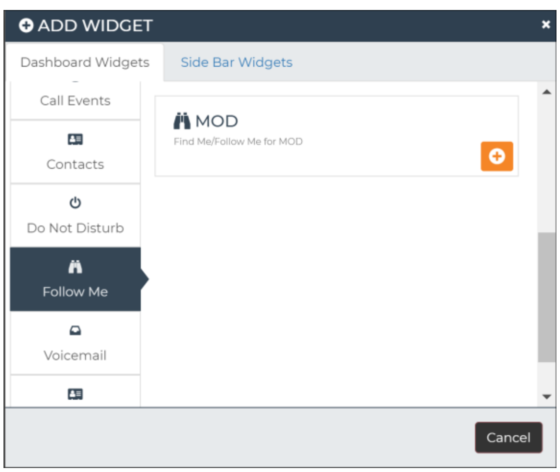
The Do not Disturb Widget allows the User to control the endpoints DND functionality.

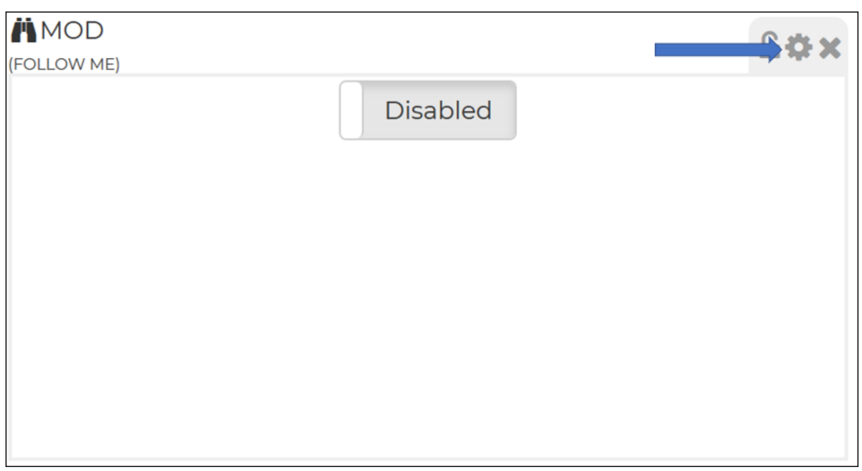


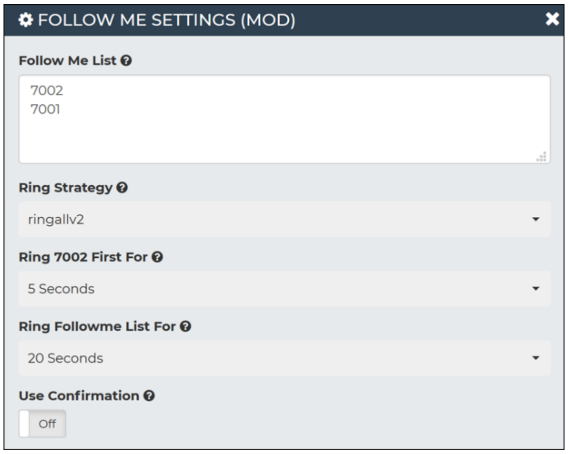


Follow Me Widget

The Follow Me Widget allows an incoming call to an extension to be received on multiple phones. Once added the user can program the Follow Me functionality by clicking on the settings cog wheel icon.

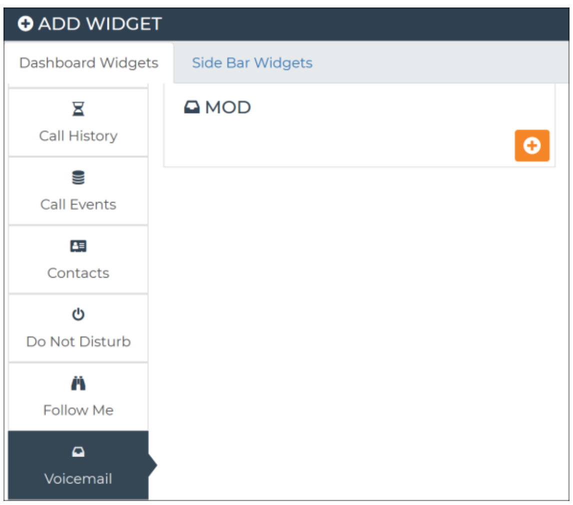


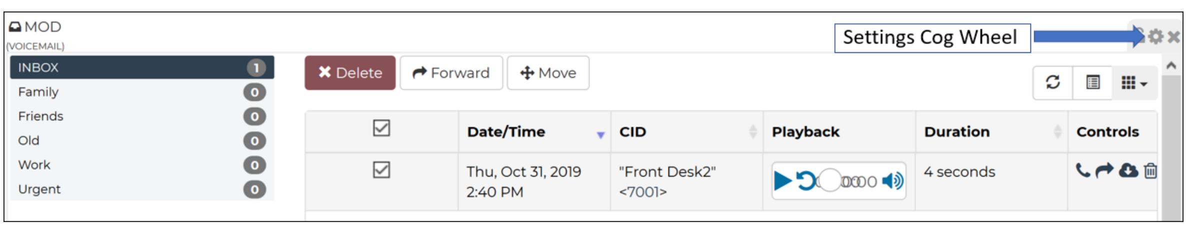




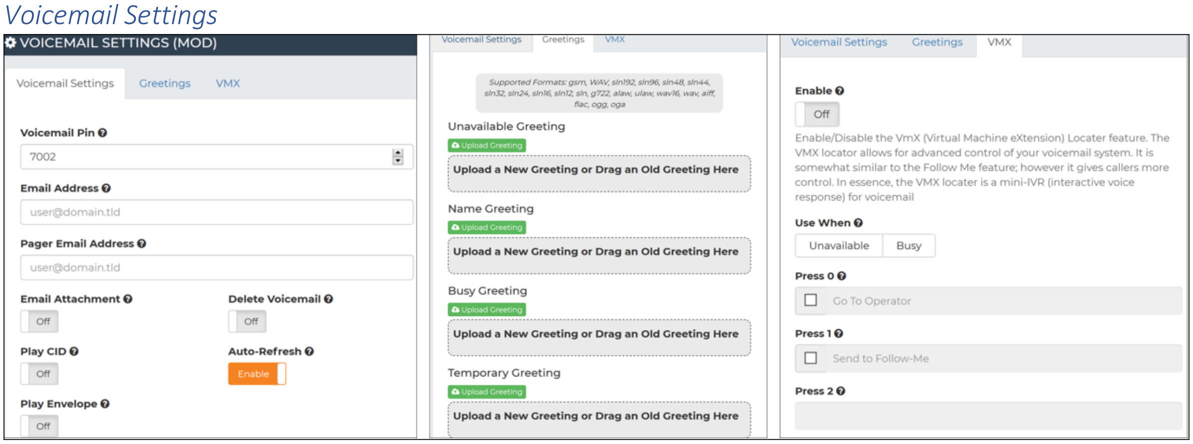
Voicemail Widget

This menu allows a user to manage and listen to their voicemail through a web interface. The User can also control some of the Voicemail settings by clicking the settings cog wheel



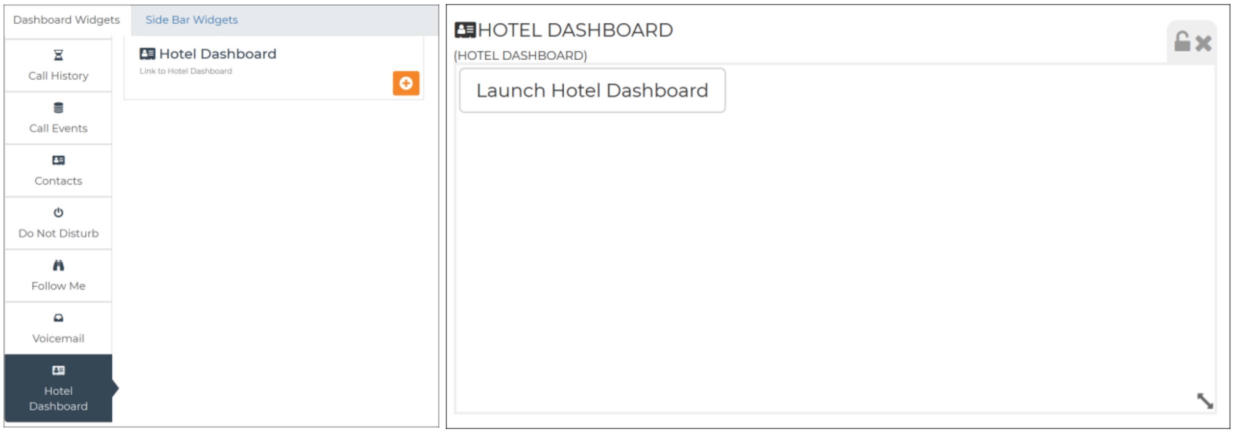


Voicemail Settings



Hotel Dashboard Widget

The Hotel Dashboard Widget adds a link to Launch the Call Accounting Module for ComX14.

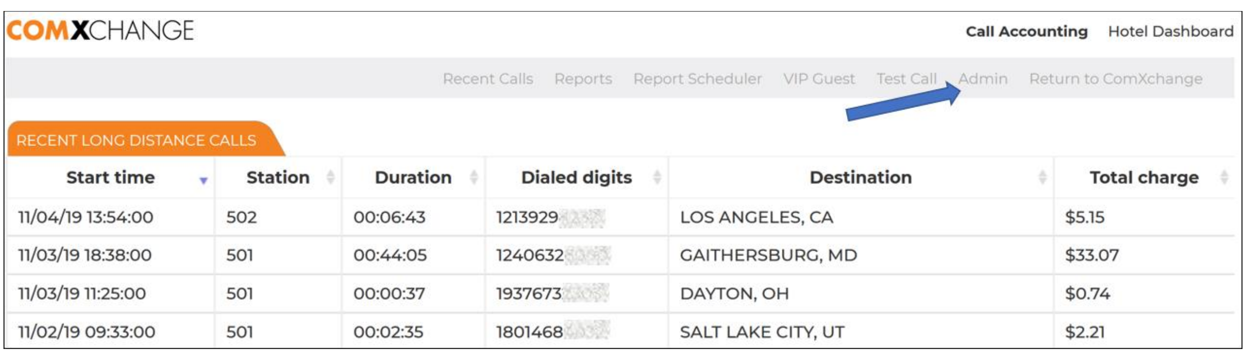


Call Accounting Setup

The Call Accounting module is used to configure the Call Accounting Server. The Call Accounting Server has two interfaces that can be configured to receive data from the PBX's SMDR interface, process that information, then send data out the PMS interface to the PMS Systems to charge billable phone calls to guests of the hotel. The call accounting must be licensed and the V&H file (Vertical and Horizontal file used to assess the distance between two points for call rating) must be ordered and delivered from 360 Networks before setup can begin. Below are the basic steps to install the V&H files and prepare the Call Accounting Server module to interface with a PMS system.

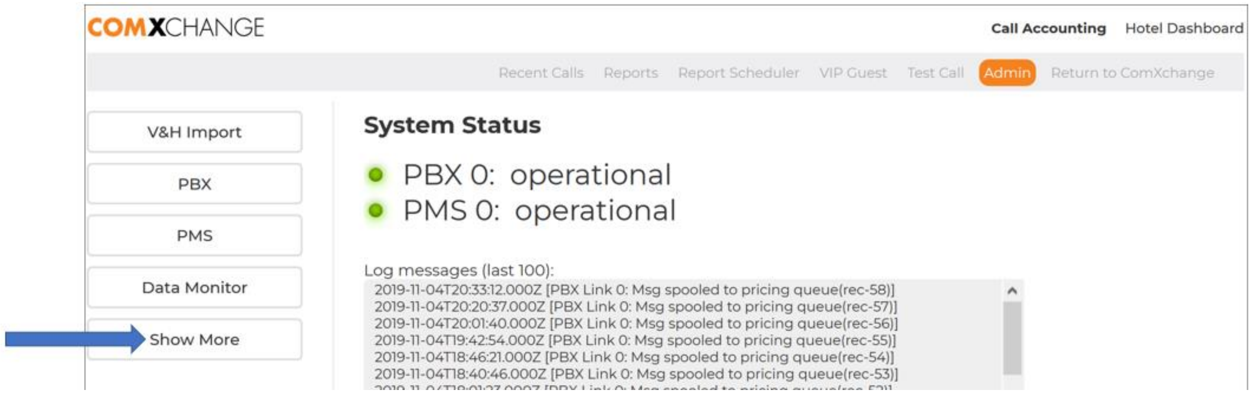
Navigate to Reports > Call Accounting

This will open to a recent call summary page. To open the Call Accounting Settings, click on the Admin Button.

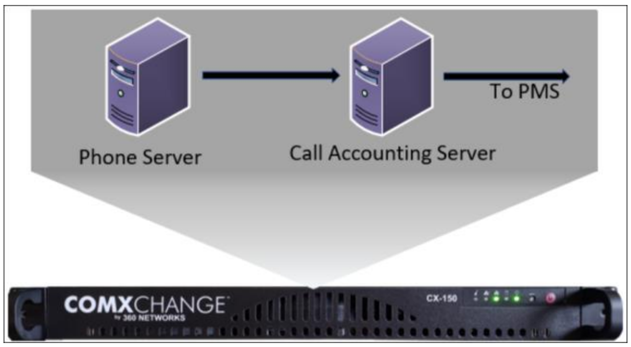


Start time	Station	Duration	Dialed digits	Destination	Total charge
11/04/19 13:54:00	502	00:06:43	1213929	LOS ANGELES, CA	$5.15
11/03/19 18:38:00	501	00:44:05	1240632	GAITHERSBURG, MD	$33.07
11/03/19 11:25:00	501	00:00:37	1937673	DAYTON, OH	$0.74
11/02/19 09:33:00	501	00:02:35	1801468	SALT LAKE CITY, UT	$2.21

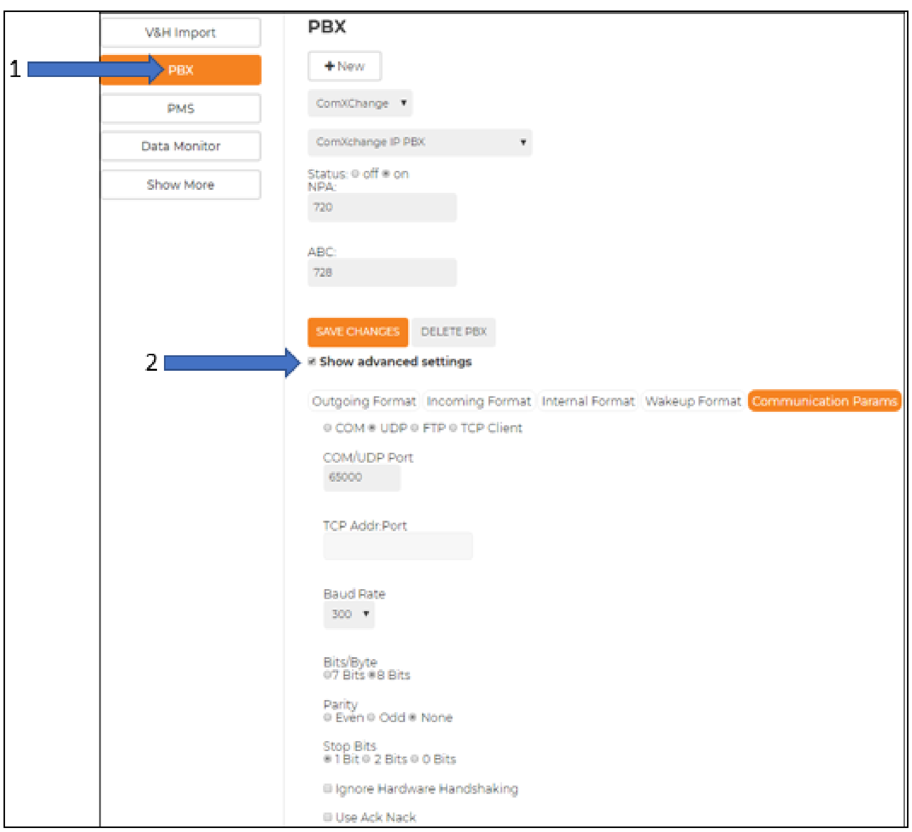
You can expand all options by clicking on the Show More button.

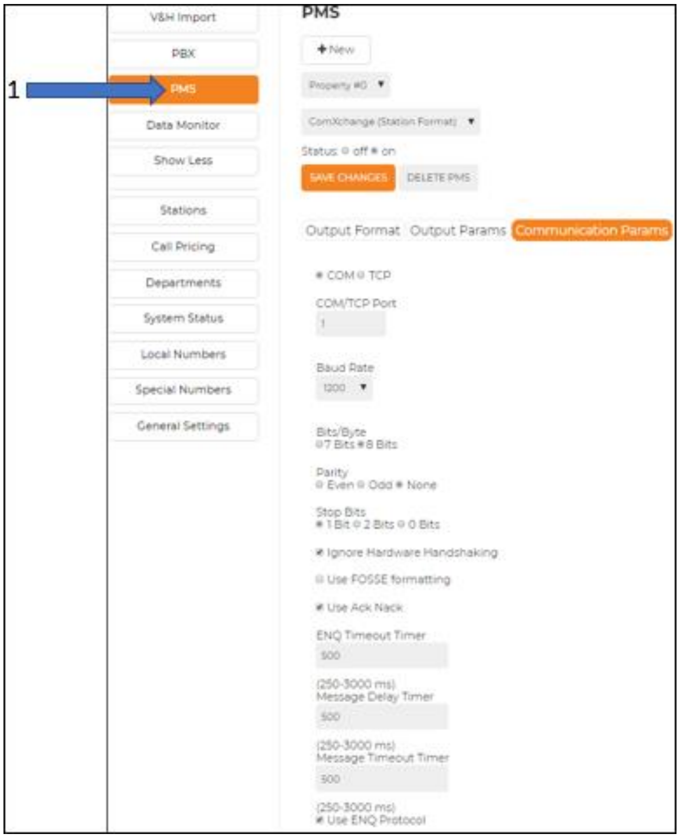


Call Accounting Interfaces



There are two interfaces that can be configured in the Call Accounting Module. The first is the interface between the PBX and the Call Accounting Server which should never need to be changed as it is set up to receive the SMDR information from the PBX by default. The second is the interface between the Call Accounting Server and the PMS System. This interface may need to be changed based on the PMS vendor.

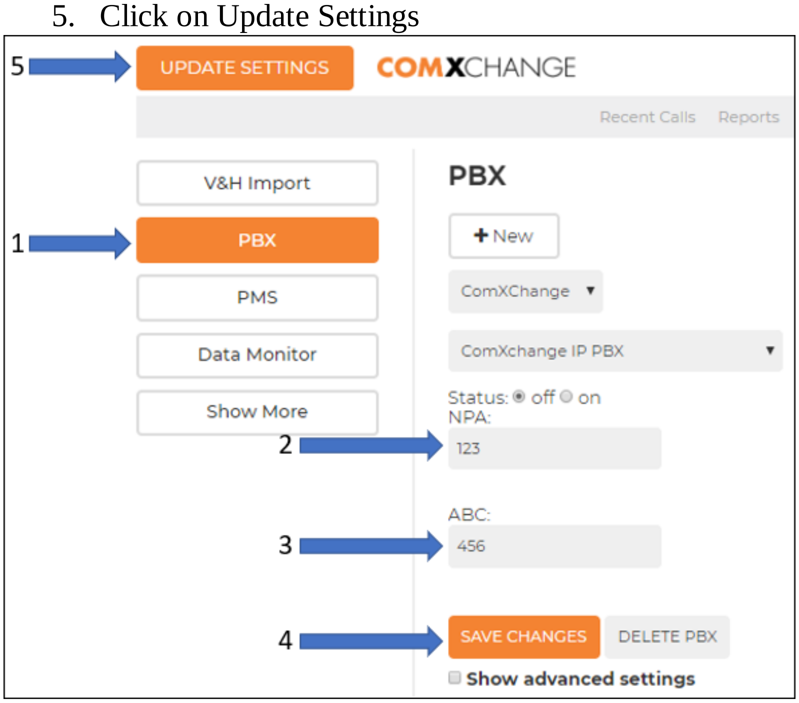




Add Local Numbers

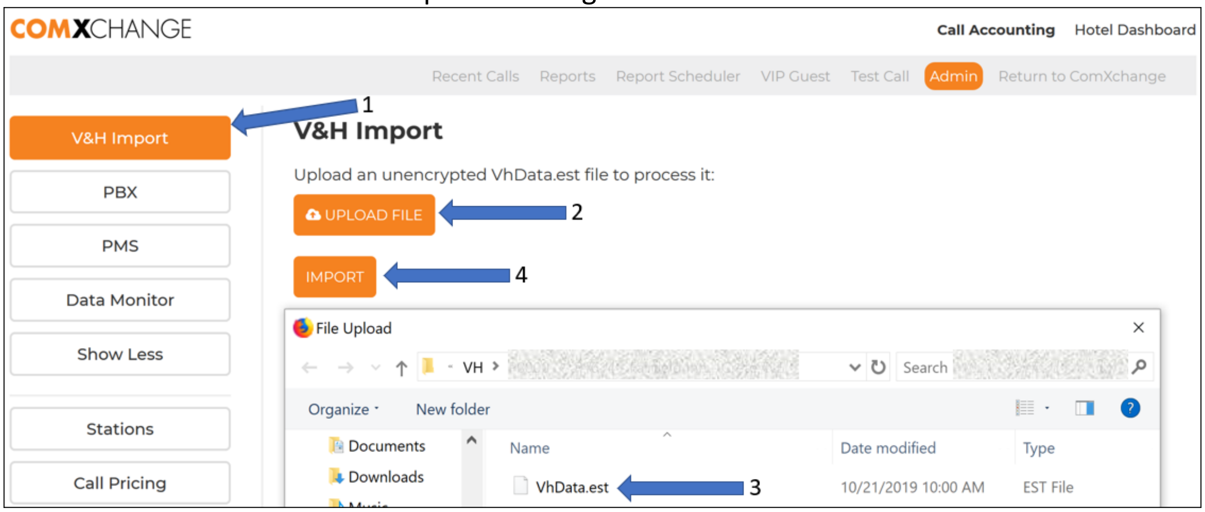
Add the Local Number NPA (Number Plan Area) and exchange Information. This will be the area code and three numbers directly after the area code.

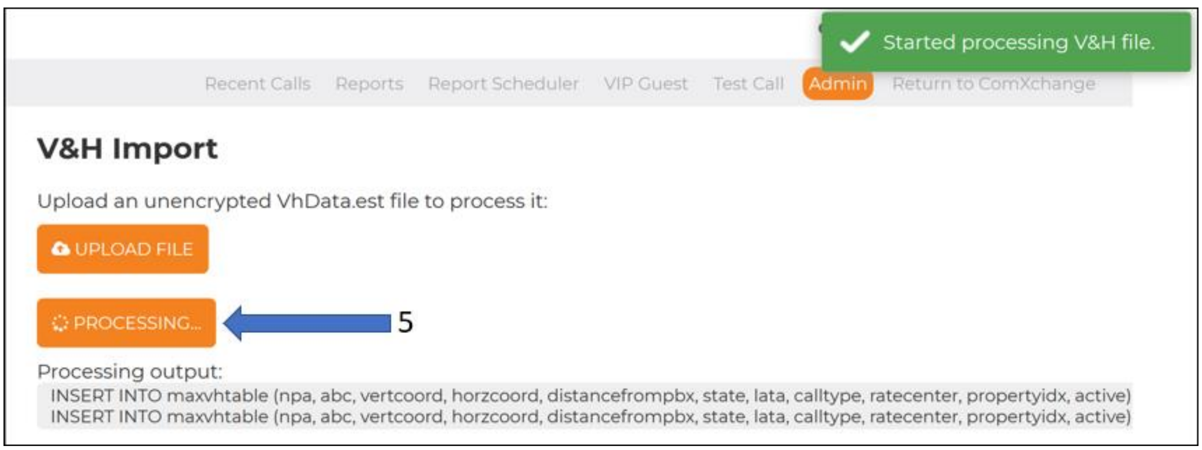
1. Click on the PBX Button
2. Add the Area Code in the NPA field
3. Add the exchange numbers in the ABC field
4. Click on the Save Changes Button
5. Click on Update Settings

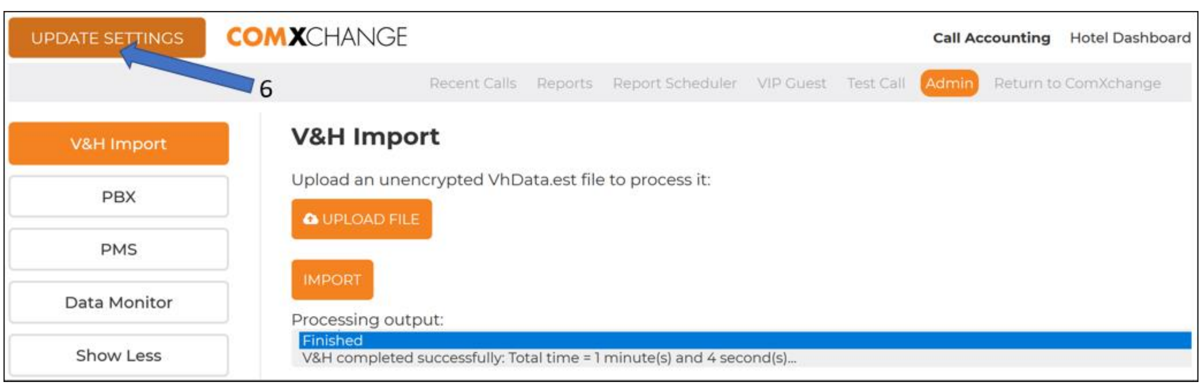


Import V&H File

1. Click on V&H Import Button
2. Click on the Upload File Button
3. In the popup navigate to the VhData.est file
4. Click on Import
5. Wait for the File to process.
6. When finished Click on Update Settings





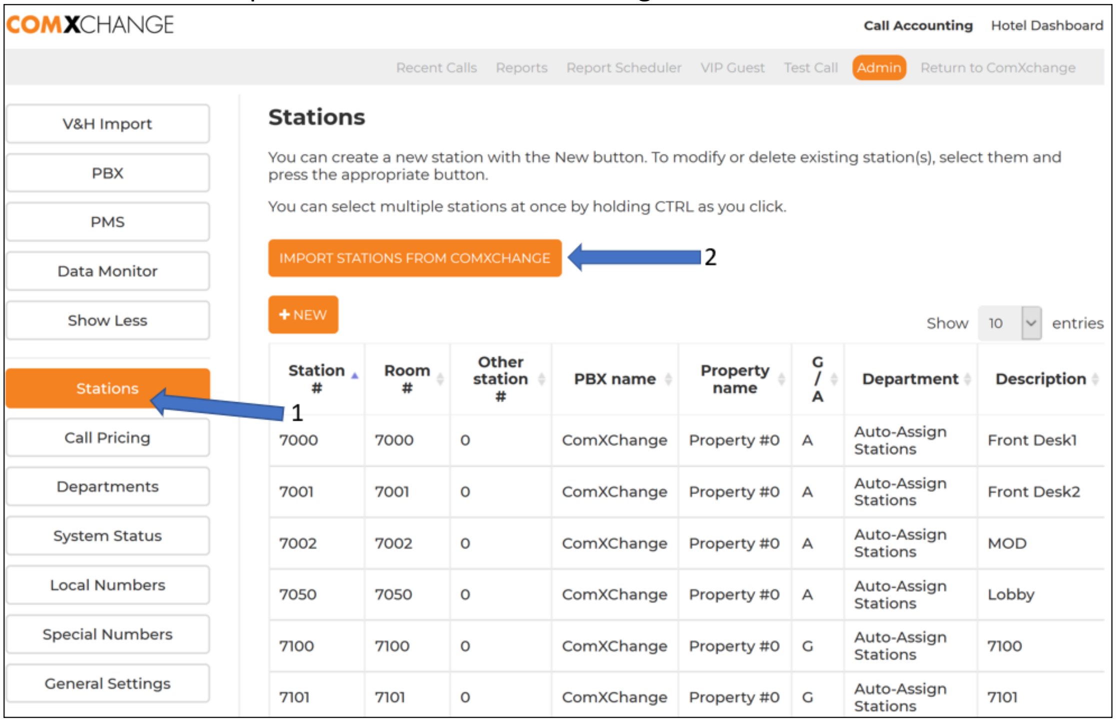


Import stations

In-order-for calls to be processed the Call Accounting must be made aware of the stations (extensions) in ComXchange.

1. Click on the Stations Button

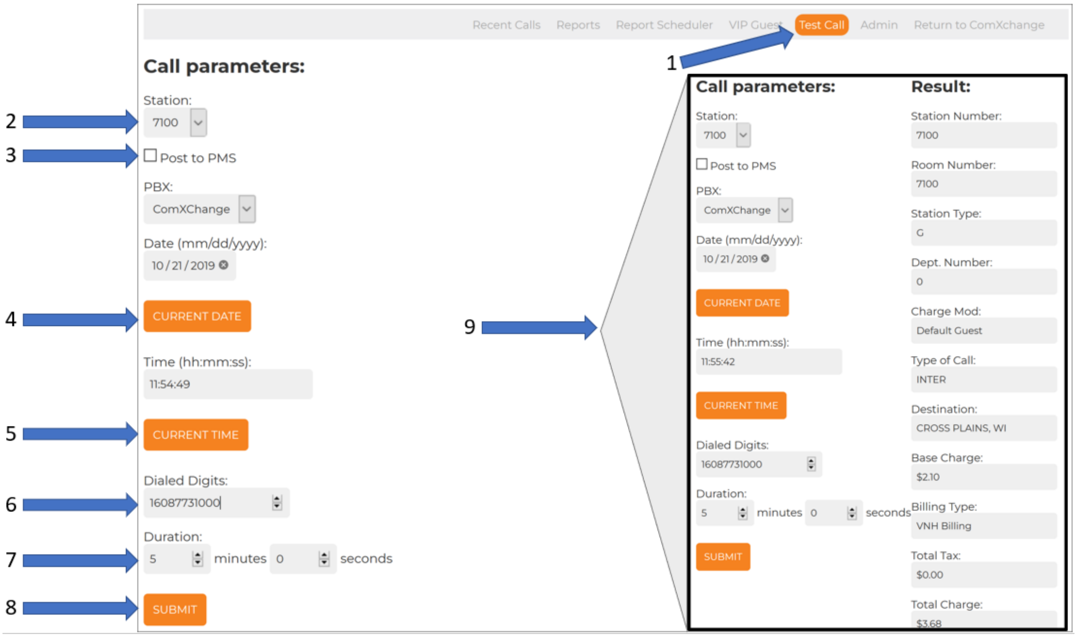
2. Click on the Import Stations From ComXchange Button



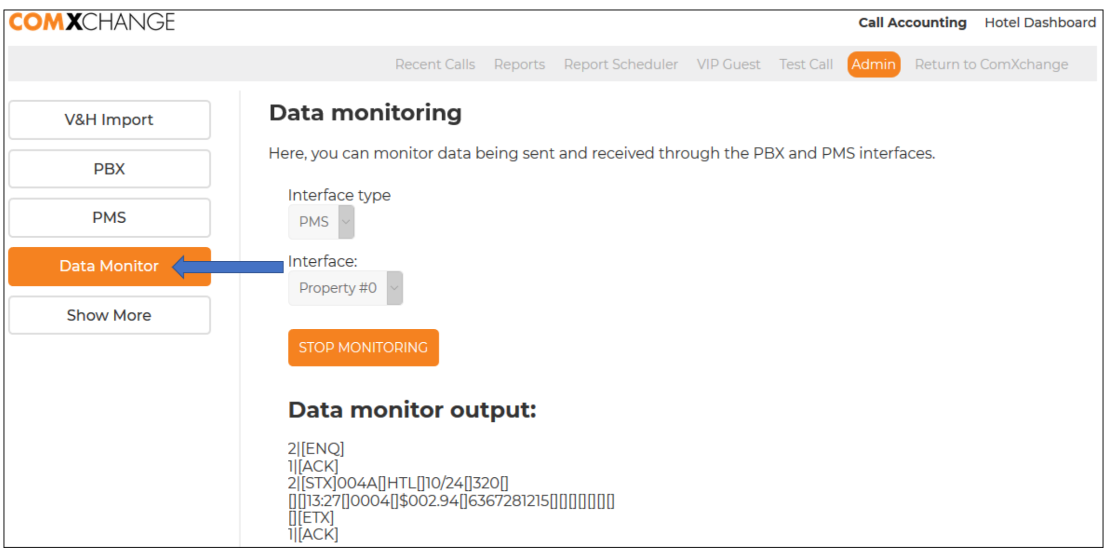
Test Call

To verify that the Call Accounting Module is configured to price calls use the test call feature to simulate a call.

1. Click on the Test Call Button
2. Choose a station to test from in the dropdown box
3. Click Post to PMS if you want to send the call information over the PMS connection
4. Click on the Current Date Button
5. Click on the Current Time Button
6. Enter a phone number in the Dialed Digits Field
7. Enter a Call duration (must be more than 2 minutes)
8. Click on Sumit
9. View the results



To monitor the information that is being sent out from the call accounting module Click on the Data Monitor Button while a test call is being sent to the PMS.



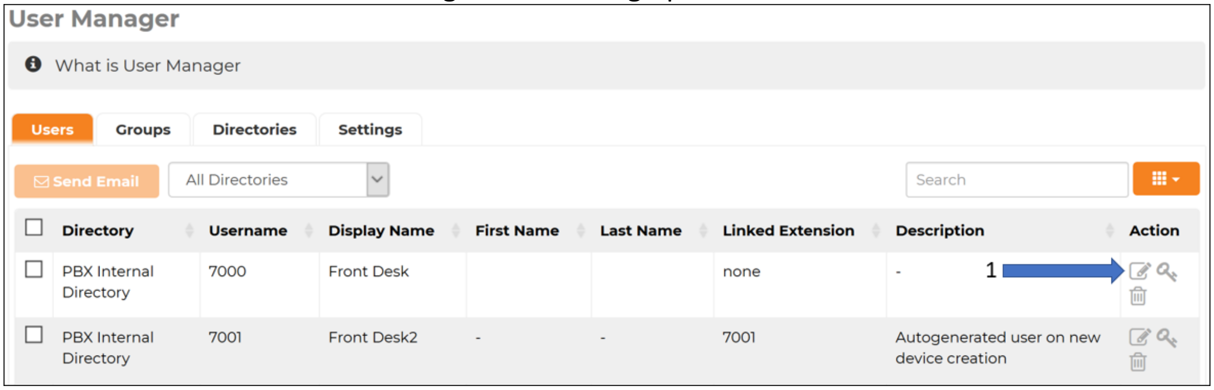
Staff Access to Call Accounting

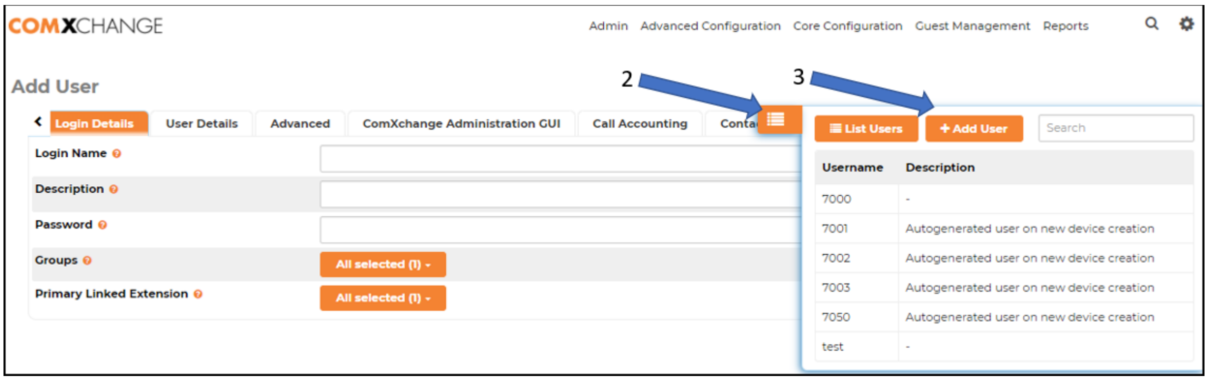
To Staff to access the Call Accounting reports and VIP rooms that won't be charged you need a user that has access to the UCP and permissions for the Call Accounting Module. While this can be an extension User that is given access in the example below, we will create a User that is used to access the Call Accounting Module. (Note: This HotelUser was created earlier for accessing the Hotel Dashboard.)

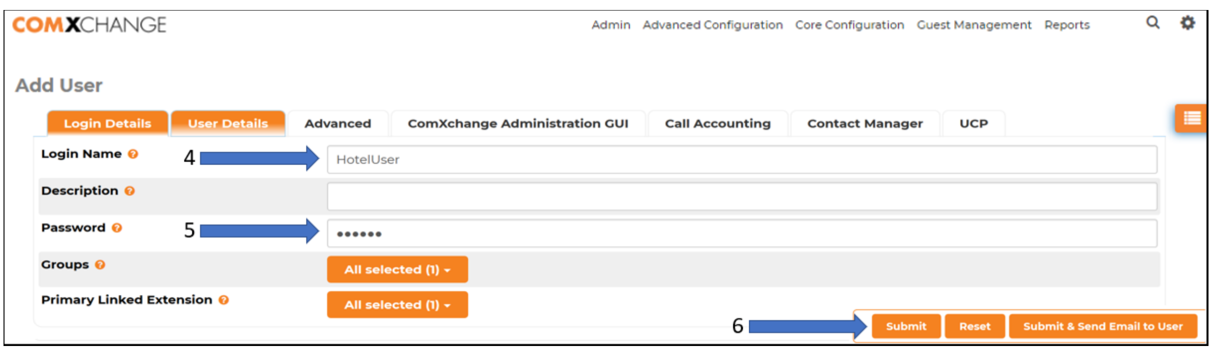
Navigate to Admin > User Management

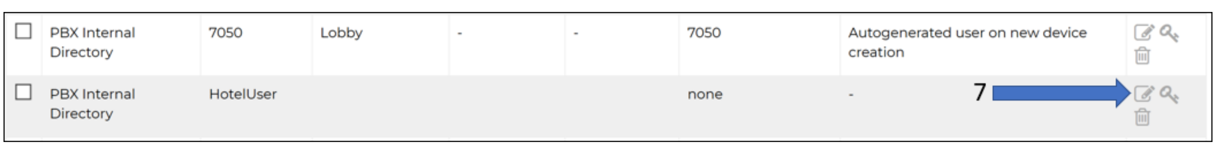
1. Choose any extension and click on the edit Action icon
2. Click on and open the Flyout Menu
3. Click on the Add User Button
4. Fill in the Login Name
5. Fill in a Password

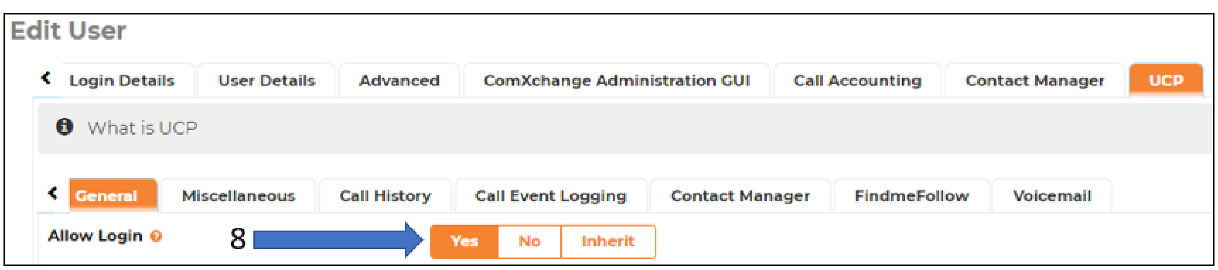
6. Submit and Apply
7. Click on the edit Action icon of the new user
8. In the UCP tab click on the Yes button to allow Login
9. Click on the Call Accounting Tab and Assign permissions for the user

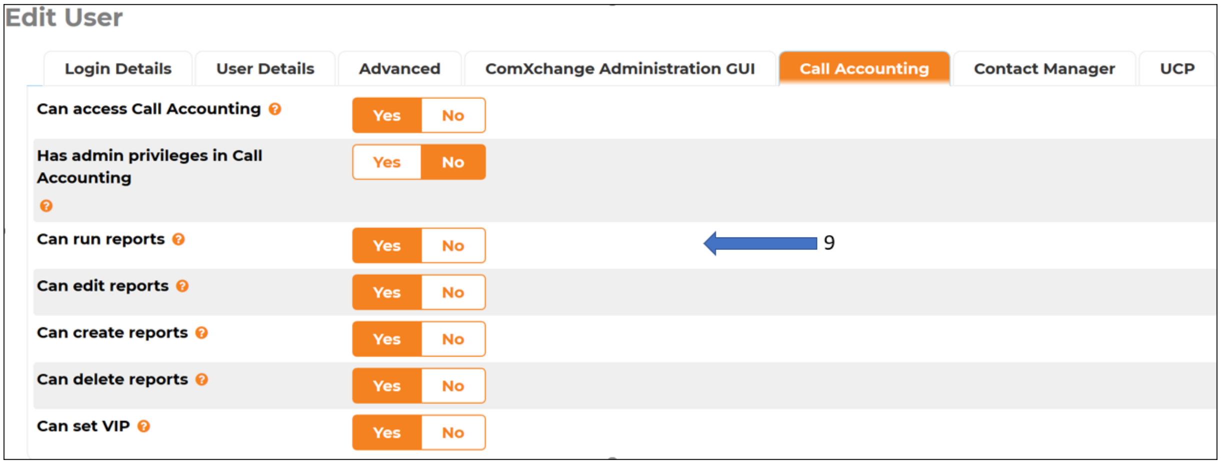




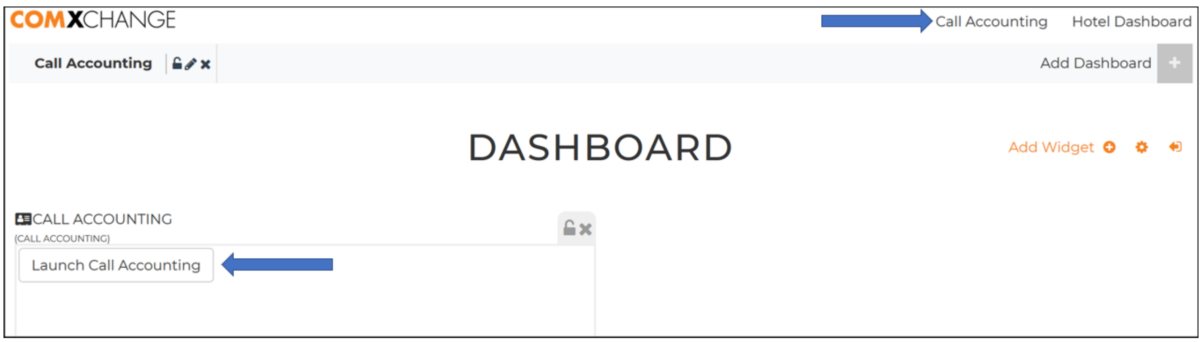




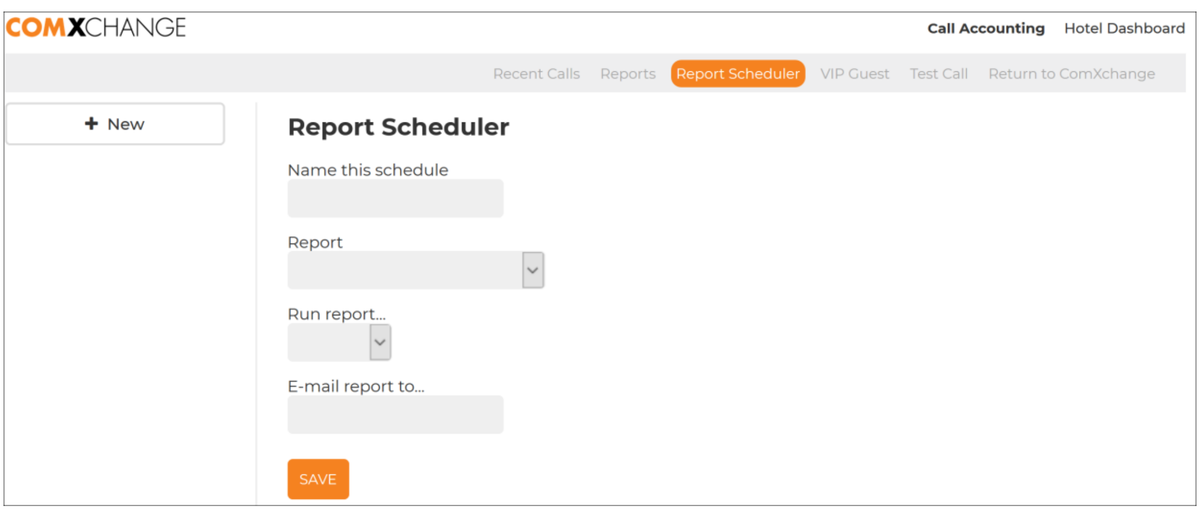




Log into the UCP at 192.168.101.2/UCP and Click on the Call Accounting Link or use the Call Accounting Widget to Launch the Call Accounting Module.



The User now has privileges to view, create, and schedule reports as well as the ability to mark Rooms as VIP's.



360 NETWORKS

Setting the New Standard